PH.D. THESIS ABSTRACT 2000-1991

Sr No	Title	Supervisor(s)	Author(s)	Year
1	A Novel Placement Strategy For Facts Devices In Multimachine Power Systems	Ghosh Arindam	Sharma Nikhlesh Kumar	2000
2	Soft-Switched Quasi-Resonant PWM Inverters	Fernandes B G& Dubey G K	Jafar Jamal Jafar	2000
3	Photovoltaic Airconditioning Refrigeration And Heating: An Environmental Friendly Option	Prasad Manohar	DhawanKirti Kumar	1999
4	Classification Of Partial Discharge Patterns Using Texture Analysis Algorithms	Arora Ravindra &Srivastava S C	El-rahman Mohamed Kamal Abd	1999
5	Control Of Subsynchronous Resonance Using Facts Devices	Ghosh Arindam Sachchidanand	Kumar S V Jayaram	1999
6	Active Harmonic Current Compensation Using Hard And Soft-WSITCHED Inverters	Doradla S R	Mahapatra Kamalakanta	1999
7	Boundary Detection In 3D Mri images Using Level SET Methods	Sharma Govind	Pila Ritesh Kumar	1999
8	Approximate Modeling Of The Call-Level Output Process Of An ATM Switch	Srivathsan K R & Bose Sanjay Kumar	Rao Thirumalasetty Srinivasa	1998
9	Parametric Synthesis Of Autopilot For Bank-To-Turn Missile	Hole K E	Narayan Shiv	1998
10	Tracking Error Based Sliding Mode Controller Design With Application To Flight Control	Hole K E	Singh Giresh Kumar	1998

11	Aspects Of Radial BASIS Function NET Design With Application In Speech Recognition	Mullick S K & Kundu Debasis	Bhattacharya Puranjoy	1998
12	Synchronous Link Converter Var Compensators And Active Power Filters	FernandesB G& Dubey G K	Chatterjee Kishore	1997
13	Coded Modulation Systems Using General Block Codes: A Structured Distance Approach	Siddiqui M U	Jadhav Ashish N	1997
14	Estimation of Depth from Monocular Defocused Imager	Gupta Sumana & Mullick S K	Rayala Jiendra Das	1997
15	Nonuniform Sampling Of Time-Limited Signals Using Time Scale Localisaiton Properties Of Wavelet Transfroms	Rao PRK& Sharma Govind	Ramesh Chaveli	1997
16	Application Of Converter Based Controllers To Electric Power Trnasmission System	Sachchidanand & Ghosh Arindam	Das Biswarup	1996
17	Morphological Processing Of Random Finite - Gray - Scale Digital Images	Siddiqui M U	Singh Balvinder	1996
18	Analysis Of Strip Excited Waveguide Slot	Sachidananda M	Sivadas Apu	1996
19	Temperature Effects On The Performance Of Mos Devices	Kumar Jitendra& Tyagi M S	Yadav Kalyan Singh	1996
20	Vertical And Horizontal Wire Antennas Above A Lossy Earth	Sachidananda M	Babu R Chaitanya	1996
21	Array Of High Directivity ENDFIRE Elements-A Study	Sachidananda M	Harish A R	1996
22	Adaptive Evaluation Of Dynamics Of The Vocal Tract Shape And Detection Of GCLs Using MP Filter	Ray G C	Dandapat Samarendra	1996

23	A Flow Model Based Control-Theoretic Approach To Dynamic Routing In Some Structrued Communication Networks	Rao P R K& Bose Sanjay Kumar	VinodkumarM V	1996
24	Parametric Conditions For Simultaneous Quadratic Stabilization Of Linear Systems	Sule Virendra Ramakant	Kar Indra Nayayan	1996
25	Connectionist Signal Processing System Characterization Of Representation In Neural Network	Rao P R K	Murthy Deepak	1996
26	A New Parallel Resonant DC Link Inverter-Fed Three-Phase induction Motor Drive	Doradla S R	Deshpande Vijay V	1996
27	Voltage Stability And Contingency Selection Studies In Electric Power Systems	Srivastava S C& Kalra Prem Kumar	Joshi Satish	1995
28	Artificial Neural Network Based Power System State Estimation	Srivastava S C	Kumar Vinod D M	1995
29	Multiple States Of Silicon Related DX Centers In ALxGA1-xAS	Sharan R & Mohapatra YN	Agarwal Sandeep	1995
30	Voltage Security And Loss Minimization Studies In electric Power Systems	Srivastava SC	Singh Sri Niwas	1995
31	Parametric Modelling Of Non-Stationary Signals By Linear Time-Dependent Processes	Sircar Pradip	Mukhopadhyay Sudipta	1995
32	Investigation Of Insluating Properties Of Vacuum Under high Voltage	Arora Ravindra	Khan Muhammad Abdul Goffar	1994
33	Investigation InTo Static And Dynamic Aspects Of Voltage Stability	Srivastava SC	Srivastava Kailash N	1994

34	Nonlinwear Dynamical Analysis And Predictive Coding Of Speech	Mullick SK	Kumar Arun	1994
35	Signals And Systems On Sets	SinhaVP	Subramanian Venkatesh K	1994
36	Collision resulution Algorithms For FInite User Blocked Random Access Channels	Chatterjee P K & Rao P R K	Karandikar Abhay	1994
37	Queueing Analysis of a Non-Preemptive MMPP/D/1/K Priority System for Applications in ATM Networks	Srivathsan K RBose Sanjay Kumar	Venkataramani B	1994
38	AC To DC BI-Directional PWM Voltage Source Converters	Dubey G K	Kanetkar V R	1994
39	Switch Mode Power Rectification	Dubey G K	DawandeM S	1994
40	Low Cost And Concurrent Algorithms For Adaptive Volterra And Linear Filters	Sharma Govind	ChaturvediAjit Kumar	1994
41	New Decoding Algorithms For Reed-Muller And Cyclic Codes Based On Spectral Domain Deconvolution	Siddiqui M U	Madhusudhana H S	1992
42	Novel Static Distance realying Schemes For Transmission Line Protection	Dubey G K & Singh L P	Gupta Ashvini Kumar	1992
43	Polyphase And Frequency Hopping Sequences Obtained From Finite rings	Siddiqui M U	Udaya P	1992
44	A Discrete -Time HVDC System Model For Stability Analysis And Self-Tuning Control Design	Ghosh Arindam & Sachchidanand	Pandey Rajendre Kumar	1991

Title	***:***	***A Novel Placement Strategy For Facts Devices In Multi Machine Power Systems***
Author(s)	***:***	***Sharma Nikhlesh Kumar***
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Abstract

A power system is a complex network comprising numerous generators, transmission lines, transformers and variety of loads. Because of the economic as well as environmental problems associated with adding new transmission lines to keep pace with the growing loads and generation, the major task utility industry today is the efficient utilization of the existing transmission systems. One of the approaches adopted for efficient utilization involves the use of power electronics technology. With the growing demand of electrical power, the existing transmission lines get increasingly overloaded. This increased loading of transmission lines has led to various problems associated with stability and maintenance of appropriate voltage levels across the system. A reliable solution of these problems can be achieved with the help of Flexible Alternating Current Transmission System (FACTS) devices, which may be connected in series, shunt or in a combination of series and shunt. FACTS devices include a host of fast, reliable solid state controllers [1]. However, these devices are effective only when they are placed at appropriately determined locations. There may further be situations where more than one FACTS device is employed in the system for voltage stabilization and stability enhancement etc. their collaborative interaction can play an important role in improving power transfer capability of the network and the overall power system stability. In this thesis an attempt has been made to evolve a simple strategy for placement of FACTS devices in a multimachine environment after a careful study of several existing techniques. A general purpose interactive MATLAB program has been developed for small signal analysis of multimachine system. Modeling of SVC, TCSC, STATCOM and SSSC is done and the system stability analyzed using these FACTS devices. Finally, an attempt has also been made to examine the interactions between shunt and series FACTS devices (namely, SVC and TCSC) and their impact on the system stability utilizing the proposed placement strategy. To obtain optimum system performance, the location of any FACTS device in an interconnected power system must be determined carefully. Four broad categories of techniques have been so far proposed for determining best suited locations. These are (1) Jacobian based sensitivity methodologies [2] (2) Non-Jacobian based techniques [3, 4] (3) Eigenanalysis based methods [5] (4) Optimization and Artificial Intelligence based techniques [6] Placement techniques (3) and (4) are complex and computationally time intensive but are claimed to be more accurate. Techniques (1) and (2) are comparatively simple and are shown to be reasonably accurate even though they do not consider system dynamics as in (3) and system nonlinearities as in (4). Between

For more details click here Back

Title : Soft-Switched Quasi-resonant PWM Inverters

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Abstract

In this thesis three novel quasi-resonant circuits are proposed and their performance is studied. These circuits are used as an interface between the dc supply and the conventional PWM inverter to provide zero voltage (ZV) instants during switching of the inverter devices. They generate a zero voltage notch of very small duration in the dc-link just prior to the inverter switching instant to facilitate zero voltage switching (ZVS) operation of the inverter devices. Because of ZVS, the switching losses and device stresses in PWM inverter are substantially reduced. As a result, these soft-switched PWM inverters can be operated at higher frequencies with reduced switching losses, device stress, EMI, and cooling requirements compared to the conventional PWM inverters. The proposed circuits offer several advantages over the ZVS topologies reported in the literature. Some of the significant advantages are the independent operation for quasi-resonant circuit, low voltage stress, true PWM operation using any conventional PWM techniques, and simple control circuit. Principle of operation of the proposed circuits is discussed and detailed mathematical analysis is provided. Guidelines for choosing the circuit parameters and evaluating the device stresses are also given. In addition, modification in the proposed topologies to improve the efficiency are also discussed. Detailed simulation studies are carried out to evaluate the performance of the soft switched converter for various load conditions. Switching instants for the PWM inverter are derived using third-harmonic injected sinusoidal PWM technique. In order to verify the simulated results a laboratory prototype is designed and fabricated. The steady state performance of the-switched inverter supplying R-L load and induction motor is experimentally verified. The experimental results are found to be in close agreement with the simulated results.

For more details click here Back

Title **:** ***Photovoltaic Airconditioning Refrigeration And Heating: An Environmental Friendly Option***

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Abstract

Synopsis The sources of conventional energy are dwindling fast with a corresponding rise in cost. Therefore, considerable attention is being paid to other alternative sources of energy. Solar energy is one of the non-conventional energy resources which has the advantage of being freely available in plenty, particularly in India. It can be harnessed as thermal energy for water heating, cooking, absorption-refrigeration and solar thermal power generation. It can be converted directly into electricity by using photovoltaic array. A photovoltaic array, called PV array, consists of a number of solar cells arranged in series and parallel. The electrical energy generated in the array can be stored in a battery and can be used to run a compressor of a vapor-compression system or to heat the generator of a vapour absorption system. The airconditioning, refrigeration and heating require energy in the form of heat (in the generator of a vapour absorption system) or in the form of work (to run the compressor of a vapor compression system). They require mostly the same type of appliances and process and differ in application. Whereas, in the context of hot coun tries like India, airconditioning is understood as a process of cooling a surrounding, the heating means the reverse of it in which a space is heated by the rejected heat of condenser. The refrigeration is defined as the artificial withdrawal of heat, producing in a substance or within a space a temperature lower than that which would exist under the natural influence of surroundings. Thus, refrigerating machine is an integral part v of airconditioning system. The term photovoltaic airconditioning, heating and refrigeration means running these systems by the solar energy converted in the form of electricity. As the so lar energy is not available round the clock and also the weather conditions are highly uncertain, the electrical energy is stored in the battery. Apart from the continuous depletion of natural resources and degradation of environment due to burning of fossil fuels, the solar energy is very much relevant for remote area applications. However, at present, the solar energy is very costly and therefore, the size of the array should be selected judicially. In the present work, a computer code is developed to find out the array size de pending on the application. A model has been proposed for finding out the array photo-generated current. The model is validated by conducting experiments on a 100 W photovoltaic array at IIT Kanpur. The cooling load, heating load and refrigeration load are computed on timely basis based on ASHRE handbook. Feasibility of two types of systems- vapour compression and vapour absorption system is considered. It is seen that the vapour compression system is economical. In this, the electrical energy will be used to run a D. C. motor which will be used to drive the compressor. As the weather conditions are highly

uncertain, the fuzzy set theory is employed to find out the array size. In this, the weather parameters are expressed as linear triangular fuzzy numbers and possibility distribution for array size is obtained. Then, a method has been proposed to find out the array size based on the subjective requirement of the designer. Once a proper size of the array is chosen, it has to be matched with the load connected to it. The load is a battery of 12 V or 24 V. By load matching is meant the proper selection of series and parallel rows of cells for a given total number of cells. A load matching factor is defined as the ratio of energy stored over the day to vi the maximum possible energy stored over a day. The number of series and parallel rows of cells and internal resistance of the battery is found by solving the optimization problem to maximize the stored energy with the constraint of total number of cells and generated voltage being greater than the battery voltage. A parametric study has been carried out and it is seen that a load matching factor of more than 0.95 may easily be obtained. However, the system will be highly inefficient in case of a load mismatch. The study has been also carried out to find out the proper inclination of the array in order to get maximum output from the insolation. For this purpose also, an opti mization algorithm has been employed. It has been also suggested to cover the space by photovoltaic array in a manner in which it will fulfill the twin objective of shading the roof to reduce the cooling load (refrigeration requirement) and generating electricity. Finally, the environmental benefits of using photovoltaic systems are studied in detail. It is seen that use of solar energy will reduce carbon dioxide level and ash particulates in the environment. Thus, the present work studies various aspects of the photovoltaic airconditioning, refrigeration and heating and discusses the importance of its use. The use of a new model for array-photo-generated current, application of the fuzzy set theory and a methodology to find out the optimum load matching and inclination of the array and environmental study are the main features of the present work. VII

For more details click here Back

Title **:** ***Classification Of Partial Discharge Patterns Using Texture Analysis Algorithms***

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Abstract

For more details click here Back

Title ***:*** ***Control Of Subsynchronous Resonance Using Facts Devices***
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Supervisor(s) ***:*** ***Ghosh Arindam & Sachchidanand***

Abstract

SYNOPSIS The continuously growing demand for electric power requires the transmission of large amounts of power over long distances An economically attractive solution to increase power transfer through long transmission lines, without building new parallel circuits, is to install the series capacitors It is known that series capacitor compensation benefits power systems in many was, such as enhancing transient stability limits, increasing power transfer capability, etc It is also known that fixed series compensation may cause sub synchronous resonance (SSR) in power systems, which can lead to the damage to machine shaft Since the discovery m 1970 that SSR was the mam cause of the shaft failures at the Mohave generating station (USA) extensive research and development efforts have been devoted to the development of effective SSR mitigation measures The definition of SSR is given by IEEE [1] Subsvnchronous resonance is andectac_ power systemjcondition where the electric network exchanges energy with a turbine genera tor at one or more of the natural frequencies of the combined system below the synchronous frequency of the system When a series compensated power system is perturbed, its equilibrium state is dis turbed, giving rise to interchange of energies between mechanical and electrical systems mutually coupled through the rotor of the synchronous generator As energies are inter changed, their frequencies of oscillations are the natural frequencies of their respective li systems The natural frequencies of oscillation of each system form their respective modes of oscillation An oscillation of generator rotor, due to disturbances, produce super syn chronous and sub synchronous components of currents (or voltages) These currents induce subsynchronous and supersynchronous torques There are many ways m which the system and the generator may interact with subsynchronous effects Couple of these interactions are (1) Torsional interaction and (2) Transient torque effect The turbine generator shaft system responds to disturbances with oscillations at its natural frequencies Oscillations of the geneiator rotor at its natural frequency result m modulation of generator voltage The subsvnchronous frequency voltage is the complement of the natural frequency of the turbine generator shaft system When this subsynchronous frequency is close to system natural frequency, the resulting armature currents produces a subsynchronous torque which re mforces the aforementioned generator rotor oscillations This can result m sustained or growing oscillations This phenomenon is referred to as torsional interaction System disturbances impose electromagnetic torques on generator rotor, subjecting shaft segments to torsional stresses Following a significant system disturbance m series capacitor compensated power system, the resulting electromagnetic torque oscillations at a frequency complement to the electrical system

natural frequency If this frequence coincides with any of the natural modes of the turbine generator shaft, there can be large peak torques This effect is referred as transient torque effect or shaft torque amplification Power electronic systems are finding increasing applications m power systems for both transient state and steady state operation requirements [2] STATCOM is a second gener ation FACTS device The STATCOM consists of GTO based voltage source inverter (VSI) with a dc capacitor coupling transformer and control circuit The VSI is connected to the AC bus through coupling transformer The variation m STATCOM output voltage can be achieved through the control of the firing angle [3] STATCOM with voltage controller for SSR mitigation is proposed in the literature The voltage controller along with the reactive in current controller may be considered for improving the performance of the STATCOM Controlled series compensation (CSC) is a type of Flexible A.C transmission systems (FACTS) controller which utilizes thynstor valves to control the degree of series compensation [4] CSC consists of two components, one element is a mechanically switched portion the second portion being a Thynstor Controlled series capacitor (TCSC) The TCSC con sists of a fixed capacitor m parallel with a thynstor controlled reactor(TCR) The TCR reactance is varied by varying the firing angle of the thynstors, which in turn varies the effective capacitive reactance Discrete time and continuous time TCSC models were pro posed in the literature to study the SSR phenomenon and to estimate the damping of torsional modes m open loop The TCSC control strategy can be utilized to study the impact of TCSC on torsional modes It is mentioned previously that series compensated power system gives raise to subsyn-chronous phenomenon Vanous methods of counteracting subsynchronous resonance phe nomenon have been considered which are great diversity m concept and methods Most of these concepts were tested with IEEE first bench mark model One of these methods is to modulate either real power or reactive power injected into the generator bus of the affected steam turbine The modulation of real power and reactive power is one of the effective means of mitigating subs nchronous resonance and this can be achieved by FACTS con trollers The real power modulation can be achieved by the modulation of TCSC reactance whereas reactive power modulation can be obtained by STATCOM The torsional interaction may be considered as small disturbance phenomenon whereas transient torque problem is due to the large disturbance The torsional interaction analysis is usually carried out utilizmg linearized models The study of transient torque problem requires the detailed modeling of all system nonhnearities Digital simulation is well suited for analyzing transient torque effect [1] Subsynchronous resonance control through FACTS controllers is gaining importance IV Based on the above considerations, this thesis focussed the attention on control of subsyn chronous resonance which includes torsional interaction and transient torque effects with FACTS controllers such as STATCOM and TCSC The objectives and scope of thesis are 1 To study the subsynchronous transient oscillation damping with the STATCOM 2 To propose a new digital control scheme for TCSC control to damp subsvnchronous resonance oscillations 3 To study the control interactions between TCSC and power system stabilizer An outline of the work reported in the thesis is given below The first chapter gives an introduction to the various aspects of the problem presented m the thesis and reviews briefly the previously published literature In Chapter 2 linearized model of generator system with detailed representation of stater, rotor and mechanical system along with AC network is presented The study system m this Chapter and subsequent Chapters is the celebrated IEEE First bench mark model [1] Eigenvalue analysis is carried out with the linearized overall system model The results obtained through eigenvalue analysis is validated with time domain simulation and Fourier analysis The effectiveness of NGH damping scheme [5,6] m mitigating the subsynchronous torque oscillations is studied through the detailed time domain simulation In Chapter 3, linearized model of 12 pulse STATCOM along with voltage and reactive controller is presented [7] The control parameters are obtained through eigenvalue analysis of the linearized model The results obtained through eigenvalue analysis are validated with time domain simulation for the study of transient torque oscillations Comparative study is carried out with NGH damping scheme in mitigating subsynchronous torque oscillations SSR analysis with TCSC in continuous time linearized domain is presented m Chapter 4 Constant angle control of TCSC is investigated The constant angle control is based on v the philosophy of maintaining the voltage drop across the compensated transmission line constant Eigenvalue anal} sis is carried out to predict the stability of the system and the results are validated with time domain simulation In Chapter 5 a noel digital control scheme is proposed for TCSC control For this purpose a discrete time domain TCSC model is developed The controller is designed using state feedback approach The discrete time model along with the controller is suitably interfaced with the generator system model to obtain a homogeneous state space equation The stability of the system is analyzed through eigenvalue analysis and the results are validated with time domain simulation The effect of excitation system that includes power system stabilizer (PSS) on tor sional modes, in the presence of TCSC controller is examined in Chapter 6 The control interaction study is carried out through eigenvalue analysis This thesis concludes in Chapter 7 outlining the conclusions drawn from the thesis and suggest some future scope of woik References [1] IEEE SSR Task force Proposed terms and definitions for subsynchronous resonance in series capacitor compensated transmission lines IEEE Trans on PA.S, Vol PAS 92 March/April 1980, pp 506-oll [2] N G Hingorani FA.CTS Flexible AC Transmission Systems, paper presented at IEE fifth international conference on AC and DC transmission, sept 1991 London, pp 1 7 [3] C Schauder et al Operation of a ± 100 MVAR TVA STATCON IEEE Trans on Power Delivery, Vol 9, No 2 1994, pp 1018 1027 [4] N Chnstl, R Heidm, P E Krause and S M Mckenna Advanced series compensation (ASC) with thynstor controlled impedance CIGRE regional meeting, session 1(92, Paris [5] N G Hingorani A new scheme for subsynchronous resonance damping ol torsional vi oscillations and transient torque Part 1 IEEE Trans on PAS, Vol PAS-100, No 4 1981 pp 1852 1855 [6] K B Stump R Heidm and N G Hmgorani A new scheme for subsvnchronous reso nance damping of torsional oscillations and transient torque Part 2 IEEE Trans on P<VS Vol PAS 100 No 4 1981, pp 1856-1863 [7] C Schauder and H Mehta Vector analysis and control of Advanced static var compen sator IEE Proc C Vol 140, NO 4, july 1993, pp 299 306 [8] 4 Ghosh and G Ledwich Modeling and control of thyristor controlled series compen sators IEE Proc Generation Transmission and Distribution, Vol 142 No 3 May 1995, pp 297 304 vu

For more details click here Back

Title : ***Active Harmonic Current Compensation Using Hard And Soft-Wsitched Inverters***

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Abstract

The extensive use of non-linear loads results in harmonic pollution in a power distribution system. This lead to waveform degradation in power supply networks. A number of techniques such as use of passive filtering, active filtering, hybrid filtering etc. have been suggested in the literature for compensation of line current harmonics. However, all these schemes suffer from one or more drawbacks such as load dependence large system size, inadequate current regulator bandwidth, poor system efficiency and complexity of control. Passive filters that are made up of inductive and capacitive elements tuned to the harmonics to be compensated are used for compensating harmonics. These filters have certain disadvantages that are well documented in the literature. Active filters are currently regarded as the most efficient option to solve the problems created by non-linear loads [1]. The parallel active filter is the most common configuration for active filtering application. The filters is connected in parallel with the load being compensated. For the power circuit an inverter operating in current control mode is used. The current compensation is usually done in time domain for faster response. The purpose is to inject a compensating current at the shunt point that the source current becomes sinusoidal. Since this compensator is used for canceling the harmonics we refer it as an active harmonic current compensator (AHCC). Te reference current that the compensator must follows or track is a function of the load current. However, the load current is not known a priori. Moreover the load may also change. Therefore a suitable strategy must be employed for the extraction of the reference current. This is one of the biggest challenges in an AHCC design. Out of the several schemes suggested in literature, the hard-switched PWM inverter based shunt compensator has gained prominence. However this compensator has its inherent limitations of high switching losses because of hard switching. Switching lasses are one of the main restrictions that is put on the high operating frequency. It also requires a large dc link filter and hence its time response it sluggish. For proper current tracking, the approximate current bandwidth is usually the PWM frequency divided by a factor of ten. Therefore, PWM based compensator fails to track high frequency components, particularly at high power level. For example if the harmonics up to nineteenth are to be cancelled for a 500 kilowatt converter load, then the compensator should have a power rating of 100 kVA and it must be capable of switching at a frequency of about 9.5 kHz or higher. This is a difficult task given the current state of the art of power semiconductor device technology. Using conventional switching techniques inverters of over 10 kW are restricted to operate at frequencies of below 10 kHz. Without adequate current regulator bandwidth compensation can never be perfect. For active harmonic current compensation a high power topology with adequate current regulator bandwidth is necessary. In essence it is required

to realize a large capacity non-sinusoidal current source generator which must follows its command instantaneously. An important inverter topology is the so-called resonant dc link inverter (RDCLI) [2]. This soft-switched inverter would provide adequate current regulator bandwidth because of its high frequency of operation and hence an ideal candidate for active filtering application. Moreover as the switches in this inverter are switched at zero voltage crossing (ZVS), the switching losses will be a minimum and hence the AHCC can achieve high efficiency. This simple topology however has few drawbacks. These are higher device voltage stresses (when the output voltage is greater than twice the dc input voltage), zero crossing failure unless the initial current in the resonant inductor is built properly. Keeping in view of the above consideration we define the following objective of the thesis. 1. To find out a suitable topology for AHCC such that the compensator would achieve high efficiency, provide adequate current regulator bandwidth and fast transient response 2. To devise a suitable control strategy for the extraction of compensator current reference. 3. To study the different control aspects of the AHCC such as current regulation within the inverter and any other control required for the inverter. 4. To evaluate the performance of the AHCC in terms of compensation response time, efficiency etc, through simulations and experiments. The major contributions of this thesis are: 1. A suitable topology for the power circuit of AHCC application is determined. Different power converter topologies are investigated for active filtering application. It is possible to achieve compensation with a hard-switched PWM inverter based compensation when its switching frequency is high. However, the increase in frequency will result in increased losses and higher device stresses. On the other hand a soft-switched inverter based AHCC is able to compensate all load harmonics because of high frequency switching. Furthermore, this compensator is efficient in the sense that the switching losses are a minimum. It is demonstrated that a three-phase AHCC would work provided the dc side capacitor is split into half and the neutral is connected to the load neutral. A topology for three-phase AHCC is proposed, in which three single-phase RDCLI energized by a single dc capacitor is used as the power circuit. These inverters are connected to the ac bus through three different isolation transformers. 2. In this thesis a new current initialization scheme is proposed for a resonant dc link inverter (RDCLI) [3]. The method of current initialization is based on the state transition analysis of the system as a boundary value problem (BVP). It is shown that for a given load current, it is possible force the dc link voltage to go to zero at a prescribed time by properly choosing the initial dc link current. This technique makes it possible to operate the resonant dc link inverter without any zero-crossing failure which is an important issue for a satisfactory operation of such an inverter. 3. In this thesis two methods are proposed for the extraction of active fundamental current of the load current. In the first method, the error in the dc capacitor charge is passed through a controller. The output of this controller is multiplied by the template of the source voltage to give the desired source current. The reference for the compensator is obtained by subtracting this value from the instantaneous value of load current. In the second method the average value of the capacitor current over one cycle is passed through a controller to give the loss component of the AHCC. This quantity is added with the peak value of the source current (obtained through power calculation) and is then multiplied by the template of source voltage. The resultant quantity gives the instantaneous value of the desired source current. The reference current for the compensator is obtained by subtracting the desired sources current value from the instantaneous value of load current. It is possible to extract the compensator reference using the above two methods. However, the response of the second controller is slow when compared to the first one. 4. For the proposed charge based model two controllers are investigated [4]. The PD controller gives a superior performance compared to the PI controller. The PD controller has a faster response better stability margins. The AHCC losses decrease with a PD controller to a certain extent. 5. PC interface is used for the implementation of the proposed current initialization scheme. The use of a PC makes the system more flexible for conducting experiments. This facilitates zero-voltage switching by taking into account of the actual circuit delays and tolerances.

For more details click here Back

Title	***:***	***Boundary Detection In 3D MRI Images Using Level Set Methods***
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Abstract

It is well known that in,Digital Imaging, .detection of boundaries both in 2D and 3D is very important. In medical imaging one is interested only in a specific region of image, so one needs an algorithm for isolating accurately the region of interest. There arc several algorithms available with their own merits and demerits. In this thesis we have used the concept of active contour method using the level set. By this method we move the initial contour till it hits the boundary of the object. To achieve the efficiency in computation we have used aLnarrow banding methodjby which only a portion of the image plane is considered for the calculation. In narrow band method we rebuild the level set function whenever it collides with the boundary and use it as zero level set(initial front) for further evolution. We have presented a number of-MRI images^with the segmented objects of interest. Examples have been given in both 2D and 3D. We have used the Visualization Tool Kit for displaying ^D images, j In our method, objects of interest are chosen by the user by specifying an initial curve or surface. v

For more details click here Back

Title **:** ***Approximate Modeling Of The Call-Level Output Process Of An ATM Switch***

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Roll No **:** ***9310461***

Supervisor(s) **:** ***Bose Sanjay Kumar&Srivathsan K R***

Abstract

Broadband services like video conferencing and video-on-demand generate bursty and high bandwidth traffic. The quality of service (QOS) requirements of these services also vary from one service to the other. Traditionally, telecommunication networks are service specific and may not be capable of supporting broadband applications efficiently [Arm 87]. Broadband integrated services digital networks (broadband ISDN or B-ISDN) are designed as an alternative to such service-specific networks. Apart from supporting broadband services, B0ISDN networks are also capable of carrying multiple number of services with different traffic characteristics and QOS requirements simultaneously. These networks also support point-to-multipoint connections. Asynchronous transfer mode (ATM) is considered to be the most preferred carrier network for broadband ISDN. In ATM networks each call will be allocated the required network resources like channel bandwidth and buffers in varying quantities. During the call establishment phase the required resources for the incoming call will be estimated and the call will be admitted only if a path between the origin node and the destination is found in which every ATM switch has sufficient resources to support the new call. For the efficient utilization of the network resources the calls are statistically multiplexed at every node in the ATM network, thereby exploiting the multiplexing gain arising from the burstiness of the sources. This however would cause intermittent congestion in the network affecting the QOS parameters of individual calls. Resources management techniques are devised to minimize this intermittent congestion. The traffic characteristics of a call would change while it passes through an ATM switch due to statistical multiplexing. Depending on the traffic load at the switch the bursty traffic may get smoothened requiring less amount of resources at the immediate downstream nodes to support the call while guaranteeing its QOS parameters. Knowledge of the call level characteristic at the output of the switch will be useful in devising better resource management techniques and also for better estimation of the end-to-end performance parameters of the ATM network, the like throughput and end-to-end cell delay. However the derivation of the call level characteristics at the output is complicated due to the statistical multiplexing of the bursty sources [Lau 93]. In this thesis we study this problem and propose two empirical approximations to arrive at the call level characteristics at the output of an ATM switch which is fed by a set of ON/OFF sources. A pure ON/OFF source (in which the interarrival time of cells during an On duration is constant) will get transformed into a distorted ON/OFF source while passing through a statistical multiplexer. In a distorted ON/OFF source the inter arrival time of cells is a random variable. At the output of a statistical multiplexer, the instantaneous inter cell departure time of cells belonging to the

target source depends on the following two factors 1. The number of cells from the interfering sources that are queued up in between two cells of the target source and 2. The queue length at that instant. The total number of cells emitted by the interfering sources during any time duration depends on the number of interfering sources that are in the ON state and their peak rates. This number of cells in turn decide the fraction of the channel bandwidth available to the cells of the target source. Based on this observations we model the statistical multiplexer of finite number of ON/OFF sources as an infinite buffer queue with only cells from the target source as input. This queue is assumed to be served by a variable service rate server whose service rate is equivalent to the share of the channel bandwidth available to the cells of the target source. The server is also assumed to go on vacations in such a way that each of the incoming cells to this queue finds the server on vacation with only one or no cell waiting in the queue. These vacations of the server will take care of the time required to serve the cells waiting in the queue of the statistical multiplexer which arrived ahead of a cell of the target source. The inter cell departure time of this modified queuing system equals that of the cells belonging to the target source in the statistical multiplexer. This is modeled in this thesis as a doubly random variable. It probabilistically equals either of the two following instances one is equal to the instantaneous service time and the other is equal to the maximum of the instantaneous service time and the inter arrival time. The probability with which the instantaneous inter cell departure time equals the instantaneous service time is the probability that at the departure instant of a cell the next of cell of the same ON duration has already arrived. This probability is calculated in terms of the queue length distribution and the probability distribution of the number of interfering sources that are in the ON state. At any instant the fraction of the channel bandwidth available to the cells of the target source is a discrete random variable. In chapter 3 we approximate this discrete random variable as a continuous one using maximum entropy principle [Shor 80, jay 57] and obtain the probability density function of such a continuous random variable. We use first order moments of the instantaneous fraction of the channel bandwidth available to the cells of the target source as a constraint in this formulation. Using this probability density functions an approximate expression for the Laplace transform of the probability density of the inter cell departure time is obtained. Assuming that during an ON duration, the changes in the instantaneous bandwidth available to the cells of the target source and those in the instantaneous queue length are negligibly small, we also obtain the Laplace transform of the probability density of the burst length of the target source at the output. The accuracy of this approximation is verified by comparing the first order moment of the burst length obtained using this approximation with that obtained from simulations. As the exact expression for the queue length distribution of the statistical multiplexer is not available, the queue length distribution is calculated using the approximate procedure proposed by Heffes et al [Hef 86]. The accuracy of the usage of such an approximate queue length distribution is also verified. We have also formulate another optimization problem using maximum entropy principle forobtaining the conditional probability density function of the instantaneous channel bandwidth available to the cells of the target source conditioned on the queue length. In this formulation we used the relationships among the conditional averages of the instantaneous available bandwidth conditioned on the queue length as additional constraints. These relationships are based on the fact that as the traffic load increases the queue length increases and the instantaneous available bandwidth decreases. We study the variations in the instantaneous available bandwidth during an On duration in chapter 4. We derive the state transition probability matrix of the instantaneous available bandwidth. From this matrix we observe that the stochastic process of the instantaneous available bandwidth is non-stationary during an ON duration. We approximate the nonstationary instantaneous available bandwidth as a stationary one whose conditional probability distribution is equal to the average of the 1-slot state transition probability matrix over all slots of the ON duration. The instantaneous bandwidth available to the cells of the target source during an ON duration varies over a range which is a subset of all positive values and the maximum of this range is observed to be a characteristic of the states of all the interfering sources at the beginning of the ON duration. Based on this we model the instantaneous service time of cells belonging to the target source as the sum of two components one remains constant during the ON duration and the other varies from cell to cell. We also model the instantaneous inter cell departure time as a conditional sum of either two truncated exponential random variables or a constant and a truncated exponential random variable. Using this empirical model of the inter cell departure time the call level characteristics of the target ON/OFF source at the output of a statistical multiplexer is described by a set of 9 parameters. This description is generic enough so that the pure ON/OFF source (in which the inter arrival time of cells during an ON duration is constant) can also be described in terms of this 9-tuple of parameters. The ON/OFF sources entering into the second stage ATM switching in the path are of a distorted type and can be described by the 9-tuple of parameters as above. For the analysis of the call level characteristics of the target ON/OFF source at this switch the queue length distribution of a statistical multiplexer which is fed by a set of distorted ON/OFF source, required. We extend the approximate procedure proposed by Heffes et al [Hef 86] and approximate the superposition of a set of distorted ON/OFF sources as a 2-state MMPP process. Using this we obtain the approximate queue length distribution of the statistical multiplexer of a set of distorted ON/OFF sources. We study the call level characteristics of the target source at the output of a statistical multiplexer of a set of distorted ON/OFF sources in chapter 5. We obtain approximate expressions for the state transition probabilities of the distorted ON/OFF source during a random duration. Using these expressions, we obtain the state transition probability matrix of the instantaneous bandwidth available to the cells of the target distorted ON/OFF source in the statistical multiplexer. The important statistical properties of this instantaneous available bandwidth the similar to those of the instantaneous available bandwidth of the pure ON/OFF source case. Based on the observation, the inter-cell departure time of the cells of the target distorted ON/OFF source at the output of the statistical multiplexer is also modeled as a conditional sum of two truncated exponential random variables or a constant and a truncated exponential random variable. The corresponding call level characteristics can be described by a similar set of 9 parameters as in the case of pure ON/OFF

For more details click here Back

Title : *Parametric Synthesis Of Autopilot For Bank-To-Turn Missile*
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Abstract

Two basic methods of controlling a missile achieve the commanded acceleration are skid-to-turn (STT) and bank-to-turn (BTT). In STT the body acceleration is attained bypermitting the missile to develop both an angle of attack and a sideslip angle .In contrast,a BTT missile should not have ideally any sideslip .To achieve the desired orientation itis rolled (banked) such that the plane of Maximum aerodynamic NORMAL force is firstoriented in the desired direction and then the force is controlled by adjusting the angle ofattack. The function of an auto pile is to case the missile to achieve the commandedacceleration as closely as possible, while maintaining closed loop stability and certainconstraints on other operating variables. ABTT missile is asymmetric in configurationhaving very strong roll-yaw coupling .Due to the asymmetric airframe the missile hastendency for sideslip .If sideslip is allowed to develop it would give rise in the mostsignificant is on allowable sideslip. Because of the large roll due to sideslip the roll-yaw channels are strong coupled andcan be considered as a single system .The linerized pitch dynamics on the other hand canbe treated separately from roll-yaw dynamics. Thus on the whole the another theautopilot design for a BTT missile consists essentially of the design of two autopilot: onefor the pitch acceleration command tracking known as longitudinal autopilot and anotherfor commanding the roll rate about the velocity vector of the missile known as the lateralautopilot. There are basically two schemes of designing a linear time invariant (LTI) controller;the first one based on analytical methods and the second one based on parameteroptimization methods .The most common examples of analytical methods are linearquadratic Gaussiaan (LQG) based designs and ?H optimal controller: The weightmatrices are specified in the objective function .The major disadvantage of these methodsis that the design specification need to be translated into objective function and theweight matrices .In complex systems this is a difficult problem . On the order hand in the parameter optimization methods one starts with the controllerstructure .The next step is to from an objective function that represents designspecifications. An objective function can be formed by weighted sum or maximum ofvarious performance indices. Certain explicit constraints may also be added and thedesign is posed as nonlinear optimization problem, which can be solved usingoptimization methods .The main advantages of parameters optimization are;(i) A much greater range of objective functions and constraints can be allowedthan those in analytical methods.(ii) If the designer is certain that some controller of the selected architecture willdo the job the parameters of the controller can be

ordained using parameteroptimization.(iii) If a controller has been designed using analytical methods, it can beimproved issuing optimization methods.(iv) The designer has the freedom as regards the structure of the controller.However the parameters optimization methods pose the problems that they donot guarantee global solution. Moreover the require initial guess to start theoptimization process. These drawbacks can overcome by using recenttechnique of Simulated Annealing (SA) which provides global solution Asregards the choice of the controller structure the controller parameterizationsbased on the factorization approaches due to Youla et and Desoer et can beused.(v) A fast maneuvering high performance BTT missile has to function in anuncertain and changing environment. This demeans certain levels of timedomain and frequency domain responses from he missile system .Theparametric methods of controller design provide a good framework formeeting the demanding requirements on the time domain and frequencydomain performances .The time domain and frequency domain performancefunctions can be defined based on the deign requirements and the designproblem can be posed as an optimization problem that can be solved usingoptimization methods. Most of the autopilot designs for BTT missiles reportedin the literature are based on the analytical methods .The autopilot design inthis framework has not been reported. Therefore the motivations behind the work reported in thesis are;(i) To explore the possibility of parametric synthesis of longitudinal (pitch)autopilot and develop design algorithm using nonlinear optimizationmethods.(ii) To explore the possibility of parametric synthesis of longitudinalautopilot in two degrees of freedom (TDOF) configuration and developthe design algorithm using Simulated Annealing (SA) technique.(iii) To explore the possibility of parametric synthesis of lateral (roll-yqw)autopilot and develop the design algorithm using SA technique.(iv) To investigate the robustness of the designed autopilot. A brief description of the work reported in this thesis is given below: The first chapter introduces the control problem for BTT missile the autopilot designrequirements and presents a brief literature survey on the subject. The motivations behindthe research work carried out in this thesis are gives along with the brief description ofthe subsequent chapters. In the second chapter the longitudinal autopilot has been designed based on the pole-zero assignment using nonlinear optimization methods .The control law is based on theYoula factorization in the ring of polynomials. The design problem has been posed asnonlinear optimization problem, which has been solved using the direct search methodsof Simplex search, Powell's conjugate direction and Pattern search. In the third chapter the longitudinal autopilot has been designed in TDOFconfiguration .The time domain and frequency domain performances have been achievedindependently through the choice of the free parameters. These two free parameters areselected using closed loop model matching concept and frequency domain loop shapingusing constrained optimization with SA technique for achieving the time domain andfrequency domain performances respectively. Inn the fourth chapter the lateral (roll-yaw) autopilot has been designed for stabilityaxis roll rate command tracing using constrained optimization with Simulated Annealing.The control law which is based on Youla parameterization facilities input-outputdecoupling of roll and yaw channels. As results of which body axis roll rate and yaw ratecapabilities are achieved independently. In the fifth chapter the robustness analysis has been carried out for neglected andmismodeled actuator dynamics and neglected sensor dynamics. In the sixth chapter the thesis is concluded and the main finding and contributions arehighlighted.

For more details click here Back

Title **:** ***Tracking Error Based Sliding Mode Controller Design With Application To Flight Control***

Author(s) **:** ***Singh Giresh Kumar***

Roll No **:** ***9210464***

Supervisor(s) **:** ***Hole K E***

Abstract

TO steady the feasibility and applicability of the studies Mode control CSMS method using nonlinear discontinuous control low and offers attractive robust off properties for the flight control system (FCS). The method involves the selection of a sliding surface (SS) & the choice of a control low to make to be complicated non-linear functions of the aerodynamics & control surface selections. Analytical control Laws for two operational modes of the aircraft and I) the velocity tracking mode 4 ii) the trimming modes are derived results of the numerical simulation using parameters of the F-16 aircraft confirm the robustness of the control Law and establish the application of the design method. The SMC design in presence of unmatched uncertain and disturbances as it world be present in most practical systems is developed based on a multiloop (ml) strategy. The f propagation of disturbances in the ML starting is better handled obviates the need of derivatives of the uncertainties disturbances leading to a simpler robustness analysis. The state and input errors are shown to be ultimately bounded.

For more details click here Back

Title : ***Aspects Of Radial Basis Function Net Design With Application In Speech Recognition***
Author(s) : ***Bhattacharya Puranjoy***
Roll No : ***9220466***
Supervisor(s) : ***Mullick SK&Kundu Debasis (Math)***

Abstract

This thesis addresses the problem of pattern recognition (PR) using a class of neural networks called the Radial Basic Function (RBF) Net. The problem of PR arises in an effort to imbue cognitive capabilities into digital computers. Most of us realize, for example, how advantageous it would be if a word-processing package would allow verbal entry of data. This would require segments of the transduced speech signal to be classified into the corresponding acoustic events; where each acoustic event could signify a phoneme, which is the simplest acoustically distinguishable portion of an utterance, a syllable, a word, or even a sentence. It is this process of classification that forms the essence of PR. PR mechanisms, like the one required for the speech-to-text problem are however, extremely difficult to automate. The primary reason for this is that we still have very little idea about how the human brain actually stores on existing data structures and models- which happen to be diverse – in realizing pattern classifiers. The processing models expanded upon in this thesis posses the following properties: • For a fixed given set of items (each item comprising of measurements made on the entities to be classified), these models are capable of reproducing the class information with arbitrary precision in case no bounds are placed on their size. This makes them suitable for application in a fairly large class of PR applications. • These models allow their classification performance to be traded against their size. An improvement in the error performance (meaning fewer misclassifications) can usually be achieved at the cost of increased computational and storage costs. • Within limits, these models allow their internal data representations to be interpreted, so that these can be modified or constrained using apriori information available to he user about the problem from other sources. This synopsis is aimed at providing a glimpse of some of the crucial ideas in the thesis, apart from summarizing its contents. The description of the notations used is followed by a brief introduction to the class of Artificial Neural Network models, of which the RBG net is an instance. After this, a statement of the design problem is made, along with an exposition of the structure of the RBF net. It is then shown as to how the RBF model fits into a PR formulation with an appropriate specification of the input and output spaces. Then the topic moves onto template-matching, which is one of the central concepts in the thesis. This is finally followed by a mention of the focal points of the thesis and a chapterwise summary of the same.

For more details click here Back

Title	***:***	***Synchronous Link Converter VAR Compensators And Active Power Filters***
Author(s)	***:***	***Chatterjee Kishore***
Roll No	***:***	***9210466***
Supervisor(s)	***:***	***Fernandes B G & Dubey G K***

Abstract

Traditional methods of Static Var Compensation based on Switched Capacitor or Fixed Capacitor and Thyristor Controlled Reactor (TCR) are increasingly being replaced by new approaches utilizing the concept of Synchronous Link Converters. The two main advantages for which this class of var compensators has drawn tremendous interest from the researchers and power regulating authorities, are –I) considerable reduction in passive element count thereby reducing size, losses and projected cost, and ii) near constant var generation capability (both leading and lagging) even during low voltage condition. Moreover, as these compensators are realized by self-commutated devices, better controllability over switching nonlinearities is ascertained. Hence not only the harmonics generated by these compensators remain low, but also the harmonics generated by the load can be reduced by proper choice of switching function. Being a recent field of study, this class of compensators are known by different terminologies such as Var Generators, Advanced Static Var Generators, Synchronous Solid State Var Compensators, PWM Inverter Var Compensator, STATCON, STATCOM, etc. in this dissertation it is termed synchronous link converter var compensator (SLCVC). When SLCVC is utilized for compensation of load harmonics, it is known as active power filter or active power line conditioner. Basically SLCVCs are of two types. One is the voltage sourced or capacitively loaded SLCVC while the other type is known as current sourced or inductively loaded SLCVC. In this dissertation only the voltage sourced SLCVCs are considered. Depending on the control strategy, SLCVCs are further classified into –I) controlled current SLCVC and ii) indirect current controlled SLCVC. A controlled current SLCVC having instantaneous compensating feature is taken up as the first investigative object of the dissertation. The scheme does not require the sensing of reactive volt-ampere or harmonics involved in the load. Unlike the instantaneous compensation scheme proposed by Akagi (H. Akagi, Y Kanazawa and A Nabae, " Instantaneous reactive power compensators comprising switching devices without energy storage components", IEEE Trans. IA-20, No. 3, May/June 1984), the present scheme is realized by a simple control structure thereby enhancing the system reliability. Moreover, the proposed technique is applicable for compensating three phase as well as single phase loads. A mathematical model for the scheme is derived. The validity of the scheme is verified through extensive simulation and experimental studies. The controlled current SLCVCs cannot be applied for high power applications. This is due to the fact that high power devices like GTOs do

not comply to the high frequency switching requirement of controlled current topologies. For these applications either Parallelly operated stepped converters or multilevel converters are currently being considered. Limitations of parallelly operated converted topology are –1) transformer bank required at the input side of the compensator is complicated and bulky, 2) poor performance/cost ratio and 3) control structure is complicated. The limitations of the multilevel converter topology are –1) poor performance/cost ratio, 2) voltage unbalance among the different levels of capacitors, 3) increase in active component count (in diode clamp topology) or increase in passive component count (in flying capacitor topology) and 4) complexity in control. In this dissertation a new concept of harmonic elimination, suitable for high power applications is proposed. High power low frequency and low power high frequency devices are effectively combined to extract superior performance characteristics from the compensator. A mathematical model for the scheme is developed. The viability of the technique is ascertained through extensive simulation and experimentation. A modified version of the scheme having simplified control structure is also investigated. The validity of this simplified scheme is again confirmed through simulation and experimental studies. SLCVCs based on indirect current controlled technique require a reactive volt-Ampere calculator (RVAC) when they are used for load compensation. These RVACs are sensitive to system frequency and parameter variation. As a results the compensation process becomes inaccurate. Moreover, the need for frequent tuning of RVAC makes the process complicated. To overcome these limitations a novel technique for load compensation is proposed wherein the requirement of RVAC is eliminated. The effectiveness of the proposes scheme is validated through detailed simulation and experimental studies. Thus in this dissertation a consolidated effort is made to improve the performance characteristics of the voltage sourced SLCVCs. Limitations of the available control strategies and topologies are identified. New proposals are made to overcome those limitations. The viability of the proposed schemes are confirmed through extensive simulations and experimental studies.

For more details click here Back

Title **:** ***Coded Modulation Systems Using General Block Codes: A Structured Distance Approach***

Author(s) **:** ***Jadhav Ashish N***
Roll No **:** ***9210462***
Supervisor(s) **:** ***Siddiqui M U***

Abstract

The objectives of the thesis is to develop a general framework for the study of block coded modulation [BCM] and present schemes for code search block encoding and soft decoding of general (non-linear) block codes to be used in BCM systems over memory-less additive white Gaussian noise [AWGN] channels. BCM is a coded modulation scheme which uses block codes. The major advantage of BCM is that general (non-linear) block codes which have a larger rate as compared to trellis coded modulation [TCM] [78] schemes, can be used with coded modulation. Three types of signal constellations are referred to in the work: (1) An actual channel signal constellation is the constellation whose signals are actually transmitted over the channel. (2) An virtual channel signal constellation is a hypothetical signal constellation used in concatenated coded modulation schemes. It is, in fact, a block code which in turn uses an actual channel signal constellation. (3) The general signal constellation which may be actual or virtual is termed as arbitrary channel signal constellation. An arbitrary channel signal constellation may be a PSK, ASK, QAM, or any other signal constellation in which the number of signals can be any number (not necessarily a prime or a power of a prime number). A set-theoretic framework has been used for representation of signals from an arbitrary channel signal constellation and for sequences of signals of finite length. A block code of length n for BCM scheme is defined to be a subset of the set S' x S' x… x S'(n-times) of all sequences of length n of signals from an expanded channel signal constellation S'. For working with various Euclidean distances a matrix theoretic framework has been used. Euclidean distances between the signals of a channel signal constellation are represented in the form of a matrix. This matrix DS is symmetric and its structure reflects the structural symmetries of the channel signal constellation. Such symmetries may be utilized to reduce the storage requirements of Euclidean distances of signal constellations. The matrix DSturn is utilized to obtain a representation of the Euclidean distance between signal sequences. For a sequence of signals of lengths n from the finite set S' forming a channel signal constellation, the Euclidean distances between all the n-tuples are represented by the matrix dSxSx..xS(n-times)=dS(Ed)dS…(Ed)dS(n-times), Where (Ed) is a matrix operator which is structurally similar to the Kronecker product of matrices. The matrix dSxSx…xS(n-times) also provides the necessary information on the Euclidean distance distribution of sequences of signals of length n from the channel signal constellation. For obtaining block codes the two basic approaches which have been sued in the coding theory literature can be classified as follows: (1) The structured coded approach: In this approach structure is imposed on the code words. (2) The structured encoder approach: In this approach structure is imposed on the encoder which generates the code words.

For more details click here Back

Title : ***Estimation of Depth from Monocular Defocused Imager***
Author(s) : ***Rayala Jiendra Das***
Roll No : ***9020462***
Supervisor(s) : ***Mullick S K& Gupta Sumana***

Abstract

The depth information of a scene is a vital cue for the purpose of scene interpretation by any biological or machine vision system. In order for the visual system to be able to interact with the three dimensional world even when the objects in the scene are not known a priori, it must be able to determine the depth of the visible surface. This depth is defined as the distance between the surface of any object to the surface of the sensor through which the information about the physical world is obtained. In the context of machine vision, this involves estimating a matrix of distance values from a plane (usually that of the camera) to the surface of the objects in the scene using one or more images obtained either through a single camera or multiple cameras.In computer vision literature, various methods for estimating the depth have been suggested. Recently a technique which uses the phenomenon of defocusing has been introduced to determine depth. When an object is imaged using a conventional camera, all those points on the object which are not in focus, will appear as uniform circular disks there by creating a blurred image of the object rather than a sharp image. This uniform circular disk is usually known as point spread function.The basic idea behind the depth from defocus approach is that instead of removing the degradation in the image because of the defocus, we can exploit this to estimate the depth information. This is possible because the parameters of the point spread function depend both on the camera and the scene characteristics and the depth is in general related to it's spread. This relation between depth and the spread is derived using the geometrical optics approximation of light propagation. Assuming that the form of the point spread function is available beforehand, depth map is inferred by estimating the spread throughout the image.The depth from defocus approach is an emerging method which promises to become an integral part of machine vision applications. Many algorithms for estimating the depth information from defocus have been suggested. In this thesis we document some of the investigations that have been carried out by us on this problem. We wish to develop and study a unified methodology for the estimation of depth which would take into consideration some important aspects of the depth from defocus problem. In deriving the method we wish to utilize whatever properties and constraints that are present in the underlying system such that the model and the method of estimation of spread are tightly coupled.The form of the point spread function determines the definition of spread. This is generally assumed to be one of the point spread functions such as a pillbox or a Gaussian function. However in practical optical systems, due to diffraction effects, aberrations etc., the effective point spread function can become quite complicated. Usage of the above functions may result in inferior depth estimates because the definition of spread is intimately related to their form. If the point spread function of the optical system happens to be different because of other effects, then the estimate of spread is bound to get affected.In some of the methods, the calibrated point spread function from the camera has been used. Measuring the point spread function from the camera itself, though, in general provides better results, is quite complicated and the experiments have to be conducted with care. Therefore it would be desirable to have the

definition of the spread independent of the form of the point spread function.We first formulate the problem as that of finding an appropriate defocus operator. A local region image is a small region around the point where depth is to be estimated. The defocus operator is an operator which transforms the local region image with less defocus into the local region image with more defocus information. This expresses the problem in terms of identification of a pertinent linear system.' We then consider a general definition for the spread parameter and reformulate it such that it can be integrated seamlessly into the system identification framework. To compute the spread and obtain analytical expressions for the same, we represent the Fourier transform of the point spread function in terms of parametric functions such as polynomial or rational transfer functions.With the above representation, the problem now reduces to that of identifying the parameters of the above transfer functions from the observed input-output data, the input being the local region from the image with less defocus and the output being the local region from the image with large defocus. The problem of identification of parameters is essentially equivalent to finding the best approximation to the original transfer function by minimizing an appropriate cost function. The methods used for the identification of parameters in general utilize the auto and cross correlation information between the input and the output data. This usage of correlation information reduces the effect of any extraneous data present in the input or the output.Another aspect of the depth from defocus problem which we consider in this thesis is the synthesis of a given point spread function. In any depth from defocus algorithm the point spread function greatly influences it's performance. The factors which affect the performance are i) the form of the point spread function ii) the other degradations present in a camera. Our objective here is to explore how a given point spread function can be synthesized. Synthesizing a point spread function would enable us to 1) study the blur formation in an optical system more clearly, 2) obtain some control over the form and 3) possibly study the effect of other degradations on it.The investigations that are carried out in the thesis are organized as follows. In chapter two, we first formulate the two image depth estimation problem as that of finding an appropriate defocus operator. A general definition for spread is intro- duced and appropriate relations between various spread parameters are derived. The problem of depth estimation is then posed as identification of a pertinent system. The definition for spread is reformulated in terms of the Fourier transform of the point spread function. Non-causal parametric transfer functions to approximate the underlying transfer function are derived. Finally analytical expressions are derived for computing the spread from the parametric transfer functions.In chapter three, we present some algorithms for identifying the parameters of the polynomial and rational transfer functions. We first consider the case of rational transfer function and show that it leads to identifying the coefficients of an appropriate auto-regressive moving-average model. Two dimensional equation error algorithms to identify the coefficients of the non-causal auto-regressive moving-average model are developed. Then an algorithm for identifying the coefficients of the polynomial transfer function is derived as a special case of the algorithm for the rational transfer function.The methods and the algorithms developed in the previous chapters are applied on synthetic and real images acquired from a camera and the results are presented in chapter four. We also discuss various aspects of the algorithms such as computational complexity, problem of lack of texture, model order determination and stability of the transfer function. Fast implementation of the two-dimensional non-causal equation error algorithms using triangular decompositions is also discussed. Finally, pseudo code for the algorithms is presented.The issue of synthesizing any given point spread function is discussed in chapter five. Using scalar diffraction theory, we show how a given point spread function can be synthesized. As an example we consider the case of synthesizing a Gaussian function and show analytically how that can be achieved.

For more details click here Back

Title : ***Nonuniform Sampling Of Time-Limited Signals Using Time Scale Localisaiton Properties Of Wavelet Transfroms***

Author(s) : ***Ramesh Chaveli***
Roll No : ***9220462***
Supervisor(s) : ***Rao PRK & Sharma Govind***

Abstract

Sampling is an indispensable operation in signal processing and digital communications. The conventional approach of collecting samples of an analog signal is uniform sampling which collects samples at equal intervals. The uniform sampling approach is not always the best in terms of the number of samples required for the same reconstruction error, therefore, the possibility of collecting samples of an analog signal at nonuniform intervals has to be explored. Given any signal, what is the best way to take th e samples or at what intervals the samples are to be taken so that the whole process is efficient (requires least number of samples to represent an analog signal for a given error and a given reconstruction method), is the basic problem of nonuniform sampl ing. The objective of this thesis is to make some contribution towards the above described problem of nonuniform sampling. The uniform sampling is based on Shannon's sampling theorem. In uniform sampling the signal is sampled at a rate greater or equal to the twice the bandwidth of the signal. This technique is not suitable in the following cases - (a) when the signal is not band - limited. (b) When an estimate of the bandwidth of the signal is not available. It is for these cases that nonuniform sampling ca n serve as an alternative. The time limited signals are not band limited and so uniform sampling is not applicable in this case. Moreover, signals found in practice are always time limited, therefore the time limited class of signals is important and hence , in this thesis, nonuniform sampling techniques for time limited signals are developed. In this thesis, two nonuniform sampling problem for time limited signals have been formulated and algorithms which provide solutions to these problems are presented. Given any time limited signal, what is the nonuniform placement of a finite number of samples in the domain of the signal such that the fractional mean square reconstruction error is minimized? This is the first problem that is attempted in this thesis. Th e basic drawback of the type of nonuniform sampling strategy that results from solving this problem is, that it requires the storage of the sampling instants as well. Therefore, another nonuniform sampling problems is proposed. Given a time limited signal, what is the optimal distribution of a finite number of samples in the partitions of a time limited signal? The optimality criterion is to minimize the fractional mean square reconstruction error. In this problem, in each of the partitions the samples are placed uniformly. Further, the reconstruction error minimization has to be performed with respect to the partition lengths as well. It is more efficient in terms of the storage requirements, to sample uniformly in the different partitions at dissimilar rat es rather than sampling completely nonuniformly, as in the case of the first problem. Because, in this scheme only the instant of the first sample in each of the partition, the partition length and the number of samples in the partition have to be noted do wn. To solve both these problems, the excellent time scale localization properties of wavelet transform have been used. Therefore, in this thesis, before presenting the solution to these problems, the wavelet transforms are discussed; further a novel

conc ept termed as scale limitedness is introduced with reference to continuous wavelet transforms. The wavelet transform of a signal is the inner product between the signal and the function)(1 a bta −y Here a is the scale pa rameter, and the shift parameter, denoted by b, shifts the wavelet to t = b, therefore, the wavelet transform gives the behaviour of the signal at t = b. at lower scales the wavelet transform gives the fast varying component present in the signal. Therefor e, the wavelet transform is enhanced at lower scales, in regions, wherein, the signal is varying fast. This has been illustrated through simulations. The concept of scale limitedness, introduced in this thesis, can be used for quantifying signal variatio ns. A signal is called scale limited, if its wavelet transform is zero for all scales below a certain scale value and for all shifts. Another definition, termed as δ0 - practical scale - limitedness is also introduced. A signal is called δ0 - practical scale - limited to a l , if a l is the largest scale value below which the fraction of the energy, for all shifts, is less than δ0 scale - limits are computed. It can be inferred from these samples, that the signals which have faster varying component have smaller sca le limits. Further, eliminating fast varying components it is possible to construct signals which are scale limited. Next we discuss the relation between the higher order vanishing moment property and scale limitedness. Any real square integrable function qualifies as a wavelet if its zeroth moment vanishes; there exist wavelets for which the higher order moments vanish. It has been shown that, signals which are slower varying have relatively larger scale limits because of the vanishing moment property of the wavelet. A new theorems have been stated and proved next. It has been shown that , a signal which is band limited is also scale limited moreover, it has been proved that there exist time limited signals which are scale limited and obviously, not band l imited. Further it is shown that, signals which have smaller scale - limits have larger upper bound on their differentials. Next is shown that the Lagrangian interpolation function is δ0 - practical scale - limited. Finally it is shown that, signals which are i n the successive approximation spaces associated with the multiresolution analysis are scale limited. As an application of scale limitedness, a sampling rate estimation method for signals is proposed. For estimating the sampling rate, a theorem is proved, which states that signals with smaller scale limits should be sampled at a relatively faster rate for the same error bound. This method proceeds by, first over sampling the signal, then finding the scale limit using the multiresolution analysis. And final ly, given a bound on the error, reduction in the number of samples is found such that the error remains less than the bound. The method gives a sampling rate reduction of a multiple of 2 j , where j is an integer greater than or equal to zero. The first non uniform sampling problem is the optimal sample placement problem. Given any time limited signal, what is the nonuniform placement of a finite number of samples in the domain of the signal such that the fractional mean square reconstruction error is minimiz ed? The sampling algorithm developed here is based on the observation that if the error in the transform domain is bounded, then the error in the time domain is also bounded. Theorems have been presented which give the bounds on the fractional mean square error. It can be inferred from these theorems, that the bound on the fractional mean square error is more sensitive to the value of the fractional mean square error in the transform domain at lower scales as compared to that at higher scales. Therefore, to make the fractional mean square error smaller, the error in the transform domain should be made relatively smaller at lower scales. Another observation is that, the mean square error in the transform domain can be made small by placing the samples at thos e points in the transform domain, at which the wavelet transform maxima occur. The algorithm for solving this problem proceeds by first sampling the signal at the highest possible rate. Then the samples are picked in the time domain at a value denoted by t i , if the wavelet transform attains maximum in the transform domain at t i . This gives the initial set of samples, if al the maxima are exhausted and still samples can be taken then rest of the samples are taken in the neighbourhood of the samples in the i nitial set. Using the optimal samples the original set of samples is reconstructed using an interpolation method. It is assumed that the interpolation is done using local functions shifted to the sample value. Further, the interpolation method should be su ch that, at the known value of the signal the interpolated value should be same as the known value. Simulation results which demonstrate the algorithm have been presented. The second nonuniform sampling problem attempted in the thesis is the following - given a time limited signal, what is the optimal distribution of a finite number of samples in the partitions of a time limited signal such that the fractional mean square error is minimum? In each of the regions uniform sampling is done. The regions are obt ained by partitioning the signal depending upon local variations. In the sampling strategy, proposed here, the rate of sampling adapts to local variations. In the algorithm, first the signal is partitioned into dissimilar regions. Each of the regions is ch aracterized by a scale limit, which is distinct from the values of the scale limits of the neighbouring regions. Then, in each region the optimal number of samples are distributed which is a fraction of the total number of samples to be distributed. It has been proved in the thesis, that segments with smaller scale limits should be sampled at a higher sampling rate. In each of the regions the uniform sampling is done. Therefore the solution to the second problem leads to a multirate sampling strategy. The algorithm proposed for multirate sampling is as follows. First a low value of the threshold is set, then the optimal sample distribution is found; using this optimal distribution the signal is sampled uniformly in the different segments. Next these samples are used to reconstruct the signal back and finally the fractional mean square error is found. This is repeated for successively higher values of the threshold and that sample distribution is selected, for which the minimum fractional mean square error is obtained. The fractional mean square error versus the threshold value is found to have many local minima; therefore, the global minimum has to be selected. The simulation results also have been presented. The main conclusion is that, in order to kept the bound on the fractional mean square error low, the number of samples in any segment should be inversely proportional to a positive power of the scale limit. To summarise, in this thesis two nonuniform sampling problems have been formulated. Given any time limited signal what is the placement of a finite number of samples such that the fractional mean square error is minimum? It has been proved, that in order to minimize the fractional mean square reconstruction error, the samples should be placed at and in the neighbourhood of those points at which the wavelet transform maxima occur and also more samples should be placed using the information of the maxima at the lower scales. An algorithm for nonuniform sampling has also been presented. Given any time limi ted signal, how to partition the signal into regions of dissimilar variation and what is the rate of sampling in each region, when uniform sampling is done is done in each region? This is the second problem attempted in this thesis. This problem has been s olved by partitioning the signal into regions of dissimilar variation using the concept of scale limitedness introduced with reference to the continuous wavelet transforms. The signal have been partitioned into regions such that each region has a scale lim it which is different from the scale limits of the neighbouring regions. It has been established in this thesis, that the error bound is minimized when more samples fall in the regions of smaller scale limits. Conclusions based on the results obtained in the present work are given below - • In regions of time domain, wherein, the wavelet transform gets enhanced at lower scales , the signal is faster varying as compared to regions where signal is slower varying. • Signals which are faster varying have relativel y smaller scale limits. • Scale - limited signals satisfy the following properties - -

Signals which are band limited are also scale limited. - there exist time limited signals which are scale limited as well. - Signals having smaller scale limits have larger u pper bounds on their differentials. - The Lagrangian interpolating function is δ0 - practical scale - limited. - Signals which are in successive approximation spaces associated with multiiresolution analysis are scale limited. • It has been shown that, for the sa me error performance, the signals with smaller scale limits should be sampled at a higher rate. • For placing the samples nonuniformly, in order to keep the mean square error low, the strategy of sampling is to place the samples at and in the neighbourhood o f the maxima of the wavelet transform. • For design based sampling, when samples are placed uniformly in each of th

For more details click here Back

Title : *Application Of Converter Based Controllers To Electric Power Trnasmission System*
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Abstract

Addition of new transmission lines to meet the ever growing need of electric energy is being constrained due to restricted right of way and other environmental reasons. Thus, the current focus is one developing suitable technologies which could not only pe rmit greater flexibility and control in the operation of transmission network but could also lead to higher power transfer over the existing transmission corridor. The advent of Flexible AC Transmission Systems (FACTS) is a step in the direction where high power semiconductor switches are expected to play a crucial role. In addition to HVDC and SVC installations, variety of converter circuits are being actively considered for applications such as bus voltage control, phase angle regulation and line impedanc e control. To assess the suitability of a converter based device in any of the above application involves a detail investigation of the system behavior through dynamic digital simulation. This requires system representation to the desired degree of detail . Modeling of converter circuit in such situation has always posed a major problem due to varying topology of the converter circuit and the complex configuration of the converter values. The need, therefore, is to develop a generalized approach for convert er modeling which could handle a variety of converter topology suitable for different applications. For bus voltage control application, the use of thyristor based State Var Compensator (SVC) is well established. Efforts are continuously being made to evo lve new configuration which could circumvent the basic deficiencies of a traditional SVC like size etc., while at the same time enhancing the system performance. In this context advanced State Var Compensator (ASVC) or Static Synchronous Compensator (STATC OM) has been proposed in the literature [1]. In its basic form, an ASVC consists of GTO based voltage source inverter (VSI) which is connected to an ac bus through a coupling transformer. The VSI is usually supplied from a dc storage capacitor. Being an al l electronic device, an ASVC has a much reduced size compared to an SVC. Furthermore, the use of GTOs permits flexibility of control through variety of switching strategies enabling better dynamic profile for voltage control. The effectiveness of voltage control provided by ASVC largely depends on its value switching stratrary. While only multi step (24 step and above) converter switching have been proposed in the literature, there seems to be a scope for using alternative switching strategies which would allow more than one degree of freedom in control. Switching strategy based on Pulse Width Modulation (PWM) could be one such potential candidate which can offer better control features. However, with the present day GTO technology, a single converter based PWM

ASVC is not suitable for high power application as the output voltage generated by such a device is rich in harmonic content arising due to switching speed limitations of GTOs. Thus, a multi bridge PWM structure may be an alternative for using PWM con trol stratrary for high power application. While control of line impedance is being used for enhancing the power transfer over a transmission corridor, the same could be accomplished by enhancing the transient stability limit. A synchronous generator tends to accelerate when a fault occurs in a network to which it is connected. It is thus of paramount importance to restrict the load angle (δ) below 30 o - 35 o to secure stable operation. This restricts the power transfer through the network. A thyristor control led dynamic brake, when connected at the generator terminal, can arrest this acceleration by dissipating some real power when a fault occurs. Consequently, the machine can operate at a higher steady state angle δ supplying more power. However, the effectiv eness of a dynamic brake depends upon the control strategy adopted for switching of the brake. Different control strategies, such as bang bang control, feedback control, variable structure control schemes have been reported in the literature. These control rules involve complex mathematical modeling and also measurement of the quantities other than the local one. In this regard, as the dynamic brake has no dynamics of its own, a simple control rule based on the local measurements without the need of compl ex mathematical technique can be formulated. Based on the above considerations the objective of the thesis are : 1. Development of an improved generalized model of Converter Bridge for electromagnetic transient analysis. The model should be capable of handl ing and complexity in the value representation and independent of any value switching strategy. 2. investigation of appropriate configuration of ASVC for power transmission system application. 3. to evolve a simple control law for dynamic brake application to po wer transmission system. The major contributions of this thesis are: 1. development of a generalized modeling technique for representing a bridge converter circuit based on state space analysis and graph theoretic approach for the purpose of electromagnetic transient analysis. A modular approach has been adopted in developing the converter system model and consequently, any complex value configuration in addition to the normal HVDC value configuration can be handled in a realistic manner. Also, no restrictio n on the configuration of the elements to be connected at the output side of the converter is imposed. Thus, this can be conveniently used for simulation of converter systems which are employed in the different FACTS devices like ASVC, UPFC etc. it can als o be easily extended to represent a multi bridge converter system. The proposed model can also bee easily interfaced to an existing electromagnetic transient simulation program like PSCAD/EMTDC. 2. determination of a suitable configuration of an ASVC and its switching technique for power transmission system application. The limitation in the switching frequency of the devices (GTO, thyristor etc.) for high power application and the problem of harmonic generation by an ASVC have been taken into due consideratio n in the investigation of the suitable configuration. Control strategies for effective operation of an ASVC for dynamic reactive power support have also been suggested. 3. development of a simple rule based control strategy for dynamic brake application. The control strategy is completely based on the local measurements available at the generator terminals. A brief description of the work reported in the thesis is given below: Chapter 1 discusses the general introduction in FACTS and reviews the relevant lite rature. In Chapter 2, a generalized modeling approach of a bridge converter system for electromagnetic transient analysis is developed. Based on the graph theory, the guiding equations of the model are developed in the state space framework. The modeling a pproach is flexible enough to handle any degree of complexity in the value representation as well as any configuration of the element connected at the output side of the converter. As a result, the proposed modeling technique can be used to represent a tra ditional HVDC converter as well as any converter based FACTS device. The initial validation of the developed model is carried out through simulation of 6 pulse HVDC rectifier and inverter circuits. To use the proposed converter model to simulate any pract ical power system, the proposed model must be interfaced with an existing electromagnetic transient program. It is due to the fact that in such an approach, the models for different power system components that are already available in the existing program can be used along with the proposed model to represent the entire system, thus rendering complete flexibility in the simulation approach. The interface of the proposed converter model with an existing electromagnetic transient analysis program (PSCAD/EMTD C) is described in Chapter 3. Exhaustive validation of the proposed model and the interfacing technique is carried out through digital simulation of a 6 pulse two terminal HVDC system and the HVDC CIGRE benchmark (12 - pulse) model. The results are also comp ared with the results obtained through the digital simulation of the above two systems using the PSCAD/EMTDC converter model. Chapter 4 starts with the application of the proposed converter model for representing a single bridge PWM ASVC connected to an a c system through coupling transformer. For comparison, the same system has been simulated using the PSCAD/EMTDC software package. The harmonic spectra of the ASVC output voltage obtained by using the proposed model with the PSCAD/EMTDC software were compar ed with theoretical harmonic spectra calculation. Voltage control capability of a single bridge PWM ASVC is studied by applying it to a low power distribution system. For high power transmission system application, four different configuration of ASVC are considered -I) single bridge SPWM ASVC, ii) single bridge ASVC employing Selective Harmonic Elimination switching scheme, iii) multi bridge PWM ASVC. These configurations are studied in detail to assess suitability for reactive power compensation form harm onic content point of view taking into account the limitation in the switching frequency of the high power semiconductor devices (GTO, thyristor etc). In Chapter 5, the two multi bridge configurations of ASVCs considered in the previous chapter viz, multi pulse ASVC and multi PWM ASVC are applied in the mid point of a single machine infinite bus (SMIB) system for dynamic reactive power compensation. Appropriate control strategies for both the ASVC configurations for the purpose of dynamic reactive power co mpensation is discussed. Transient performance of these two ASVCs are investigated through faults studies. The faults are applied at different locations at generator power transfer levels close to the thermal limit of the transmission line. The study of t wo different thyristor based dynamic brake configurations in a SMIB environment for transient stability enhancement is reported in Chapter 6. A simple rule based control strategy, based on the local measurements only has been proposed. The performance of t he dynamic brake with the proposed control strategy has been examined in detail through fault studies at two different very high power transfer levels. Chapter 7 outlines the conclusions drawn form the thesis and suggests some future scope of work.

For more details click here Back

Title : ***Morphological Processing Of Random Finite - Gray - Scale Digital Images***

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Abstract

The objective of the thesis is to develop a theoretical framework for morih'~v>7i'vJ processing of random finite-gray-scale digital images. The work involves selection and study of an appropriate algebraic structure for finite-gray-scale morphology and showing how it fits naturally within a stochastic setting. Certain issues relevant to statistical analysis, modeling, filtering, and estimation of digital images using the morphological approach are also addressed. The study reported here is important for two reasons. First, random images arise naturally in a variety of applications wherein one is interested in the spatial structure in the scene. This may be important from the point of view of making certain mea surements or for the purposes of identifying, locating or recognizing regions/objects of interest. For such cases, a stochastic view for the particular image realization under consideration is adopted, i.e., it is assumed that there are underlying characteristics of the scene that produce many similar images. A random image may therefore be regarded as a realization of some stochastic process. Secondly, finite-gray-scale dig ital images are currently being used for majority of image processing, analysis, and synthesis applications, either off-line on digital computers or through the use of VLSI digital processors in real-time imaging systems. Given the rate of advancement in VLSI technology and digital computing devices, this situation is likely to continue in the foreseeable future. An appropriate framework for a quantitative analysis of the geometric content of random images is provided by random set theory, which is intimately related to mathematical morphology. Since its inception, continuous-domain random set the ory has progressed considerably and the basic concepts of the underlying theory are now quite well established. However, despite the profound accomplishments on the theoretical front, random set theory has not emerged as a viable and advantageous alternative to existing statistical image processing techniques. This is because there arc inherent analytical complexities and technical limitations which restrict the use-Synopsis v fulness of continuous-domain random set theory. It lacks the necessary tools from a statistical perspective. This is due to the absence of variety of tractable random set models and non-existence of Maximum Likelihood (ML) and Maximum-A-Posteriori (MAP) methods for statistical inference and estimation. Recently, a number of investigations on discrete random sets (DRS) defined over finite and discrete domain sets have renewed the interest in random set theory. These investigations have addressed issues related to statistical inference, estimation, model ing, filtering, and analysis of random digital binary images by morphological methods within the DRS framework. For the gray-scale case, there have appeared in litera ture, a number of articles reporting results on statistical analysis of morphological operators and statistical optimization of morphological filters. However, these have not been posed and solved within a unified statistical framework, as is the case with binary images. This thesis is concerned with the development of a theoretical frame work of Discrete Random Functions (DRF) within which one can address problems related to morphological processing of random finite-gray-scale digital images. The main contributions of the thesis are described in the following. The recently

proposed complete lattice ordered commutative monoid (c/c-monoid) structure for morphology is presented from the viewpoint of its appropriateness for the objective of this thesis. A number of theoretical results concerning the alge braic properties of mathematical morphology are proved within this framework. It is shown that well known results related to increasing property, associativity, distribu-tivity, etc. of morphological operators such as dilation, erosion and their combinations hold under the c/omonoid structure. Various algebraic paradigms of fuzzy morphol ogy, computational morphology, matrix morphology, and lattice transforms of image algebra are interpreted within this algebraic structure. These also demonstrate the broad nature of the selected framework as it is capable of incorporating a wide variety of diverse problems in a single unifying setup. Subsequently, the concept of decimation as applied to morphological operators is used to develop a new class of non-linear transformations termed as Decimated Morphological Filters (DMF). Depending on the use of a specific morphological op erator in the proposed decimated filter structure, one can obtain various DMF's, the algebraic properties of which are studied. It is shown how decimated morphological operators which are shift-variant and/or non-idempotent are obtained from standard shift-invariant and/or idempotent morphological operators. Consequences of using such DMF for noise removal applications are studied via simulations and their ad-Synopsis vi vantages as regards image feature preservation are illustrated. The axiomatic theory of discrete random functions (DRF) defined over discrete and finite range and domain sets is developed in analogy to the discrete random set (DRS) theory. The fundamental functionals of the DRF theory, namely the Discrete Capacity Functional (DCF) and the Discrete Generating Functional (DGF)5 are de fined and their properties derived. An important result is obtained which states that DGF of a DRF completely determines the probability measure over the set of real izations of DRF. Further, it is shown that DRF can also be studied by considering their thresholded level sets. This considerably reduces the number of test functions required for the probabilistic specification of the DRF. Issues related to stationarity, statistical independence, and moments of DRF are discussed next. Random finite-gray-scale digital images are modeled as realizations of DRF and tools for the analysis of statistical aspects of morphological transformations on DRF are described. Specifically, relationships between the basic characterizing functionals of DRF and its morphologically transformed DRF are derived. Application of these results for statistical analysis of finite-gray-scale images is demonstrated through sim ulations. Finally, Discrete Boolean Random Function (DBRF) and Discrete Radial Boolean Random Function (DRBRF) models of the germ-grain DRF type construc tion arc defined and their characterizing functionals obtained. Representative sample realizations of DRBRF morphological model of DRF are also included. The problem of estimating realizations of DRF corrupted by noise of the supre-mum/ infimum kind is considered next. It is proved that morphological operators such as opening, closing, supremum of openings, and infimum of closings are opti mal Maximum-A-Posteriori (MAP) estimators. These results are obtained under an appropriate and minimal set of assumptions relating to the structural and statistical constraints on image DRF and noise DRF. Noise is assumed to be independent and identically distributed (iid) pixelwise for single and multiframe observation scenarios. For the case of multiple frame observations, the sufficient statistics for the estimation problem is obtained and strong consistency of the resulting optimal MAP estimator is also proved. Next, the assumption of iid noise is relaxed and the MAP optimality and strong consistency of morphological filters for filtering image DRF's degraded by morphologically smooth noise is proved. Simulations on actual image data are carried out in support of the validity of these theoretical results. The statistical optimization of morphological filters is posed within the rigorous statistical framework of discrete random functions. The mean absolute error (MAE) Synopsis vn is reasoned out to be appropriate as a distance measure between the original and the estimated image for the filter optimization purposes. The optimal filter that yields minimum MAE is characterized in terms of the discrete generating functional of the image DRF and the noise DRF under the assumption of statistical independence of image and noise. The MAE of estimation by the morphological filter is represented in terms of MAE incurred by each of the basis filter elements. Recursive expressions can be used to compute the MAE for an n element basis filter in terms of (n — 1) element basis filters. Examples illustrating the application of these techniques for designing the well known annular opening/closing operator and morphological edge detector are considered. Finally, simulation experiments to design morphological filters and decimated morphological filters are detailed out along with conclusions drawn from the results.

For more details click here Back

Title **:** ***Analysis Of Strip Excited Waveguide Slot***
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Abstract

Wave guide fed slot arrays are extensively used in many radar and microwave communication systems. Though there exist many types of slots which could possibly by used as radiating elements in an array, the longitudinal broad wall slot and the inclined narr ow wall slot have received extensive consideration. Two other slots, namely the centered longitudinal slot and the vertical narrow wall non - wrap - around slot are recognized as ideal candidates to build an array due to their properties of having no cross -polarization and the ability to build a true linear array. The vertical narrow wall waveguide slot in particular is preferred, specifically in all applications where scanning in E -plane is desired. There have been a number of attempts at making the narrow wal l waveguide slot radiate, by attaching metallic wires in the neighborhood of the slot. Specifically, this technique makes use of either a bent screw or a pair of tilled wires, attached to the edges of the slot to achieve the necessary coupling from the inc ident wave. This thesis introduces a fundamentally different method of exciting the vertical narrow wall slot. It makes use of a thin printed strip bebind the slot to excite it. The strip can either be inclined with respect to the slot or can take up an L shape. The slot, along with the exciting strip, can be printed simultaneously on a dielectric substrate and can replace the narrow wall of the waveguide. This method possesses all the advantages of the printed circuit technique and therefore can achieve g ood accuracy and square ended slots, which are generally hard to achieve by the conventional slot machining process. The coupling to the slot can be adjusted predominantly by varying the strip inclination with respect to the slot. There exist several possi ble parameters in the structure which could be made use of in achieving the desired excitation level. For the analysis, this structure is modeled as a coupled integral equation and is solved by the Method of Moments (MoM). The integral equations are set u p by making use of the Green's functions of an in homogeneously filled waveguide. The required Green's functions are derived using the standard technique of normal mode expansions. The analysis of this structure via. MoM turns out to be extremely involved due to the e - γ z -z ' dependence of the waveguide Green's functions. This, together with the inclination of the strip along the axis of the waveguide, makes all the source and field domain variables coupled, resulting in serious implementation problems in setting up the MoM matrix. This thesis presents a systematic quadrature scheme for making this problem tractable and computationally efficient. The thesis also introduces algorithms for solving the eigenvalues of LSE and LSM normal modes in an in homogeneously filled wav eguide. Among the Green's functions made use of in the analysis, it is found that, the Green's function corresponding to the axial E field due to an axial electric current is a true distribution, due to the

presence of the Dirac delta function . Method of computing the E z field, in cases where one on the source domain quadratures are dropped, leads to divergence in the series representing the E z filed. This problem is identified as one involving the correct interpretation of distributions, and is solved by representing the singularity as an orthonormalized LSM series. Since the LSM eigenfunctions are not orthonormal to begin with, they are orthonormalized using Gram - Schmidt procedure has been developed. This result will have implications in cases where an ax ial radiating source in an in homogeneously filled waveguide is considered. A program is developed which computers the S parameters equivalent circuit, slot voltage and higher order mode scattering coefficients etc. Extensive parametric studies are condu cted by varying strip offset, strip angle, strip length, slot width, substrate thickness and substrate permittivity. In order to validate the theoretical model, experiments are carried out at X -band. Waveguides with a part of the narrow wall machined off a re used to fabricate the test pieces. A dielectric substrate, printed with the slot and the exciting strip on either side, is fitted into the waveguide in the region where the narrow wall is machined off. Through Reflect Line (TRL) calibration is used to e stablish the measurement plane at the center of the slot. The necessary TRL calibration standards are fabricated in the laboratory. The measurements are done on a HP8510C vector network analyzer. Experiments are also conducted to ascertain the effect of the ground plane. The present study on the strip excited waveguide slot indicates the suitability of this structure in building planar arrays working at Ku - band and beyond. It is possible to build a planar array, by attaching, a printed dielectric subst rate consisting of slots and exciting strips to a machined waveguide channel. An array of this type has all the slots along the array axis and does not suffer from the problem of cross - polarization.

For more details click here Back

Title **:** ***Temperature Effects On The Performance Of MOS Devices***
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Abstract

Metal - oxide semiconductor field effect transistor (MOSFET) is the most important micro electronic device for memories and logic, and is a building block of very large scale integration (VLSI) circuits because of its high packing density, low power consumpt ion and high yield. N - channel MOSFET (OR NMOS) has received importance in microprocessor and memory chips largely due to its low cost and density advantages. It has also demonstrated circuit speeds as good as any other high density technology. The basic t heory MOSFET operation was established in early sixties and extensive studies on devices were taken up in the subsequent years. The newly acquired knowledge was utilized by the industry to produce digital IC's and the need immediately arose for the develop ment of efficient models for computer aided design and simulation. The efforts in this direction were made quite successfully for long channel devices. However, accurate modeling of short channel and small geometry devices evoke little interest as it neede d the numerical solutions of two or three dimensional Poisson and current continuity equations, and was very time consuming. But, circuit design, simulation and process characterization prefer simple analytical models even for short channel devices. An eff ort has therefore been made in the present work to develop simple and accurate models with as few parameters as possible. Several applications of MOSFET require its operation over a wide temperature range (200 - 525 K). The advantages offered by the low tem perature operation of MOS transistor have made it a subject of current interest at device, circuit and system level. Some industries dealing with well logging instruments, jet and automotive engine, rocket technology, etc. are greatly in need of reliable e lectronic equipment which can work under harsh climatic conditions. At some remote places where sufficient cooling may not be possible for the electronic circuitry and the equipment has to withstand substantial ambient temperatures, MOS devices have shown significant advantages in these applications. Although functionality at high temperatures is commonly observed for discrete silicon devices, there is no clear cut procedure for prediction of their terminal characteristics. The temperature dependent study o f MOS devices is therefore important as it determine the sensitivity of the parameters and enables optimization of process variables to obtain ultimate level of performance. For instance bias point (or V GS) can be set such that the circuit is least sensiti ves. The information pertaining to temperature dependence is vital and must be known to a circuit designer in advance. This enables development of circuits which can function reliably and accurately in temperature variable environment. The temperature de pendence of MOSFET characteristics has been investigated

extensively in the past and several models developed for them. However, these are mainly limited to low temperature operation (i. e, below 300K) and relatively less work has been reported on the effe ct of high temperatures on MOS devices. Moreover, some of the temperature models have assumed a simple structure fabricated on a uniformly doped substrate while many others are based on empirical parameters fitting with inadequate emphasis on device physic s. Thus, there is a need for a more systematic work on the effects of high temperature on long and short channel MOS transistors. The present work is an effort in the direction where the temperature dependence of MOSFET parameters have been studied in deta il in the temperature range 291 - 473K. The investigations have covered the long channel devices with and without ion implantation and short channel ion implanted devices. The temperature dependent parameters that have been identified to affect the MOSFET ch aracteristics are: (a) the threshold voltage, (b) the carrier mobility in the channel, and (c) the leakage current through the drain/substrate junction. Models have been developed to describe the temperature dependence of threshold voltage and channel mobi lity and then used to generate the I P - V GS characteristics for long and short channel devices of known geometries, process parameters and device constants. The I P -V GS characteristics of long channel devices at T ~291K and that having W/L =50 µm /50 µm at hig her temperature have also been generated. The results of simulations are compared with the measured characteristics at different temperatures to verify the accuracy of the models. Chapter 1 gives the motivation for choosing the present problems, importanc e of the work and the objectives. A brief description of the historical development of MOS devices has been included with major areas of their applications. The temperature dependent parameters that effect the characteristics of MOSFETs are outlined. Also, the need for development of real models useful for carrying out simulation in MOS circuit and devices over a wide temperature range is emphasized Chapter – 2 provides a brief description of MOS structure. Basic theory of MOSFET has been reviewed and a gen eral expression for I - V characteristics is worked out. Subsequently simplified expression for the drain current in linear and saturation regions of MOSFET operation under strong inversion condition have been derived. The concept of channel length modulatio n has been introduced to account for the drain current in the saturation region. Also, I - V characteristics for short channel devices are discussed in the light of velocity saturation effects. Finally, a section is exclusively devoted to the device fabricat ion, I - V measurements above 290K and procedures used for extraction of MOSFET parameters. Chapter – 3 describes the studies on the threshold voltage. In the beginning a brief account of the commonly used definitions of the threshold voltage (V T) and its imp ortance in modeling of MOS devices are presented. Temperature effects on the surface potential (ª s) and the threshold voltage are discussed at length for long and short channel devices formed on both the uniformly doped and ion implanted substrates. It includes description on the determination of the threshold voltage by linear extrapolation of the I P - V GS characteristics of a number of devices having oxide thickness 50 µm, channel lengths 3 µm, 50 µm, jand 128 µm and doping (1 - 2.1 x10 16 cm – 3). The measured values are shown to be higher than the estimated (V T ’s) using the conventional definition that the onset of strong inversion is characterized by a surface potential of (2 φb + V SB) at the source where φb is the bulk Fermi potential and V SB is the source substrate bias. It is pointed out that the condition (2 φb + V SB) actually signifies the end of weak an d beginning of moderate inversion and therefore underestimates the surface potential. The discrepancy has been resolved by introducing a more appropriate value of surface potential as (2.2 φb + V SB) for strong inversion to occur. This is shown to predict to predict the threshold voltage accurately for long n - channel MOSFETs over a wide range of temperature and substrate bias without the need of any empirical fitting parameters. In case of short channel devices, a charge reduction factor has been introduced a ppropriately in the expression of the threshold voltage while maintaining the new condition of strong inversion. The resulting model is shown to yield threshold voltage in good agreement with the measured data. Modeling of channel mobility forms the subje ct matter of Chapter – 4. The effective mobility has been determined at various temperatures from measurements of drain current at low values of drain to source voltage (V DS) in a number often and short channel devices. It is shown that through channel mobi lity is technology dependent up tom some extent, correlation of the measured results is still possible in terms of relevant scattering mechanisms to obtain the nature of its dependence on the transverse electric field and the temperature. An empirical mode l has been developed which takes into account the dependence of channel mobility μeff on the gate electric field and temperature and has the form μμeff - μo T - m with σ (T) = σo T - n where μo m, σo and n are fitting parameters. In yet another but physically base d semi empirical model developed the individual contribution of various scattering mechanisms is accounted for by their unique dependence on the electric field and temperature. It is shown that the dominant mechanisms responsible for mobility degradation a t high temperatures are the surface and bulk phonon - and the surface - roughness scatterings. The surface phonon limited mobility varies as T - V x−effE With V ~1.85 and ε~0.3. The models developed for threshold voltage and channel mobility have been utilized in Chapter – 5 to simulate the I P - V GS and I P - V DS characteristics of various devices above 290 K. the simulation results are shown to be in excellent agreement with the experimentally measured characteristics justifying the validity of the mod els and their utility in device and circuit simulation above room temperatures. Moreover, sub threshold and p - n junction (drain to substrate) reverse leakage current measurements have been discussed with the aim to determine the highest temperature of sati sfactory operation of devices. The slope of In (I D) versus gate to source voltage V GS in the sub threshold region is shown to decrease continuously with increasing temperature. At a sufficiently high temperature (e.g., ~ 470 K for a device with channel do ping of ~ 10 16 cm – 3) when slope becomes very small, the sub threshold region disappears, i.e., there exists no gate voltage below which the device can be effectively turned off. Further, it is pointed out that the drain/substrate p - n junction reverse cur rent has two components involving the carrier generation in the depletion region and diffusion of minority carriers form neutral region. While, the former increases linearly with n, (T) and dominates at low temperatures, the latter exhibits)(2 Tn i dependence and becomes important as the temperature increase. Further, the devices with doping concentration of ~ 10 16 cm – 3 are shown to operate satisfactorily upto a temperature (~469 K) at which the magnitude of the reverse leakage current becom es comparable to the bias drain current. Finally, chapter – 6 gives a brief account of the conclusions drawn.

For more details click here Back

Title : *Array Of High Directivity Endfire Elements-A Study*
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Abstract

Printed planar arrays using microstrip or stripling feed is very attractive at highermicrowave because of low profile low cast ease of fabrication light weight etc. As thefrequency of operation increase the physical space between the elements decreases butthe width of the feeder line for specified characteristic impedance remains same.Therefore the space available for the feeder network is reduced .Due to the decreasedspace between the elements the interaction of the feeding lines with the radiatingelements and the coupling between the feeding lines to be taken into account whiledesigning the antenna array which is very difficult to compute. In a practical array it isnecessary to make moiré space for the feeder lines to minimize the coupling effect, whichessentially involves increasing the spacing between elements. This results in a loweraperture efficiency and appearance of grating lobes when the spacing is greater thanabout a wavelength. In this thesis arrays with inter-element spacing greater than onewavelength are considered. This reduces the number of elements in the array for a give aswell as creates enough room for the feed lines. To compenasate for the decrease inaperture efficiency High Directivity Endfire elements are used as array element and thegrating lobes are suppressed via the design of appropriate element pattern. Front this isnot a major disadvantage in the frequency band of interest, which is Ku Band beyond. High Directivity Endfire (HiDE) element is an endfire antenna with a speciallyshaped radiation pattern. Hide element can be an array of specially faced dipolesradiating in the pattern. HiDE element can be an array of serially fed dipoles radiating inthe endfire direction Yagi-Uda array dielectric rod antenna etc .A theoretical study iscarried out to ascertain the flexibility available to shape the radiation pattern of theseelements. Theoretical models are developed to study the behaviour of Yagi-Uda arrayand dipole array. The flexibility available in the dipole array can be studied byconsidering an array of isotropic radiators. The Yagi-Uda array is assumed to be made ofcylindrical radiators. Solving Pocklington's integral equation using the method ofmoments carries out the analysis of Yagi-Uda array. In the case of dielectric rod radiatorpublished results have been used to demonstrate its capability to from the HiDE element. An algorithm is presented to design the array of HiDE elements. The goal ofthe algorithm is to meet the specifications with minimum number of elements andachieved iteratively. The design begins by choosing an array factor. Since the elementpattern has high directivity in the present case the final pattern, which is a product ofelement pattern, and the array factor will not be optimum if which is a product of elementpattern and the array factor will not be optimum if such a choice is made. Therefore thearray factor is modified taking into consideration the element pattern so that the finalpattern approaches the optimum pattern. Arrays of HiDE elements with differentspecifications are

designed with various HiDE elements. A compression is also madewith a half wavelength speared array excited by optimum distribution and it is found thatin all the cases studied radiation pattern of HiDE array is slightly inferior to optimumpattern. Theoretical results indicate the possibility of suppressing the grating lobe with aproper choice of array factor and element pattern. This is demonstrated experimentally byconstructing a linear array of HiDE elements operating in Ku band. The HiDE element isarray of dipoles kept in front of reflector and radiating in the endfire direction. Thedipoles are printed on a thin dielectric subtracts and fed by a two strip transmission line.Various practical issues involved in the d4esign of the HiDE element are also discussedin the thesis. The HiDE ELEMENTS. Radiation characteristics of the HiDE element aswell as array of HiDE elements are measured and are also compared with the theoreticalresults. At higher frequencies the analysis of dipoles with simple feed models does notyield accurate results. Therefore the design data for the dipole are experimentallygenerated .The problems due to reflections at the discontinuities are eliminated byfollowing TRL calibration procedure. The required TRL calibration standard is fabricatedin the laboratory. Fitting polynomials to the measured data generates the design data fordipoles and matching elements. The radiation characteristics of the HiDE elements and the array of HiDE elementsare measured in an anechoic chamber. An anechoic chamber is constructed in thelaboratory for this purpose. Details of the anechoic chamber and related instrumentationare also presented in the thesis. The present study on the High Directivity Elements indicates that it is possible toincrease the inter-element spacing beyond a wavelength and still meet the patternrequirements. However it is found that the radiation pattern of the array of HiDEelements did not elements did not exactly match with the optimum pattern. It is alsofound that using this technique about 2to4 times reduction in the number of elements ispossible for a given antenna size very little degradation in the radiation pattern. The studyalso shows that at higher frequencies it is advantageous to use the space available in frontof the antenna by about one or two wavelength resulting in an antenna thicker than planarconstruction by a small amount. It is found that the endfire antenna design procedurespublished in the literature are directed are towards increasing the gain of the antenna.However the requirement of a HiDE element is an appropriately shaped pattern.Therefore while choosing a HiDE element configuration for a give array application it isnecessary to carry out investigations on endfire element to identify the flexibilityavailable to shape the radiation pattern. To demonstrate the feasibility of HiDE element array a linear array of HiDEelements was designed fabricated and tested. However the real application will be inplanner arrays were the reduction in the number of elements would be square of thereduction obtained for a linear array. This will significantly simplify the feed network.The trade-off is a little more complicated HiDE element and some degradation in thearray pattern. The advantages of this technique of building an array appears to outweighthe loss of pattern performance at lest at higher microwave frequencies.

For more details click here Back

Title : *Vertical And Horizontal WIRE Antennas Above A Lossy Earth*

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Abstract

The study of electromagnetic radiation due to antennas in the presence of a two layered region, like air and earth (or sea), or a three layered region like air and dielectric coated conducting medium, has many applications in communications, radar, geophysical exploration remote sensing etc. the antennas used either individually or combined into directive arrays, mainly take the form of cylindrical antennas including traveling-wave structures strip antennas or loops. The circuit properties (current distribution and driving point impedance) and the field properties (both near and far fields) of these antennas are significantly modified by the presence of the neighboring material media. The investigations in this subject may be separated into two groups. The first group is aimed at the determination of the radiated fields of an infinitesimal electric or magnetic dipole kept above or in a layered region, and to understand the electromagnetic wave propagation in such an environment. The second group deals with the analysis of finite antennas or scatterers in the presence of a layered medium. Following the pioneering work of Sommerfeld, the electric and magnetic field components of vertical and horizontal electric and magnetic dipoles in the presence of a layered medium can be derived in terms of the derivatives of Hertz potential functions. Alternatiely, they can be derived directly from Maxwell's equations without using the intermediate potentials. The field components contain infinite integrals, often called Sommerfeld integrals, which in general cannot be expressed in analytic form. In the numerical computations, they are slowly convergent because of the oscillatory behviour of their integrands. Initial studies to evaluate the integrals corresponding to two and three-media problems, have made use of analytical techniques where as, numerical methods came into use after 1970's. Analytical solutions were derived for certain operating conditions (bounds on the parameters and distances involved), by integrating along a contour in the complex plane utilizing saddle point techniques and by taking account of the singularities encountered during the shift of the path of integration. Early important comprehensive contributions are documented in the monographs by Banos [1] and Wait [2]. With the advent of high speed computers, attention was also directed at the numerical evaluation of the integrals, mainly those associated with the Hertz potentials. Various numerical methods reported in the literature can be placed into three categories: (a) numerical integration along the real axis, (b) numerical integrations in the complex plane, and (c) numerical integration based on integrand approximation. Computer based numerical calculations are particularly used in the accurate evaluations of the fields close to the antenna surface, which are required in the integral equations solutions of the current distribution along the antenna. On the other hand, analytical formulas when they are accurate, are equally

useful as they provide an insight into the understanding of the complicated fields in a simple manner. Significant contributions, aimed at obtaining simple approximate but accurate formulas for the field components due to elementary dipoles in the presence of a lossy earth have been made notably by A.Banos [1], J.R. Wait and co-workers [2, 3], P.R.Bannister [4], and R.W.P.King and co-workers [5]. Depending on the type of application and hence the possible ranges of the parameters involved two important approximations have been usually made in the evaluation of the complex integrals. (a) At low frequencies the wave number in air (Region 2), K2, is nearly equal to zero and may be neglected. Therefore, the spectral number in Region 2 in the exact integral is made equal to the integration variable. This approximation known as the quasi-static approximation is valid at close observation distances in terms of the free-space wavelength. The solution so obtained, can also be interpreted as the field from an image source in the complex space. Many papers have been published concerning the complex image theory applied, to half-space as well as multi-layered configurations, and at high frequencies (b) when |k1| >> k2, the spectral number in Region 1 is made equal to the wave number in Region 1 in the evaluation procedure, as significant contributions to the integral are assumed to occur when the integration variable is of the order of k2. Extensive series of paper by king and co-workers utilize this approximation. One objective of the present work is concerned with the approximate formulas developed by Bannister, and King for the electric and magnetic fields of vertical and horizontal electric dipoles in the presence of a conducting half-space. The validity of these formulas is tested by making comparisons of the exact integrals, over a wide range of parameters. Simples formulas for the fields in Region 1 due to the dipoles in Region 2 are obtained and their validity is also checked. The properties of the electromagnetic fields in both the regions are summarized using the valid simple formulas. The analysis of finite vertical and horizontal wire antennas located above a lossy half-space has been reported using both approximate (variational) and the more rigorous methods of moments (MoM) solution of the corresponding integral equation. Though the circuit properties of wire antennas, mainly the input impedance results, have been reported for some typical cases (for example, [6-10], their variations when the antennas are kept very close to the boundary of the earth, are not discussed in detail. Moreover, the mutual impedance between two coupled vertical or horizontal wire antennas kept above a lossy half-space has not been studied. In this work a general purpose sinusoidal pulse MoM solution for vertical and horizontal wire antennas kept above a lossy earth is developed. The Sommerfeld-type integrals encountered in the analysis are evaluated using an accurate method. The MoM solution is extended to the calculation of self and mutual impedances of coupled antennas kept at the same height from the earth. The impedance of wire antennas placed above a two-layer model of earth is calculated by incorporating the appropriate Green's function in the MoM analysis. From these calculations, the thickness of the upper region of a two layer earth, above which the properties of lower region show no effect on the antenna characteristics is determined. Thesis Contents In chapter 1, the problem of antenna radiation in the presence of a two or three layered region is introduced. The relevant literature is reviewed and the objective of the thesis is presented. Finally, the organization of the thesis is outlined.

Chapter 2 is devoted to the electromagnetic fields of a vertical electric dipole kept in air above a conducting half-space. The exact electric and magnetic field components are derived directly from Maxwell's equations. Various numerical methods, reported to evaluate the integrals in the field expressions, are reviewed. Two techniques, henceforth called, Method A and Method B to evaluate Sommerfeld-type integrals in this work are described. Method A uses Gaussian integration between the zeros of the integrand and Shanks transform to sum the respective contributions from the integration. Method B is based on QAGI, a powerful routine widely used in Mathematical software libraries (eg: NAG). An asymptotic extraction technique with effectively reduces the range of integration in the direct numerical evaluation of the infinite integrals, is incorporated in the numerical integration. The quasi-static formulas of Bannister and the recently derived analytical formulas of King for the field components in air are presented. They are extended to obtain simple expressions for the fields inside the conducting half-space. The test the accuracy of the approximate formulas in both the regions the values obtained from them are compared with the corresponding ones from the numerical evaluation of the exact integrals. The validity of the analytical formulas is discussed. The properties of the radiated fields of a vertical electric dipole above a conducting half-space are summarized. Chapter 3 is concerned with the analysis of finite vertical wire antenna and its two-element configuration kept above a lossy earth. The chapter starts with the formulation of Pocklington's integral equation for the current on a wire antenna in free-space and its solution using method of moments. Piecewise sinusoids (pws) as bases and pulses as test functions are shown to be a useful combination in the MoM solution. An integral equation for the current on a thin vertical wire above a lossy half-space is derived in a simple manner. The MoM solution with pulse-point matching is seen to be ineffective when the rigorous Sommerfeld formulation is used. On the other hand, pws-pulse solution gives input impedance results which show good rate of convergence. Extensive results of the driving point impedance as a function of antenna height from the air-earth interface and parameter of the earth are obtained and compared with the available data. Coupled vertical antennas placed above a lossy half-space are analyzed by solving the corresponding zero and first phase sequence integral equations. Self and mutual impedances between two vertical antennas above a lossy half-space are computed. The analysis of a vertical wire antenna above a two-layered earth is treated to study the effect of the additional layer on the antenna characteristics. Chapter 4 is devoted to the study of electromagnetic fields of a horizontal electric dipole in the presence of a conducting half-space. First the field expressions in both the regions when the dipole is kept in air are derived. The quasi-static formulas due to Bannister and the recently derived analytic formulas by king for the field components are presented. Their usefulness is tested by comparing the fields obtained from them with the values from the numerical evaluation of the integral expressions. Based on the formulas in Region 2, simple expressions for the fields in the conducting medium are obtained. The radiated field properties are studied. Finally, the electromagnetic field in Region 1, due to a horizontal electric dipole also in Region 1, is investigated with emphasis on the approximate formulas. Chapter 5 treats horizontal thin wire antennas and coupled antennas kept above a lossy earth. An integral equation for the current on a thin horizontal wire antenna above a conducting half-space is derived and its solution with pws-pulse MoM obtained. The input impedance variation with the number of modes used in MoM, height of the antenna from the interface, and parameter of the earth calculated over a wide range of parameters. Coupled horizontal antennas (both broadside and collinearly coupled) kept at the same height from the air-earth interface are analyzed for self and mutual impedances. The effect of dielectric or a conducting layer of certain thickness above the uniform earth, on the impedance properties of horizontal wire antennas is investigated.

For more details click here Back

Title **:** ***Adaptive Evaluation Of Dynamics Of The Vocal Tract Shape And Detection Of GCLS Using MP Filter***

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Abstract

Determination of vocal tract shape from the speech signal has applications in various fields of speech such as diagnosis of dysfunctions of the neuromuscular control system of speech production mechanism biofeed back system for the training program of hearing impaired persons, speech recognition and speech synthesis. The recovery of the vocal tract shape from the speech signal has been attempted for last thirty years. The idea of recovery of the vocal tract shape from the speech signal stems from the following facts. The spectrum of the speech wave is shaped by the frequency selectivity of the vocal tract. These are formant frequencies. The formant frequencies depend on the shape and dimensions of the vocal tract that is each shape is characterized by a set of formant frequencies. Hence the shape of the vocal tract can be recovered from the formant frequency informations. The vocal tract formant frequencies for vowel speech signal can be estimated from the linear prediction (LP) model parameters. This has resulted in the use of LP model for the reconstruction of the vocal tract shape from vowel speech signal. Conventionally vocal tract shape is estimated in the form of a set of concatenated area function from a stationary segment of vowel speech signal using autocorrelation analysis of linear prediction. The area functions are estimated from the reflection coefficient of the acoustic tube model of the vocal tract. The reflection coefficients are evaluated by using the linear prediction acoustic tube (LPAT) model of the vocal tract. The changes in the vocal tract shape are evaluated by analyzing succeeding segments of frames of vowel speech signal. At present major research efforts have been directed towards various applications of the changes in the vocal tract shape during the vowel speech utterances. However, the evaluation of the changes in the vocal tract shape by analyzing the vowel speech signal frame by frame does not represent the true variations of the vocal tract shape. A major purpose of our study is to replace the frame based method by an adaptive filtering method. The thesis documents our investigation of a new approach to evaluate the dynamics of the vocal tract shape. The complete study can be classified as an investigation of three related problems. First we have develop a new approach for the reconstruction of the vocal tract shape from the vowel speech signal using an adaptive filtering techniques. The linear prediction acoustic tube (LPAT) model suggests that the reflection coefficients obtained by the autocorrelation analysis of the linear prediction model for vowel speech signal are equivalent to the reflection coefficients of the acoustic tube model of the vocal tract. The gradient adaptive lattice (GAL) filter for linear prediction has been proved as equivalent to the optimal adaptive autocorrelation method for linear prediction. Hence the reflection coefficients obtained by the GAL filter are equivalent to the reflection coefficients of the acoustic tube model of the vocal tract. We have tested the GAL filter for the reconstruction of

the vocal tract shape from the vowel speech signal. The GAL filter is observed to converge faster for the decreasing values of the forgetting factor, but the steady state misadjustments at the glottal closure instants (GCIs) increase. Hence it is required to avoid the vocal tract parameters at the GCIs. This requires apriori detection of the GCIs. As the second problem we have investigated the problem of detection of the GCIs in vowel speech signals. This problem has been dealt as a general problem of spike detection. We have used a new filtering techniques called midprediction error filter for the detection of the GCIs. As the third problem we have developed a variable forgetting factor gradient adaptive lattice (VFFGAL) filter for the purpose of faster convergence and smaller steady state misadjustments. The limitations of the GAL filter arise because a single value of the forgetting factor results in opposite desirable outcome as far as the speed of convergence and the steady misadjustments are concerned. The VFFGAL filter is developed with the idea of introducing a time varying forgetting factor. In the VFFGAL filter it has been possible to identify two different parameters, the initial forgetting factor value and the step size updation parameter to control separately the speed of convergence and steady state misadjustments. Finally we have compared extensively the changes in the vocal tract shape evaluated by the VFFGAL filter with the earlier frame based method. The evaluation of dynamics by adaptive filtering technique is shown to be more reasonable. In the following we give a chapter wise summary of the thesis. Chapter 1 beings with the significance of the vocal tract in speech science. This is followed by a detail review of the methods of recovery of the vocal tract shape from vowel speech signal. We have discussed the methods of evaluation of changes in the vocal tract shape and their limitations. Thereafter we have presented different adaptive filtering techniques used for the estimation of vocal tract parameters. Then we have presented a review of various methods for the detection of glottal closure instants (GCIs) in vowel speech signal. Finally, we have discussed the variable step adaptive filter. In chapter 2, we have evaluated and discussed the present day methods used for the determination of the changes in the vocal tract shape during vowel speech utterances. The chapter begins with the discussion of the acoustic tube model of the vocal tract followed by a presentation of linear prediction (LP) lattice filter. Then we have presented the relationship between the acoustic tube model of the vocal tract and the LP lattice filter which is known as the linear prediction acoustic tube (LPAT) model. We have discussed the limitations of the LPAT model. Thereafter we have evaluated the limitations in earlier methods of determining the changes in the vocal tract shape. There are two methods of evaluation of the dynamics of the vocal tract shape. The first one is pitch synchronous method. In this method the closed glottis segment in a single pitch period of the vowel speech signal is used for analysis. Using this method, the dynamics of the method are as follows. Though it is possible to find out the exact instant of glottal closure the instant of opening is difficult to ascertain accurately. Hence it is difficult to find out the closed glottis segment of the speech signal. For an average male speaker the closed glottis interval is 2 to 3 msec in duration. Hence for a 10 kHz sampled signal the closed glottis segment contains 20 to 30 samples. Such a small number of samples prove inadequate for a 10th order linear predictor determination. With simulated data we have shown that the area functions recovered by this method do not match the original. In the second method a stationary segment of 20 or 25 msec of speech signal is considered as a single frame. The glottal filter, G(z) and the radiation load, R(z) are filtered out first. Then from the residual speech signal the reflection coefficients are estimated by solving the Yule-Walker equations. Using this method the changes in the vocal tract shape are evaluated frame by frame. The vocal tract shapes thus estimated frame by frame show inconsistent variation. By inconsistent variation we mean that most of the time the variation of the area function magnitudes are considerably large for adjacent frames which may not be possible practically. As there is no direct method of evaluation for the correctness of the variations of the vocal tract shape apart from simultaneous imaging of the vocal tract we have proceeded as follows. The variation of the vocal tract area functions are within 20 Hz. As a result if the dynamics of the vocal tract shape is estimated above a frequency of 40 Hz which is the Nyquist frequency for the area function variations then the dynamics should not show any introduction new frequency components. The results show that higher frequencies are introduced as we go for higher area function sampling frequencies even above the Nyquist frequencies. This proves that the dynamics evaluated by estimating the vocal tract shape frame by frame does not represent the true temporal variation of the vocal tract shape. We obtained the same results with direct implementation of lattice filter such as forward prediction error lattice filter Burg's lattice filter and Itakura lattice filter. In chapter 3, we have investigated the problem of developing an adaptive filtering approach for the reconstruction of the vocal tract shape from vowel speech signal. The chapter begins with the discussion on the proposed adaptive filtering scheme. The performance of the gradient adaptive lattice (GAL) filter is equivalent to that of the adaptive autocorrelation method of linear prediction. The GAL represents an approach that is usually more rapidly converging than LMS (least mean square) but not as computationally intensive as the least square methods. We have tested the GAL filter for the reconstruction of the area functions for vowel speech signal. We have used a preemphasis filter (differntatiator) to remove the effects of the glottal filter, G(z) and the radiation load, R(z). at each sample instant the reflection coefficient of the acoustic tube model of the vocal tract are estimated from the preemphasized speech signal. The vocal tract area functions at each sample instant are found out using the acoustic tube model relations between the area functions the reflection coefficients. To prove the robustness of the proposed adaptive scheme, we have used glottal inverse filtering. We have shown that the glottal volume velocities are reconvered successfully by using the vocal tract transfer functions obtained from the GAL filter. This shows the correctness of the vocal tract transfer function estimated by the GAL filter. The mean square error and the trajectories of the filter parameters are commonly used as a measure of the performance of the adaptive filters. For our purpose we have used a new convergence factor based upon the mean square error of the area functions so that an overall picture of the convergence of the whole vocal tract shape can be obtained. The performance of the GAL filter is observed to depend upon the value of the forgetting factor. With the decreasing values of the forgetting factor the initial convergence becomes faster but the steady state misadjustments at the glottal closure instant (GCIs) increase. In chapter 4, we have investigated the problem of detection of glottal closure instant (GCIs) in vowel speech signal. It is required to avoid the vocal tract parameters at the GCIs as the steady state misadjustments increase at the GCIs. Hence apriori detection of the GCIs is required. The problem of the detection of the GCIs in vowel speech signal has been dealt as a general problem of spike detection. Other examples of spikes are epileptic seizures in electroencephalographic (EEG) signal fine crackles in vesicular sound signal, QRS complex in electrocardiogrphic (ECG) signal. Because of the high frequencies contents the spikes are filtered out using high pass filter. We have sued midprediciton error filter for the detection of GCIs. In midprediction (MIDP), the present sample is defined as the weighted sum of p recent past, p immediate future samples and some weighed value (ao) of the present sample. We have evaluated various properties of the MIDP error filter and compared them to those of the linear prediction (LP) error filter and two-sided prediction (TP) error filter. For vowel speech signals the MIDP error filters is observed to have a high pass magnitude response. Hence spikes at the GCIs appear in the residual error signal. By assuming a lower value of ao, the higher frequency gain of the MIDP error filter can be increased. This helps in reproducing the spikes at the GCIs with higher magnitudes in the residual error signal. As a result it is possible to detect the GCIs with the MIDP error filter for slow glottal closure vowel utterances. For vowels such as /u/, the MIDP error filter is capable of producing prominent spikes at the GCIs which is other wise not possible with conventional linear prediction error filter. The MIDP error filter has zero phase response. As a result the spikes or GCIs in case of the vowel speech signal appear at their original position in the residual error signal. The symmetrical nature of the MIDP help the spikes to appear in the error signal with their base widths. Base width of the spike is an important parameter in characterizing the epileptic spikes in the EEG signal. Hence the MIDP error filter is a better choice compared to the linear prediction error filter for identifying the spikes in an EEG signal. We have compared the performances of MIDP error filter with the LP error filter and the TP error filter with different simulated and natural vowel speech signals. A smaller value of the forgetting factor enables the area function to converge faster, but the misadjustments are large at the glottal closure instants. This is due to the enhanced tracking capability of the GAL filter with a lower value of the forgetting factor. As a results the filter is capable of tracking the nonstationarities present in the signal at the glottal closure instants (GCIs). For the purpose of evaluation of changes in the vocal tract shape, the tracking of nonstationarities appearing in the signal at the GCIs are not important. Because these nonstationarities at the GCIs, appear at a frequency of the 100 to 300 Hz (pitch frequency) where as the frequency of the vocal tract area function variations is within 20 Hz. Hence the filter which is to be used for the evaluation of the changes in the vocal tract shape, is desired to have the property of faster convergence and tracking of long time variation in the signal which corresponds to the variations of the vocal tract shape. The limitations of the GAL filter arises because a single value of the forgetting factor results in opposite desirable outcomes as far as the speed of convergence and the steady state misadjustments are concerned. It would be more appropriate to use a time varying forgetting factor as the speech signal statistics change with time. In chapter 5, we have presented a new filter named as variable forgetting factor gradient adaptive lattice (VFFGAL) filter. The variable forgetting factor gradient adaptive lattice filter (VFFGAL) filter is developed with the idea of introducing a time varying forgetting factor. The idea is to update forgetting factor value by somehow sensing how far away the values of the adaptive filter coefficients are from the optimal value of the filter coefficients. For the purpose we have used the polarity of the error gradient at the successive samples. In the VFFGAL filter it has been possible to identity two different parameters the initial forgetting factor value and the step size updation parameter to control separately the speed of convergence and the steady state misadjustments. For deciding the local convergence we have considered the error energy over a sliding window as the decision-making parameter. Finally we have compared extensively using six different vowel speech signals uttered by thirty different subjects, the changes in the vocal tract shape evaluated by the VFFGAL filter with the earlier frame based method. The evaluation of changes in the vocal tract shape by adaptive filtering techniques is shown to more reasonable compared to the frame based method. With proper choice of filter parameters, the true temporal variations of the vocal tract shape can be evaluated. Faster convergence and less steady misadjustments are obtained using the VFFGAL filter. Finally, we conclude in chapter 6 by summarizing the contributions of the thesis and noting some directions for further investigations in these areas.

For more details click here Back

Title : *A Flow Model Based Control-Theoretic Approach To Dynamic Routing In Some Structrued Communication Networks*
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Abstract

In this thesis the problem of synthesizing optimal dynamic routing strategies for some specific network topologies is investigated. This investigation has been carried out within the framework of dynamic flow models. Traffic routing and related areas of network management are currently undergoing review in many of the existing networks. The concept of dynamic routing which allows for the continuous updation of the routing tables depending upon the instantaneous traffic conditions and the network status, is aimed at profiting from the existence of spare capacities in parts of the network while other parts are over loaded. Dynamic routing strategies are known to offer significant improvements in the efficiency over static routing strategies, provided a proper exploitation of the traffic variations is carried out. The main limitation with dynamic routing has been that it requires a large amount of data of be processed to perform the complex routing calculations. This, however has-been substantially overcome in the recent times, due to the continued improvement in computer technology. Therefore there is a shift in trend towards implementing dynamic routing strategies in both packet-switched and circuit switched networks. State dependent routing schemes for telephone traffic, in which the route for a call is chosen on the basis of the occupancy-state of trunk groups at call arrival have received much attention in the last few year [1]. Along with this shift in trend towards dynamic routing there has also been a growing need to have theoretical frameworks within which the problem of routing more specifically that of dynamic routing can be formulated. While traditionally traffic phenomena in communication networks have been analysed with queneing models [2], these traditional analytical techniques are usually inadequate in dealing with networks with state sensitive dynamics. Queueing analysis is often found to be intractable under non-stationary traffic conditions. In the context of routing these descriptive queueing models tell us something at best about the performance of a network a specified routing strategy. The task of synthesizing the best routing strategy (or a least a good strategy) for a given system is rarely addressed to. Alternative traffic models some of which at least are prescriptive [4] in nature, have been suggested in the context of traffic control, in the recent times. A significant contributions in terms of providing a modeling scheme which allows the synthesis of advanced traffic control rules within the framework of optimal control network by dynamic flow models. The basic model of dynamic flows relates the worth in the amount of packages messages in the systems by means of a deterministic differential equation, to the intensity of newly arriving traffic intensity of traffic discarded by the system and the intensity of successfully delivered traffic. It was assumed by Filipiak in [3] that the intensity of outgoing traffic can be approximated by a (non-linear) function of the system state. The work in this thesis starts with this framework. In a packet switching context it is reasonable to assume that the flow out form any buffer onto its associated

link will increase with increasing buffer occupancy, and will saturate at a value equal to the channel capacity of the link. It is also reasonable to assume that the flow out will be zero when the buffer is empty. Based on these consideration we concentrate in this thesis on a model in which the flow out function increases linearly with the buffer occupancy and saturates at a value equal to the channel capacity of the link. Apart from these practical reasons as against more complex form of functions for the flow out given in [3] we consider that this class of models (in which the flow out increases linearly with buffer occupancy and saturates at the channel capacity of the link) will make an interesting study in itself. The emphasis in out work is therefore on this model. The possibility of treating the network as a dynamical system with the routing variables as control variables, allows us to formulate the problem of optimal dynamic routing as an optimal control problem. This problem is solved by a well known technique of optimal control theory namely pontryagin's maximum principle. The necessary conditions the optimal solutions has to satisfy are specified in terms of a two point boundary value problem in the state and costate variables. It is well known that it is not easy to obtain the solution to these equations. The problem becomes more complicated when the dimension of the network is large since the dimensionality of the system of differential equation in the state and costate variables also increases correspondingly. The emphasis in Filipiak's work [3] is to consider the steady state solution to the costate variables when the duration of network operation tends to infinity. Under this assumption the solution to the costate variables (steady stet) can be obtained by solving a set of algebraic equations. In this thesis we relax the assumption of an infinite time duration operation of the network. Since in practice, networks are subjected to shut downs and need rebooting from time to time we consider that the assumption of a finite time duration of operation corresponds to a practically more realistic situation. An optimal dynamic routing strategy is defined as one which minimizes the total buffer occupancy time. The rationale for choosing this is that a waiting cost is incurred at a rate proportional to the number of customers in the system. Since large networks can be viewed as being composed of simpler network structures the problem of synthesizing optimal (or at least good sub optimal) routing strategies for these large networks can be approached by synthesizing them for the simpler network units. With this perspective we investigate the problem of optimal routing in two simple structures namely a two node unit and a three node unit in this thesis. We also consider some network topologies which are composed of these units. The main results regarding the nature of optimal routing strategy for these network structures are summarized below. Main Results The thesis is organized into six chapter. In chapter 1, we give a control theoretic perspective to routing and state the motivations as well as the contributions of this thesis. In chapter 2 we review the various modeling schemes used in the performance study of networks. We review the dynamic flow model proposed by Filipiak [3] in detail and highlight the reasons for the choice of this modeling scheme in the synthesis of traffic control rules. A brief survey of routing strategies employed both in the context of circuit switched networks and packet switched networks is also presented in this chapter. In chapter 3, we investigate the problem of optimal routing in a two node network in which the faster link has a finite channel capacity. Under the assumption that the initial buffer occupancy on the faster link is below the saturation value (we argue that this assumption though may appear to be overly restrictive, is not quite so inn practice) we drive the set of equations in terms of the link parameters and the input traffic the solution to which specifies the optimal routing strategy. Solutions to these equations requires the knowledge of the load pattern for the entire duration of network operation, and this in turn necessitates on off-line computation. We therefore propose an on-line implementation sub optimal routing strategy for this network. Some numerical examples comparing the performance of the optimal and sub optimal strategies are given. In chapter 4, we investigate the problem of optimal routing in a three-node network. We start with the assumption that all the links of the netwok have infinite channel capacities. The optimal routing strategy in this case is bang-bang and can be completely specified in terms of a single switching instant. The equations that specifies this switching instant is also obtained in terms of the link parameters of the network and the duration (T) of operation of the network. It is shown that the optimal routing strategy has the loop free property. We then relax the assumption of infinite channel capacities and consider the case wherein one of the direct links has finite channel capacity. We show that the loop free property holds good under this situation also. The routing strategy is not bang-bang in nature (we provide example wherein the routing variables taken on-zero non-unity values over intervals). The various modes of network operation are defined and it is shown that out of the 27 possible modes of operation of the network, the optimal routing strategy admits only four of these in a terminal interval. These cases are illustrated with examples. We then impose certain additional assumptions on the link parameters of the network which in practical terms imply that corresponding to a situation in which all the links have infinite channel capacities, the optimal routing strategy is to route the entire traffic arriving at a source node onto the direct link for the entire duration of network operation. Under the assumption that the initial buffer occupancy on the link with finite channel capacity is below the saturation value the set of equations (in terms of the link parameters and the input traffic) required to specifies the optimal routing strategy is derived. Solution to these equations requires the knowledge of the load pattern for the entire duration of network operation and this in turn necessitates an off-line computation of the routing strategy. As in the case of chapter 3, we propose an on-line implement able sub optimal strategy for this network. Some numerical examples comparing the performance of the optimal and sub optimal routing strategies are given. In the final section of chapter 4, we investigate the case wherein the flow out on a link is an exponential function of the (associated) buffer occupancy. The optimal routing strategy in this case is obtained by numerically solving the two point boundary value problem in the state and costate variables. From the numerical investigations carried out for various choice of link parameters, initial buffer occupancies and input traffic, we conjecture that the optimal routing strategy has the loop-free property. It is also observed that there is at most one switching instant and that the network operation always ends with a direct routing of packets at both the source nodes. In chapter 5, we investigate the problem of optimal routing in some specific networks which can topologically be viewed as being composed of the two node network units and the three node network units. We fist consider a network which is a cascade of n two node units considered in chapter 3. For the case wherein m of such network units have their faster link of finite channel capacity (both the link of the remaining (n-m) units are assumed to be of infinite channel capacity) we study the properties of the optimal routing strategy. It is proved that in all the network units which have both the links of infinite channel capacity all of the incoming traffic is routed onto the faster link for the entire duration of network operation. We then consider the specific case where only one unit has a link of finite channel capacity. Under the assumption that the network operation starts with an initial buffer occupancy in the unit which is less than the saturation value (of the buffer associated with the link of finite channel capacity) we drive the set of equations required to specify the optimal routing strategy. Since the solution to this set of equations require the knowledge of the traffic pattern for the entire duration of network operation, and this in turn necessitates an off-line computation, an on-line implement able sub optimal algorithms is suggested along the lines as done for the individual unit in chapter

3. some numerical examples comparing the performance the optimal and sub optimal stratigies are given. We then consider a network topology composed of the three-node structure of chapter 4. In chapter 4, we have specified the optimal routing strategy for the constituent unit in terms of a single switching instant, when all the links of this unit are of infinite channel capacity. We consider a situation in which all the units of this larger topology have links of infinite channel capacity. The globally optimal strategy and under the strategy synthesized form locally optimal strategies for the constituent units are compared for various choice of network parameters in order to get a quantitative idea of the performance difference. Finally we consider a four node hub network in chapter 5, which can be topologically viewed as being compose of the three node units of chapter 4. Under the assumption that all links of this network have infinite channel capacity we investigate the nature of optimal roeuting strategy in this network. It is shown that the optimal routing strategy for this network has certain interesting properties as in the case of the constituent unit. The network operation ends with a direct routing the case of the constituent unit. The network operation ends wit a direct routing of packets at all the three source nodes. The optimal routing strategy for this network also is shown to have the loop free property. The conditions on the link parameters under which the optimal strategy for the network is the same as the synthesized from the locally optimal strategies for the constituent units is obtained. When thesconditions are not satisfied a sub optimal way of traffic routing in this network can be obtained from the locally optimal strategies for the network units. We compare the performance of thesis algorithms with that of the optimal one in order to get a quantitative idea of the difference (in performance). Finally in chapter 6, we summaries the major results in this thesis and indicate some directions for future research

For more details click here Back

Title ***:*** ***Parametric Conditions For Simultaneous Quadratic Stabilization Of Linear Systems***

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Abstract

An important problems in control engineering is to design a single feedback control law which will stabilize finite number of plants. This is known as the simultaneous stabilization (SS) problem. This problem is motivated by several practical considerations, for instance the dynamics of an aircraft varies greatly with the altitude and speed and requires stabilization at all flight conditions. An industrial plant is required to remain stable in different modes due to possible failures of components such as sensors and actuators. Often a linearized model of a non-linear system operating at different operating points has time varying parameters. In such situations, it is important to design a feedback controller which will stabilize the system in spite of such changes. This thesis is aimed at studying the simultaneous quadratic stabilization problem. Consider a finite collection of Linear Time Invariant (LTI) systems denoted by {Ai, Bi}, I=1, 2,…, I and described by the state equations X = Aix + Biu Where x ∈ Rn is the state vector and x ∈ Rm denotes control input. The basic problem considered in this thesis is to find a single state feedback control law u = -Kx such that there is a quadratic Lyapunov function V(x) = x* Px (P being a positive definite matrix) which has negative time derivative along the solution of each of the closed loop systems. This problem is referred to as the simultaneous quadratic (SQ) stabilization problem. More specifically, SQ stabilization problem involves the design of a single control law u = -Kx and a positive definite matrix P such that the derivative of a common quadratic Lyapunov function V(x) = x* Px will be negative definite for the solution of each of the closed loop system. For a general finite set of systems {Ai, Bi}, the problem of showing the existence of K and P is quite difficult and open. In the present thesis we attempt to identify different structure of state space matrices {Ai} and {Bi} for which SQ stabilization problem is solvable. The thesis is outlined as follows. Chapter 1 discusses the importance of simultaneous stabilization problem is control theory with a brief literature review of the methods of controller design for uncertain systems. In chapter 2, definitions and preliminary results for SQ stabilization are introduced. It is then established that if a finite set of systems {Ai, B} with fixed input matrix B, are simultaneously transformable into Controllable Companion (CC) form or more generally into the Hessenberg form (called as Hessenberg family) then they are SQ stabilizable. Algorithms are also proposed to achieve stabilization using a single linear static state feedback controller. It turns out that the family of systems, which are simultaneously transformable into CC form, satisfy the matching condition. However the Hessenberg family is shown to be larger than the family satisfying the matching conditions. At the end of this chapter, conditions for SQ stabilization of a set of systems {Ai, Bi} with uncertain input matrix, are discussed. Chapter 3 reports three new families of systems which are SQ stabilizable. These

families of systems are called (1) partially commutative (2) partially normal and (3) a more general family. Each of these families is defined by imposing certain assumptions on the state space matrices {Ai} and {Bi}. These families of systems are shown to be different from the matched uncertain systems. For each case the existence of a simultaneous quadratic stabilizing controller is ensured. For state feedback controller it is necessary to measure all the state variable for feed back. However in practice, the design of output feedback controller is a realistic approach. Chapter 4 is devoted to the design of static output feedback controller for the simultaneous quadratic stabilization problem. For this purpose we consider a finite collection of minimum phase systems {Ai, Bi, C} described by X = Aix + Biu, y = Cx Where y is the output vector. These systems are in addition assumed to be square systems (equal number of inputs and outputs). Two new classes of systems are then defined which are SQ stabilizable by output feedback controller. An algorithm for computing a single static output feedback stabilizing controller is also presented. Chapter 5 is concerned with the design of a simultaneous stabilizing controller for time delay systems and stabilization of time varying systems. Effectively the results of chapter 2 and 3 are extended for special classes of time delay systems and the existence of a single stabilizing controller is proved. More specifically different class of time delay systems are shown to be SQ stabilizable by a single memoryless state feedback controller. The stabilization of time delay systems is addressed here using a quadratic Lyapunov functional. This chapter concludes with the design of a single stabilizing controller for special classes of time varying systems. Finally in chapter 6 we take two practical examples for which the solution of SQ stabilization problem is applied. These examples are (1) stabilizing controller design for an uncertain aircraft model (2) Tracking controller design for a robot manipulator. In this thesis parametric conditions (structures of systems matrix Ai, and Bi are derived for which SQ stabilization problem is solvable. Chapter 7 summarizes all such parametric conditions and the overall conclusions of the thesis.

For more details click here Back

Title ***:*** ***Connectionist Signal Processing System Characterization Of Representation In Neural Network***

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Abstract

Conventional signal processing based on a parametric representation of signal spaces and as a consequence, operations involving either an estimation of the parameter(s) given sufficiently observations or a manipulation of the parameter(s) to obtain the desired signals needs an exhaustive account of the dependence of the output signal on the input signal in terms of criteria with global scope if not in terms of closed form expressions. Alternative approaches, generally non-linear based on grammatical formulations of signal spaces and operation have also been suggested in the literature to overcome some of the limitations attributed to linearity in conventional signal processing. Several instances of signal processing, generally involving a subjective element in the processor-though not devoid of invariance, eg, recognition of hand written characters, facial recognition, texture identifications etc, exist wherein neither a parameterization nor a grammatical formulation of signal spaces is feasible due to insufficient understanding of the processes underlying the signal generation and/or the volume of data being grossly inadequate to establish input-output relationships. However such situations can be described through finitely many prototype inputs for which the outputs are known either completely or partially. Processors of this kind have been realized through a class of hierarchical non-linear dynamical systems termed artificial neural networks wherein the processor belongs to a parametrically described space and the objective is to estimate (learn) processor parameters(s) given examples of input-output association. Current neural network, research exhibits a plethora of networks, each concentrating on representing specific types of input-output relationships with accompanying procedures to estimate the processor parameters given the examples of association. However the issues of representation have not been adequately investigated with the consequence that no satisfactory criterion exists whereby one can decide on the architecture of the network necessary for a given situation of processors realization. In additions, little attempt is made at providing an axiomatic framework in which neural networks architectures and processors realization can be discussed. Overview of the Thesis I focus my investigations on four key issues related to the representation of signal processor with neural networks, each not unconnected with the others. Representations is interpreted, in this thesis as a decomposition and/or synthesis of the desired function through basis, functions that are not chosen a priori, but are synthesized to suit the requirements of processor realization the requirements are specified through finitely many examples of the desired association. The basis function are however not restricted to be dependent on the family of functions under consideration. The investigation is initiated by considering representation in isolated neurons from the perspective of perseverance of input spaces in input-output associations perseverance of mappings is defined in terms of one-one correspondence, order preservation and preservation of regularity. I establish the existence of

weights, corresponding to isolated neurons, that accommodate a perseverance of the collection of binary vectors in an Euclidean space and of binary vectors. Though this class has uncountably many elements, these elements are organized in finitely many directions the number of directions are related exponentially to the dimensionality of the collection of binary vectors. Perseverance is shown to extend to enlarged discrete spaces derived as certain finite unions of scaled and translated versions of the collection of binary vectors, however without an alteration in the class of perseverance weights. Functions on such discrete spaces under perseverance are equivalent to sequences on the input space and as a consequences linear separability is characterized in terms of the number of sign-transitions in the sequences and learning is shown to be equivalent to an enumeration of weights in the class of perseverance weights and a search for threshold in a linearly ordered spaces. The radix of numbering is shown to have little influence on preservance though the cardinality of the discrete space preserved increases with the radix and the preservance weights tend to bunch around the coordinate axes. Corresponding to every non-null weight vector in an Euclidean space a preservance input space, defined as a discrete space for which this weight would be a preservance weight is identified and the preceding discussion is shown to extend to such an input space, though with appropriate rotations of the relevant coordinate frames. Representation in layered signal processors is the next issue considered in this thesis, however, the investigation is restricted to the specific case of feed-ward ensembles realizing maps on preservance input spaces as linear combinations of neural responses. A single layer processor restricted to have identical weights in all nodes is first considered. The thresholds are in contrast allowed to be distinct. Such a structure is shown to represent all dichotomies on a preservance input space whose preservance weight is used as the common weight. The number of nodes is no more than the number of distinct level-transitions in the sequences along the preservance weight that represent functions over the preservance input space and the number of level-transitions is used to test minimality of an architecture. Learning in this processing structure is shown to involve a process of approximating the collection of inputs described in a training set by a preservance input space a search for a threshold in a linearly ordered space and an analytical solution for the coefficients of linear combination. A similar situation is shown to exist when the weights in the constituent neurons are distinct preservance weights of the same preservance input space. Multi-layered neural processors modeled to realize function as linear combination of neural network responses are shown to be functionally equivalent to single layer neural signal processors, however, fewer nodes are needed to represent a given function when compared with that necessary in the corresponding minimal single layer processor. This is true only when the number of layers is smaller than the number of nodes needed in the minimal single-layer processor. As the preservance input space is discrete an algebraic characterization of function realization with neural networks is considered to establish that linearly separable dichotomies are exactly those partitions on (semi) lattices wherein each member, of the partition, is a semi-lattice. Issues of representation in neural signal processing architectures form the third topic of investigation in this thesis. Typed neural signal processors are defined on continuous spaces, the type number reflecting the degree of layering. Functions realized by neural signal processors of all types are shown to be dense in the space of continuous functions: this is an extension of similar results established, on equivalents of type-1 processors, by Cybenko (1989) Hornik, Stinchcombe & White (1989), to cascades of type-1 processors. Through in study of the functional nature of neural signal processors, four axioms are suggested to describe the current architectural commitments in neural signal processing activity these axioms are sufficiently general to and a unified study of neural signal processarchitectures. 1. Axiom of organization. A neural signal processor is composed of (layers of) three operational stages: measurement, discrimination and aggregation in that order. Preprocessing, if any (preceding, or incorporated in the measurement) is sought to be represented in a neural basis. Measurements are effected on an observation space constructed as the Cartesian product of the input space and a relevant subspace of a union of the space of responses of the distinct layers. 2. Axiom of measurement. A neural signal processor, through the measurement functions in each of the processing (decision making) nodes, induces a foliation of co dimension at least one in the input manifold. This foliation forms the basis of synthesizing (approximating) the desired level curves of the function. 3. Axiom of discrimination. A neural signal processors, through its discriminatory functions renews the foliations induced on the input space by the measurement functions, through a transformation of the stems of the foliations, with at least one of the following properties: (a) Alter the indexing of leaves to retain distinctness in a finite non-zero number of local regions of the input space, (b) Introduce multiple components in the leaves, (c) Associate to at least one component of a leaf of the foliation due to discrimination uncountable many leaves of the foliation due to measurement. Re-foliations provide the basis for establishing equivalences between members (elements) of the input space in ways not possible through the chosen measurement functions. 4. Axiom of Aggregation. A neural signal processor through its aggregation function synthesizes (or approximates) the level regions of processor response through a foliation on the Cartesian product of the stems of foliations on the input space due to discrimination. Concepts in neural signal processors are identified with the level regions of processor response. These axioms, coupled with the earlier stated algebraic characterization of linear separability suggest that the paradigm of neural computing, (specifically the notions of learning and generalization) is not restricted to processors effecting maps between (vector) spaces of numbers. As the notion of a foliation (Lawson, 1974) is one of inducing a partition on a space such that the members of the partition belong to an indexed collection, these axioms allow attention to be directed towards a unified treatment to neural computing especially the analysis (and synthesis) of representation with neural networks. In particular these axioms provide a framework wherein a formulation of problems related to decidability solvability and completeness that dominate the theory of computing these problems lead to queries about the capability of the neural computing paradigm to neural computing-and a relative evaluation of the formalism of Turing machines with the paradigm of Neural Networks can be attempted. A unification. However is not in the scope of this thesis. At an operational level neural signal processors effect (point-wise(nonlinear transformation between integral transforms: this interpretation allows representation in neural networks to be contrasted with other approaches to signal processor realization. The resulting constituents are used to suggest an interpretation to a function representation theorem due to Komogorov (1957a): this interpretation is different from that provided by Hecht-Nielsen (1987a), Kurkova (1992) and Kovacec & Ribeiro (1993). Learning under this interpretation is equivalent to kernel design. The possibility of solution to learning with a priori but partial knowledge of weights a situation relevant in hybrid networks, is indicated by incorporating neural network based function realization for the kernels of the integral transforms. Localization in the representation of neural signal processors is the final issue considered in this thesis. A localization in representation is shown to result from an influence of the kernels of integral transforms as well as from the mechanisms of (point-wise) associatibetween integral transforms. Localization resulting from kernels is shown to restrict the choice of weights in individual neurons to the linear span of window functions 9sequences), however, there is no restriction on the constituent window (sequences). I also establish that the mechanisms of association is restricted to have all derivatives (those that exist) in the linear span of window function effectively suggesting that in the connectionist approach to signal processor realizations signals and their processors are both capable of being described in comparable possibly same, basis space: this feature would be helpful in a formulation of neural network bases systems which decide on the processing characteristics of neural networks. A characterization of localization in terms of wavelet transforms is considered to suggest the operational sense of basis function synthesis in neural network representations. This characterization is different from that provided by Zhang & Benveniste (1992) and Pati & Krish-naprasad (1993). Concepts represented in neural signal processors are shown to reflect evaluation of intra-pattern and inter-pattern feathres the former is influenced by localization due to measurement and aggregation kernels and the latter is a consequence of the mechanism of association between the integral transforms of measurement and aggregation. A also establish that localization in the intra-pattern and inter-pattern predicates restricts concepts represented by every node in a neural signal processor to a localized region with one or more components, in the sheaf of input patterns. I have considered kernels of the reproducing type as a specific example of localization in the integral transforms of measurement and aggregation. These reproducing kernels have been shown to extend the notion of preservance-defined earlier on discrete input spaces to input spaces that are continuous, however with the limitation that not all reproducing kernels are representative of preservance weights, and in these same way not all kernels representing preservance weights exhibit the reproducing property. Based on the discussion of Nashed & Walter (1991) that every reproducing kernel is associated with a sampling theorem, I have established that the nature of representation in neural signal processors is in the sense of approximating concepts that are defined on continuous domains through finite number of (non-uniformly) spaces samples: the finiteness of the number of samples is assured when the concepts are of a localized nature and nonuniformity in sampling is admitted by the Paley-Wiener sampling theorem (op cit). This result implies that conventional neural network-ie, networks of finitely many neurons, each with finitely many inputs-represent concepts in a continuum if the kernels are of the reproducing type. An attempt at representing the (reproducing) kernels of the integral transforms of measurement and aggregation via the paradigm of neural signal processing suggests that in the earlier stated notion of representation, the basis functions synthesized are related to members of a (wavelet) frame. The characteristics of the basic wavelets n the frame are decided by the degree of layering incorporated in the neural networks that synthesize the kernels of the measurement and aggregation integral transforms: larger the number of layers, greater is the degree of localization effected by the basis wavelets. Based on the characterization of representation in neural networks presented in this thesis, I have conjectured that the nature of representation in multi-layered networks is of the following kind: shallow networks are well suited for representing processors that have formal descriptions (ie, a description involving rules of association) whereas deep networks are necessary when the entities operating in a formal system needs to be identified/discovered. In other words, shallow networks are good in symbol processing while deep networks are necessary for symbol synthesis. Present neuro-anatomical evidence does not seem to refute this conjecture in that the cortex and neo-cortex, the seat of (conscious) symbolic activity, is organized to have few layers, each with a wise spread of interconnections. In contrast the midbrain whose functionality is not known in sufficient detail but is believed to be responsible for (sub conscious) associations (part of which is the long term memory trace) are deep networks with localized connections. Organization of the Thesis The findings of my investigation together with a review of signal processing with neural networks is organized as a report consisting of seven chapters. An introduction to the idea of automated information processing stressing on he connectionist approach to signal processing is presented in the first chapter. Some of the historical aspects of the connectionist approach to information processing are also incorporated. This chapter also dwells on the motivations for the present investigation and as a preface, presents an overview of the thesis accompanied by an outline of the thesis organization. A review of signal processing with neural networks is presented in chapter 2. The notion of signals, their processing and associated abstractions followed by prominent models describing the processing in isolated neurons and neuronal ensembles are briefly introduced to provide the relevant background, terminology and notations. An outline of the approaches available in the literature for the realization of signal processors through neural networks is also incorporated. In addition the notions of intelligence and information processing are cursorily reviewed in an appendix to supplement the contents of the first two chapters. The issue of representing signal processors in isolated neurons is taken up in chapter 3. In this chapter, I introduce the notion of preservance and establish the existence of preservance weights. Preservance is initially established on the collection of binary vectors in a Euclidean space of dimensionality n and extended to discrete spaces constructed from the collection of binary vectors through scaling and translation. A characterization of the discrete input space accommodating preservance, the collection of weights that form preservance weights and functions represented on such spaces are incorporated. This chapter ends with a discussion on the extension of preservance to discrete spaces identified with numbering systems of a radix other than binary and a construction of preservance input spacecorresponding to arbitrary, but non-null, weights. Neural signal processor realization in layered ensembles of neurons is focused in chapter 4. The influence of preservance on function realization in single layered neural signal processors is taken up first and this study is utilized in the study of function realization in multi-layered neural signal processors. An identification of preservance input spaces appropriate to the collection of inputs described in a training set and the attendant issues in the representation of input spaces is considered in this chapter. The algebraic characterization of representation in neural signal processors on discrete input spaces forms the final component of this chapter. Characterization of neural signal processing architectures forms the theme of chapter 5. In this chapter, I introduce neural signal processors with types and consider the potential for representation in neural signal processors: the processors are considered operating on continuous input spaces. The functional characteristics of neural signal processors, axioms of neural signal processing and the suggestion for an operational paradigm of neural signal processing are considered in this chapter. A study of representation in neural signal processors in terms of function approximation is the final topic in this chapter. In chapter 6, the issue of localization in the functions represented by neural signal processors on continuous input spaces is investigated. The nature of localization is first studied in the case of isolated neurons and then the study is carried over to feed-forward layered ensembles of neurons. Characterization of localization in terms familiar in the literature of signal processing and implications of localization on the nature of processing are considered in this chapter. The basis functions through which signal processors are realized in neural networks are related to wavelet transforms. In chapter 7, I summarize the findings of my investigation and suggest directions of further study.

For more details click here Back

Title : ***A New Parallel Resonant DC Link Inverter-Fed Three-Phase Induction Motor Drive***

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Abstract

In this thesis a new topology for the parallel resonant dc link is proposed. It offersreduced peak dc link voltage without generating high di/dt. The effect of variousparameters on the link voltage wave form is discussed. The state space analysis is carriedout and the design equations are derived. A current prediction scheme to facilitateinterfacing the resonant link topology to an ac motor is proposed. The operation of thelink with and without clamping circuit is discussed in detail. This study makes it possibleto design drives without clamping. A detailed study of the effect of various parameterssuch as the characteristic impedance, Q factors, input dc voltage and the resonantfrequency on the link losses is presented. The voltage and current stresses on various linkcomponents are also given. The losses are determined and then compared with those ofthe actively clamped resonant link (ACRL). The RVRL topology compared with theACRL topology may reduce the link losses by about 25% of total link losses. The newtopology being truly resonant offers less EMI and requires less number of power devices.It is particularly suitable at high resonant frequencies.A line voltage referenced sigma delta modulation (LVRSDM) scheme is proposed forresonant link inverters. The formulation of the decoding table and the choice of afreewheeling vector are discussed in detail. A comparative study of three differentdiscrete pulse modulation schemes based on the dc link current performance is presented.The proposed scheme is found to be superior than other commonly used discrete pulsemodulation schemes. The study reveals that the LVRSDM scheme reduces losses andprovides improved performance for resonant link type inverters.Lastly, the details regarding the experimental setup are included. The fabrication of theresonant link, the control circuit and the implementation of a current prediction schemeare given. The details regarding the driver circuit and the IGBT inverter are presented.The interfacing of the drive to a personal computer is also given.

For more details click here Back

Title : *Voltage Stability And Contingency Selection Studies In Electric Power Systems*

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Abstract

The present day power utilities are facing the problem of maintaining the required bus voltage due to stressed operation of the network. The power systems which used to be transient (angle) stability limited have now become voltage stability limited. Hence the focus of there search in the power system area is being more concentrated on voltage security and voltage stability studies. Voltage security has been defined[1] as the ability of a system not only to operate stably but also remain stable (as far as maintenance of system voltage is concerned) following any reasonably credible contingency or adverse system change. Voltage stability [1] is the ability of the system to maintain voltage so that when load admittance is increased load power will increase an d so that both power and voltage are controllable. The security assessment involves contingency analysis which can be performed by AC load flow for various outage cases. However, in a practical system number of contingencies are so large that they can no t be analyzed online by the AC load flow methods. Hence the contingencies are first ranked in rough order of their relative severity employing contingency selection algorithms. Then full AC load flow is run for only severe contingency cases. Existing cont ingency selection algorithms employ approximate methods such as local solution method [2], one iteration of AC load flow [3], the linearized load flow methods [4] and distribution factor method [5]. These methods are either quite inaccurate or require larg e computational time. Artificial neural network are being popularly used in in power system and other engineering applications due to their capability to map nonlinear functions accurately and the fast computational speed. Voltage security has been conven tionally assessed with respect to operating limits of bus voltage and or reactive power output of the sources. However, the secure operation of present days system requires ensuring a minimum margin of voltage stability. With the increased use of compensat ing devices such static VAR compensators, which raises the critical voltage to normal operating range, makes the bus voltage itself a poor indicator of voltage stability. For contingency ranking required to assess the relative severity of contingencies, s calar performance indices have been used. The performance indices suffer from two typical problems of misranking and masking effects. One of the attempts to overcome these problems was suggested by Hsu et al. [6] by using Fuzzy Logic. In most of the resea rch works the voltage stability have been considered as static phenomena due to slow variation over a long time until it reaches near to maximum load ability or collapse point. Extensive works on predication of voltage stability exist which can be classifi ed into Direct and indirect methods. In direct method stability margin is defined in terms of additional demand that will take the system to stability boundary. Its used methods such as optimization

method [7], multiple load flow solution[8]etc. indirect methods utilize some indices such as singularity of load flow Jacobian [9], minimum singular value or condition number of load flow Jacobian [10] etc. however, these methods, in general require large computing time and may not be suitable for certain applic ations such as the voltage contingency selection. From literature survey it has been observed that even some simplified power system models exhibit oscillatory type unstable response due to Hopf bifurcation much before the saddle node bifurcation [11] i.e . maximum load ability limit. Dobson et al [12] derived the first order sensitivity factor of system parameters with respect to Hopf bifurcation however, no proper method has been suggested to predict the numerical values of stability margin with respect t o Hopf bifurcation required for the secure operation of power system. Weakness of transmission boundary around the group of buses are the structural weak ness that causes voltage collapse. These group of buses are called as voltage control areas. To deter mine voltage control areas Schlueter et al. [13] developed an algorithm based on load flow jacobian sensitivity. However, with this method, the control areas have to be recomputed with slight change in system loading conditions. One of the recent operating concerns in the power system network is to ensure a minimum margin of voltage stability. This can be achieved by installing compensating devices in the system and taking appropriate control actions. An optimal generation rescheduling scheme with respect t o maximization of minimum singular value has been presented in ref. [14] for enhancing the voltage stability margin. Flexible A.C. transmission system (FACTS) devices are being popularly used in the power systems for improvement of system stability. From t he literature survey it appears that the effect of optimal setting of FACTS devices on voltage stability margin has not yet been explored. Therefore the motivation behind the work presented in this thesis are: (i) To develop a new method for contingency sele ction combining an Artificial Neural Network (ANN) for post outage voltage prediction and a fuzzy logic based ranking method considering both the bus voltage deviations and system voltage stability margin. (ii) To develop a fast method for prediction of nearest saddle node bifurcation point in the system using simulation based Artificial Neural Network [15]. (iii) To present a new method based on optimization technique for determining the distance to closes Hopf bifurcation. (iv) To explore a new method to determine voltag e control area which remain valid for wide range of operating conditions. (v) To study the effectiveness of optimal adjustment of generators and FACTS devices' output in voltage stability enhancement. A brief description of the work reported in the thesis is given below: Chapter 1 introduces the voltage security and stability problems presents a brief survey of research work carried out in the areas of voltage contingency selection and the voltage stability studies. In Chapter 2, a fast method based on Artific ial Neural Network (ANN) has been developed for post outage bus voltage prediction. Contingency ranking has been performed considering both voltage stability margin as well as bus voltage deviation and employing a Fuzzy Logic based method. Chapter 3 presen ts the development of a new method employing an analog simulation based neural network to determine the nearest saddle node bifurcation point in loading parameter space. Chapter 4 reports the development of a method for predication of minimum distance to hopf bifurcation in terms of system loading parameters. The estimation of closet Hopf bifurcation has been formulated as an optimization problem. Chapter 5 describes a new method based on entropy concept to determine voltage control area and explores its va lidity for wide operating range. In chapter 6 a simple approach for sitting the FACTS devices has been developed based on identification of weakest bus and the knowledge of voltage control areas. An optimal power flow formulation has been suggested to dete rmine the optimal adjustment of output of generators and FACTS devices. Their impact on enhancement of the voltage stability margin has been studied. Chapter 7 concludes the main findings and significant contributions of the thesis and provides a few sugg estions for further research work in this area.

For more details click here Back

Title : ***Artificial Neural Network Based Power System State Estimation***
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Abstract

State estimate in electric power systems [1] is a function used to determiner the systemoperating state (complex bus voltages) by filtering the noise, which is usually in theeliminated measurements acquired through the data acquisition system. The output ofthe state estimator is used as in feed to various advanced functions of the EnergyManagement Systems (EMS). Depending on the variant or environment nature of measurement s and model of thepower system state being utilized the state estimation can be broadly classified into threecategories.(i) Static State Estimation(ii) Tracking State Estimation(iii) Dynamic State Estimation The static stat estimation is defined as the data processing algorithm for convertingredundant meter readings and other available information into an estimate of the statevector while measured data are taken to be time invariant and static model of thepower system is considered. The tracking state estimation algorithms are based on a simple extension of the staticstate estimation techniques. The utilizes the recent available value of the system state toupdate their eststimated values no iteratively during the subsequent sampling periodThis class of estimators has arisen from the natural need of making static estimators asefficient as possible in regard to the computational speed making them more suitablefor real-time implementation. The dynamic static estimator utilizes in addition to the present states the previousestimates of the state. The capability of forecasting the state vector one step has is animportant advantage of the dynamic estimators. State prediction gives a longer decisiontime to the system operator because economic dispatching security assessment andother functions can be performed in advance. In dynamic state estimation dynamicmodel for the time behaviour of system state is utilized whereas tracking and static stateestimators do not require any dynamic model of the system states. N the present thesisonly static dynamic state estimators have been studied. Several static state estimation algorithms [2] have been reported in the literatureamong them Weight least squared (WLS) algorithm developed by Schweppes et al[1] is quite popular. Some of the several of the WLS methods such as fast-decoupledstate estimator algorithm [3] are being presently used by many utilities. For dynamic state estimation Kalman filter based algorithms have been utilized. Thedynamic state estimators based on extended Kalman filter (EKF) technique [6] havebeen popularly used .The dynamic state estimation is solved in two steps viz. stateforecasting step and state filtering step. Some of the modules required to be performed along with the main state

estimationinclude the operability analysis network topology and bad data processing. Bad data processing detects the presence of gross error (bed data) in the measurementset and identifies the bad data which are eliminated from the set of measurement to beutilized for state estimation. For bad data detection weighted sum of squared residual(WLS formulation) and for their identification the normalized residue algorithm basedon some hypothesis testing such as chi square test have been used .In general the beddata processing involves interactive schemes which are time extensive in nature as itrequires a state estimation run in each iteration .In order to reduce computational timedecoupled detection and identification schemes [4] have been suggested. Topology of the power system network can be generated directly using the breakersstatus. However presence of noise in the status measurements arising out of missing orcorrupted data poses difficulty in the direct determination of the network topology.Several models of topology processors have suggested on similar lines as the bad dataprocessing .The algorithmic models apart from being iterative in nature are unable todetermine the network topology in those case where some of the models or the systembecomes unobservable [7]. Before performing the state estimation it is necessary to check whether the system isobservable or not. Operability analysis is carried out to determine whether the set ofmeasurements are enough in number and geographically located in the system to makeit observable .Two types of algorithms are used for the observability analysis viz. Thenumerical based and topology based methods [8]. Some of the problems faced by thesemethods include the ill-conditioning problem due to floating point calculating in thenumerical operability methods and misclassification of the observable andunobservable case in the topology observability methods. Artificial neural networks (ANN) have been recently applied to various powersystem problems [9] because of the variety of advantages (such as high computationalrates robustness fault tolerance etc.,) they offer over the conventional computingsystems. Only a small effort has been made in applying the ANN models to stateestimation . Nakagawa et al [10] used Hopfield network for static state estimation using secondorder load flow model and tested on a simple six bus system. They found that theHopfield based state estimator model was not superior to the computational point ofview over the conventional methods .For network topology and bad data processingAlves da Silva et al [11] used pattern analysis approach based on multiplayerperception model using Optimal estimate training (OET) whereas in [12] modifiedoptimal estimate tranning (OET2) was utilized. One of the major challenges to the researchers in the real-time implementation ofthe state estimators is to bring down its computational time within the scan cycle of thedata acquisition system .The periodicity of the exerting conventional algorithmic stateestimators is of the order of a few minutes whereas data acquisition cycle is of the orderof few (1-10) seconds. The Artificial neural network offers great promise to solve those problems, whichare time extensive in nature and have no clear-cut mathematical model. The literaturesurvey reveals that the artificial neural network application to state estimation has notbeen fully explored. Only few ANN models have been tried out for static estimationbad data and topology processing. Therefore the motivation behind the work reported in this thesis has been to explorethe feasibility of applying ANN models to different module of the state estimation suchas for (I) Static state estimation, which can be run at a periodicity matching with the scancycle of the data acquisition system;(ii) The bad data processing, which can be carried out without running the stateestimator thus offering substation computational time advantage;(iii) The dynamic state estimatation (to both of its state forecasting and stateFiltering steps) in order to overcome the problems related to modeling of system statedynamics and to meet the real-time requirements;(iv) The network topology determination to avoid iterative process required by thecomputation intensive algorithmic methods; and(v) The observability analysis. A brief description of the work reported in the thesis is given below.Chapter 1 introduces the problem of state estimation presents a brief survey of literaturein this area and sets motivation behind the present research work. In Chapter 2 basic concepts of Artificial neural networks a brief survey of ANNapplications to some of the power system problem problems have been presented. In Chapter 3 observability analysis using two Artificial neural network and a briefsurvey of ANN applications to some of the power system problems have been presented.In Chapter 3 observability analysis using two Artificial neural network model viz.Multilayer perception based on Back propagation algorithm (BPA) and Counterpropagation network (CPN) have been tried out. These ANN models for power systemobservability have been trained and tested for both the observable and unobservable case.The results have been compared with those obtained from an integer algorithm [5]. In Chapter 4 Artificial neural network based topology procedure has been explored. Fordetermination of the power system network topology three ANN models based onMultilayer perception using Back propagation algorithm Counter propagation networkand Functional link network (FLN) have been utilized and tested for both noisy as wellas noise free measurement sets .The effectiveness of the ANN based network topologyprocessor has also been tested in those case where system becomes unobservable andalgorithmic methods face difficulty in determination of the topology. Chapter 5 presentthe development of the static stets estimator based on four different Artificial neuralnetwork models viz. Multiplayer perception based on Back propagation algorithmCounter propagation network Functional like network and Hopfield network. The abilityof the proposed ANN based static estimators to process the non-Gaussian noise has alsobeen explored .The results of ANN based static estimators to process the non-Gaussiannoise has also been explored .The results of ANN based state estimator models have been(FDSE ([3]. Chapter 6-report s the development of artificial neural network based model for the baddata detection and identification .A Counter propagation network has been used for boththe detection and identification of the bad data. Further the capability of the Functionallink network based static state estimator to inherently filter out bad data has also beenestablished .The results have been compared with those obtained from a conventionalfast decoupled state estimator and decoupled bad data processing algorithm [4]. Chapter 7 present development of dynamic state estimation using artificial neuralnetwork models. The dynamic state estimation problem has been attempted in two steps.The state filtering step and state forecasting step. For the filtering step. Functional linknetwork model and for the state forecasting step a Time-delay neural network (TDNN)model have used. The results of the ANN based dynamic state estimator has beencompared with the results of an extended Kalman filter algorithm. The studied in the above chapter have been conducted on three sample systems. Finally Chapter 8 concluded the significant contributions of the thesis and identifies thefuture scope of work in this area.

For more details click here Back

Title : *Multiple States Of Silicon Related DX Centers In ALxGA1-XAS*
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Abstract

Donor impurities in III -Semiconductors and their alloy give rise to deep level defects known as DX centers. They will rise from isolated donor impurities a pressure of current debate in literature brings forth several unresolved issues fundamental to the understanding of DX phenomena. There is a lack of consensus not only on microscopic structure of this deep donor but also its charge state and the nature and perspective or the role of multiplicity to understand ing DX related phenomena. This thesis focuses on the study of multiple related DX state in AlGAS using capacitance stats have been studied using the multiple DX related state has been studied using the isothermal time analyzed transient spectroscopy (TATS) in high composition alloys for distinct emitting state have been obtained using higher orders TATS in the some silicon doped AlGaAS sample for the first time the local model of multiple DX states which observed experimental test by monitoring occupancy of the four DX states. A novel method has been suggested to test the found that the multiplicity in captures barrier energies. The methodology provided into work and can be applied to the study of energy clam to met stable defects in semi conductions.

For more details click here Back

Title : *Voltage Security And Loss Minimization Studies In Electric Power Systems*
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Abstract

Power system security has been defined as the ability of the system to meet its load without unduly stressing its apparatus or allowing network variables to stray from prescribed range. Voltage security has recently gained importance due to stressed operat ion of power systems. To assess voltage security of a system operating limits of bus voltage magnitudes and reactive power output of sources are checked for various contingency cases. Two major functions of power system security are security assessment and security control. The security assessment involves contingency analysis, which can be performed by AC load flow for various outage cases. However, the number of contingencies in a practical system is so large that they cannot be analyzed on line by AC loa d flow methods. In order to reduce the computational time, the contingencies are first ranked in rough order of their relative severity employing contingency selection algorithms. Then full AC load flow is run for only severe contingency cases. Contingency selection algorithm requires some fast methods, which may be approximate, to assess the system states following each outage. Most of the works on voltage contingency selection have attempted used of approximate methods such as local solution method, one iteration of AC load flow, the linearlized load flow models [2,9] and the distribution factors [8]. Existing models of these methods, in general, provide results with large errors and many of them are based on unrealistic assumptions. The distribution fact or method suggested in reference [8] is based on decoupling assumption, which may not be applicable in heavily stressed condition of power systems [12]. In order to assess the relative severity of the contingencies scalar performance indices [1] have been used. Two typical problems with the performance indices are the masking and misranking effects. To avoid the misranking the proper selection of weights for voltage and reactive power performance indices can be utilized. One attempt to use optimal weights in the performance indices is by Hal pin et al. [4] who have utilized to probability based model. After performing the security assessment, if the system is found to be in insecure state, security controls are exercised in order to bring the system into s ecure state. It can be achieved by corrective rescheduling or by running security constrained optimal power flow, provided the available controls in the system are capable of overcoming the system insecurity. Most of the work on optimal power flow uses the minimization of fuel cost of the fossil units and or system transmission loss as objective [7]. However, cost characteristics of generating units may not be available with certain utilities. Hence, optimal power flow can not be formulated on cost criteria .

Such utilities also need some guide lines to optimally adjust the real and reactive power outputs of sources. One possible criteria to allocate the optimal settings of both the real and reactive power outputs of sources, can be the minimization of the sy stem transmission loss. This may also help in enhancing the load delivery capacity of those utilities which are facing shortage of power. For solving optimal power flow, the conventional methods include the classical coordination method, linear programmi ng quadratic and Newton's method etc. however, these methods in general require close initial guess of variables and suffer from the convergence difficulties and local optima. Genetic algorithm (GA) [6] is becoming popular due to its robust ness in finding near optimal solution. GA works on coding of parameters instead of their actual values and used multiple path search along with probabilistic transition rule in parameter space. It has been applied to some of the power system problems and is yet to be tri ed for the solution of complete optimal power flow problem. Conventional security constrained optimal power flow considers the contingency constraints to the optimal power flow problems which leads to the implementation of preventive control action. Thoug h, this formulation may ensure a higher level of system security, but reduces the economic benefit of the optimal operation. In order to maximize the economic benefit of optimal power flow, system operation in correctively secure state has been suggested [5]. Base case optimal power flow results are not modified for the contingency cases. The corrective actions for each of the contingencies are planned in advance with the help of optimal power flows. However, sometimes, it may not be feasible to bring the system to normal state with available controls, specially following the severe contingencies. This leads to infeasibility of the optimal power flows. Very few works have been reported to handle infeasibility of optimal power flow or divergent power flow ca ses [3, 10, 11]. The corrective action planning to solve infeasibility of optimal power flow has been attempted in reference [3] by adding fictitious reactive power sources at problematic buses. Therefore, the motivations behind the work reported in this thesis are: (i) to develop more accurate models of linear zed load flow suitable for predicting the bus voltages which can be utilized for voltage security assessment. (ii) To explore new set of distribution factors to directly compute post outage voltages and rea ctive power output of the sources following the outage of a transmission branch or a generator. (iii) To explore new higher order voltage and reactive power performance indices in order to eliminate the masking effects and a method for selection of optimal weigh ts of the performance indices to elimates the misranking problem. (iv) To develop a new optimal power flow model, considering system transmission loss minimization as an objective, to determine the optimal settings of the real and reactive power outputs of the sources and apply the genetic algorithm for its solution. (v) To suggest a method based on eigenvalue analysis for planning corrective controls in order to enhance the system voltage / reactive power security and to eliminate the infeasibility of optimal powe r flow problem. A brief description of the work reported in the thesis is given below: Chapter 1 introduces the power system security and optimal power flow problems, present a brief state of art survey on the subject and sets the motivation behind the research work carried out in the thesis. In Chapter 2, six different models of new linearlized load flows for voltage security analysis have been developed based on the principle of linear zing non linear power flow equations (in both polar and rectangula r coordinates) around complete operating range using the least square error and integral square error minimization principles. Chapter 3 presents the developments of new set of distribution factors which have been computed using the sensitivity property o f Newton Raphson load flow Jacobian at a base operating point. These have been used to calculate the post outage voltages and reactive power output of the sources following the outages of both transmission branches and generators. Chapter 4 investigates f ew higher order voltage and reactive power performance indices to remove masking effect. In order to avoid the misranking effect, a new procedure to compute the optimal weights has been suggested. Chapter 5 formulates the optimal flow problem having objec tives of system transmission loss minimization to obtain optimal scheduling of both real and reactive power output of sources. The genetic algorithm has been used to solve the loss minimization problem and results have been compared with Flechter's quadrat ic programming technique. Chapter 6 presents the development of a method for corrective action planning which determines the controls to enhance the voltage / reactive power security. The left eigenvector corresponding to the minimum eigenvalue of reduced power flow Jacobian has been utilized to compute the corrective controls. Chapter 7 highlights the main findings of the thesis and identifies the scope for future research in this area.

For more details click here Back

Title **:** ***Parametric Modelling Of Non-Stationary Signals By Linear Time-Dependent Processes***

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Abstract

In this thesis, various aspects of modeling of non - stationary signals are considered. A variety of non - parametric and parametric methods has been introduced in literature to analyse non - stationary signals. The present thesis deals with relating various mod els to a unified framework and addressing key issues of non - stationary modeling. This, in turn, consolidates the existing techniques and finds new avenues in parametric modeling of non - stationary signals. The non - stationary signals that we come across in n ature vary over time in the following aspects , viz , in frequency content or in energy content or in both. For estimation of evolutionary spectrum by parametric rational transfer function model, so far, attention was focused only on the time variation of AR /ARMA coefficients. The time variation of gain (i.e. the input noise energy) was ignored in this regard. In the present work, it is illustrated that both of those features are important in their own ways. The time variation of AR co - efficients indicates th e change in spectral shape with time, and the time varying gain stores the information about signal intensity. Moreover, the time variation of tone feature can not compensate for the time variation of other feature in the estimated evolutionary spectrum. W hile modeling non - stationary speech signals by time - dependent AR model, the time variation of gain becomes prominent. This feature is really a characteristic of the production mechanism of speech. We present here a non - iterative technique to estimate the g ain parameter. Our study reveals that the AR model with time - varying gain is a suitable model for speech phoneme. Various types of parametric models for non - stationary signals are in use at present. These are the damped sinusoid or complex exponential mod el, amplitude modulation (AM) frequency modulation (FM) and amplitude /frequency modulation (AFM) models, damped sinusoids are used for modeling transient signals like nuclear magnetic resonance (NMR) and electromagnetic pulse (EMP) responses. It is known that the speech signal can be generated by the AFM model. In radar and sonar applications, the FM and AFM signals are encounted. The main difficulty that we face in employing the above mentioned modulated processes (AM, FM and AFM) to model non - stationary signals is that it leads to a set of non - linear equations, which are to be solved. Our study reveals that all the above mentioned models are only the special cases of the time - dependent ARMA (TARMA) model. Remember here how an indamped sinusoidal signal ca n be modeled by a constant coefficient AR process and the signal parameters can be estimated through and estimation of the process parameters. It is shown that in the general framework these models keep their own distinctive features, and they can be class ified according to the time dependence of position/locus of the pole of the TARMA model. Since various models can be related to the general TARMA model, the equivalence provides a unified approach for synthesis and analysis of the signal. By using a set of basis functions to represent the time variations the time variation of the parameters of the TARMA models, one may transform the convoluted non - linear

estimation problems to the simple linear ones. This new approach of parametric estimation increase the a ccuracy of estimation of the base band signal, compared to what can be achieved by the existing non - parametric methods. The introduction of time - dependent damping factor adds a new dimension in modeling transient data in terms of accuracy/flexibility of th e models, as shown in the thesis, The ECG is an example of a non - stationary signals with its characteristic pseudo -periodicity. The amplitude modulated sinusoidal model is a natural choice for such signals. It is found that when the modulating signal is e xponential function of time, it can represent the burst type QRS complex very well. This model not only can synthesise the ECG signal as a whole but also its constituent waves. As the consitituent waves of ECG give indications of various activities of the heart, this model may be useful in diagnosis. The thesis is written to highlight the need of modeling any natural signal according to its own characteristics. The study shows effectiveness and area of application for each modeling technique, and how estim ation of parameters for various models can be carried out under a unified framework. Finally, the thesis directs attention to identify the topics for future research.

For more details click here Back

Title	***:***	***Investigation Of Insluating Properties Of Vacuum Under High Voltage***
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Abstract

Vacuum has very old history of being used as electrical insulation. It was used in X - ray tubes, electronic values, high frequency capacitors and interrupters etc. it has good are quenching property also. For this reason it is being now used widely in power circuit breakers. Other proposed application are in vacuum insulated fuses, lightning arresters and cryocables. The most recent application of vacuum is in space based high voltage apparatus, such as satellites and space stations. The advantages of use of vacuum as electrical insulation are that it reduces the weight and volume of the system considerably and offers no manufacturing and aging problems. It is also safe and self healing. While going through the international literature, it was found that mos t of the experiments performed for the investigation of insulating properties of vacuum under high voltage have been on highly clean conditions of vacuum where sophisticated methods of cleaning and electrode preparation were adopted. This may have been pos sible for a few number of electrodes to be tested under ideal conditions in the laboratory. However, it is not practically feasible in case of large manufacturing process. It is also questionable if it is maintained over the life time of an apparatus of 40 - 50 years. Different electrode conditioning and baking methods are reported which have given rise to very high breakdown strength temporarily. Most of the results are reported for a particular pressure by individual authors. Moreover, different shapes and sizes of electrodes have been used with different types of applied voltages in different publications. Hence, it was not feasible to make a comparative statement of these results. It was also found that different terminologies were adopted to define an ele ctric field configuration. The concept of weakly no uniform field was not mentioned in any publication. Keeping all these in mind. It was proposed to investigate properties of vacuum under normal conditions in practice adopting simple methods of cleaning a nd without electrode conditioning and baking. Experiments were carried out at a wide range of pressure between 0.34 and 10 - 6 Torr. Same electrode pairs investigated for both ac and impulse voltages. Electrode shape and size were chosen to remain within wea kly no uniform field configuration still having wide range of degree of uniformity, n. the objectives of the work were (a) to investigate the insulating properties of vacuum under practical conditions (b) to estimate the degree of uniformity by Change Simu lation Method and (c) to find the dependency of the breakdown strength with ac power frequency and lightning impulse voltages and the degree of uniformity. A brief description of the work follows. In Chapter 1 a comprehensive literature survey is reported and some of the experimental work. Performed earlier by other researchers, under different conditions of vacuum are described. Different factors influencing the breakdown strength are

discussed. After a criteria examination of earlier research work, the sc ope of the present work is described in this chapter. In Chapter 2, the suitability of the Charge Simulation Method (CSM) for the estimation of electric fields is described. This method has been applied to investigate the electric field and the degree of uniformity of twelve different electrode pair configurations under two different conditions of voltage applications, (a) symmetrically applied voltage of ± U/2 and (b) asymmetrically applied voltage of U and O. only point charges were adopted to simulate th e equipotential electrode surfaces although different types of charges and their combinations such as line charge, ring charge etc. have been used by other workers. The results are in good agreement with others. Numerical value of degree of uniformity.n ha s been estimated for different electrode configurations having a unit gap distance. The results obtained enable to estimate the effects, such as that of grounding of one of the electrodes on the field, by calculating the value of n . the method applied in t his work for electrodes having rotational symmetry can be conveniently extended to any electrode used in practice. It may be useful in designing GIS and space based systems more confidently. The development of the experimental setup is described in Chapte r 3. a vacuum chamber (test vessel) was needed to be fabricated as the commercially available chambers, made of stainless steel, were not compatible to the specific needs for this work. The detailed of this vacuum chamber are furnished. A mechanism was fa bricated which helped to set a desired gap distance between the electrodes without disturbing the vacuum conditions. A commercially available vacuum pumping system was procured for the work which was not suitable for high voltage applications. several modi fications had to be made to this pumping system which include the design and fabrication of a heater and a solenoid operated value which is used to minimize the backcrossing of oil. This way a know how for making conventional vacuum pumping system compati ble for high voltage application was developed. Fabrication and cleaning of the electrodes with the help of commercially available methods are also described in this chapter. The experimental results under alternating power frequency voltage (ac) and light ning impulse voltage (ii) are discussed Chapter 4 and 5 respectively. Photographs showing stages of glow phenomena observed at low vacuum levels are given. Variation of breakdown voltage of vacuum has been investigated with changing gap distance and pressu re. Characteristic curves have been measured revealed the typical behaviour of vacuum as a dielectric medium. It is interesting to observed from the measured results how the breakdown strength of vacuum first decrease from a high value at slightly less th an atmospheric pressure to very low values at low vacuums (ambient plasma state) and then rises again to very high values at high vacuums(10 - 4 to 10 - 6 Torr). The results obtained under high vacuum have been analyzed for their dependency with the degree of uniformity, n of the field. In chapter 6, the results obtained from the present investigations have been compared with those obtained by other workers. It can be seen that the breakdown strength of practical vacuum are slightly lower than that obtained un der ideal conditions by others. The experimental setup developed in this work could be used for further investigations under switching impulse voltages and with solid insulators in vacuum. Suggestions and the scope for a continued research has been discuss ed at the end of this chapter.

For more details click here Back

Title : ***Investigation Into Static And Dynamic Aspects Of Voltage Stability***

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Abstract

Synopsis Voltage stability has been defined [1] as the ability of the system to maintain voltage so that when load admittance is increased, load power also increases and both power and voltage re main controllable Voltage collapse is the process by which voltage instability leads to very low voltage profile m a significant part of the system resulting into partial or total blackout. oltage stability studies have been carried out for variation (increase) m system loading or for contingency conditions Most of the works have considered voltage stability as static phenomena due to slow variation of voltage over a long time until it reaches near to maximum loadability or collapse point Load flow equations have been used as model and to explain the phenomena, singularity of Newton-Raphson load flow Jacobian has been observed [2-4]. Minimum singular value and condition number of the load flow Jacobian [5-7] have been popularly used as indicators to predict static voltage stability in A C systems Several other techniques such as optimisation method [8], multiple load flow solutions [9] etc have been used to predict extreme loading conditions and statu voltage stability margin l From the literature survey, it appears that most of the works [10,11] on voltage stability analysis of AC-DC systems have represented AC systems by their Thevenm equivalent The concept of minimum singular value and condition number has not been applied to combined AC-DC systems considering the detailed representation of AC systems One of the recent concerns of maximising voltage stability margin has led the researchers to find out new objectives for rescheduling real and reactive output of the sources An optimal generation scheduling scheme with respect to maximisation of minimum singular value of the Jacobian has been presented in ref [12] Traditionally, optimal power flow (OPF) is run for generation rescheduling in view of minimisation of the total fuel cost of generation or minimi sation of the system transmission loss From the literature survey it appears that the effect of ' the various optimal power flow schemes on the voltage stability margin has not been explored One important outcome of the studies reported m the literature is the general consensus that bifurcations in underlying mathematical models of a power system are closely linked with various types of instabilities This is especially true for voltage collapse precipitated by the slow variation m a system parameter such as load etc A bifurcation is a qualitative change in the phase portrait of a dynamical system that occurs as a system bifurcation parameter is varied Bifurcations and chaos have been demonstrated by several researchers [13] m simple

models of power system Hopf bifurcation phenomena has been studied m ref [14-16]. A fourth order generator model along with excitation system have been considered in ref [15] and it has been shown that a complex pair of eigenvalue associated with excitation system results m Hopf bifurcation Most of the works [17,18] showing chaotic invariant set have considered only the load dynamics along with swing equations and have observed only the period doubling route to chaos The literature survey reveals that the detailed model of induction motor with speed depen dent generalised loading and speed governing control loop has not been considered Also the dynamics of various other dynamic elements such as OLTC and SVS have not been investigated The bifurcation and chaos are unwanted phenomena in the power system Hopf bifurcation reduces the stability margin, especially the subcntical Hopf bifurcation in which the operating point is stable but its region of attraction is reduced by the surrounding unstable periodic orbit A power system though being dynamically stable can exhibit a bounded behaviour when the stable operating print is perturbed to attracting region of chaos The chaotic systems have continuous broad band spectrum and can introduce harmonics especially low frequency harmonics in the system, even in the absence of any harmonic source u One of the simplest ways to avoid bifurcation and chaos is to move the system out of the chaotic region by restricting the parameter to enter this region However, it reduces the system loadability Recently new devices for the control of voltage and power have been developed [19] The prominent amongst these components are static var compensators (SVC), static phase angle regulator (PAR) and controllable services capacitors (CSC) The effectiveness of these devices for controlling bifurcation and chaos has not been explored Therefore, the motivation behind the work reported m this thesis are (1) To extend the concept of minimum singular value and condition number of load flow Ja-cobian to combined AC-DC systems considering the detailed representation of AC system and also to AC systems having voltage dependent loads (11) To explore the effect of different generation rescheduling schemes on the static voltage stability margin considering voltage dependent capabilities of synchronous machine (in) To observe various types of local bifurcations such as static and Hopf bifurcations and their effect on stability considering the dynamic models of generator and loads, to determine critical loading and control parameters' values and to ascertain the predominant states causing Hopf bifurcation using participation analysis (IV) To observe various types of global bifurcations and chaos considering the OLTC, SVS, load and generator dynamics and possibility of voltage collapse due to these phenomena (v) To study the effectiveness of FACTS devices in eliminating dynamic bifurcation and chaos A brief description of the work reported in the thesis is given below Chapter 1 introduces various aspects of voltage stability and presents the state-of-art survey on the subject In Chapter 2 the minimum singular value and condition number of load flow Jacobian has been used to predict static voltage stability of integrated AC systems considering voltage de pendent characteristics of loads and extended, for the first time, to combined AC-DC systems considering detailed representation of AC systems. The effect of control mode switching m the DC link and bus type switching in handling the reactive power limits of generators on the static voltage stability has also been demonstrated In Chapter 3 four different schemes of generation rescheduling such as minimisation of total cost of generation, system transmission loss, maximisation of minimum singular value and a in new scheme have been used to maximise the static voltage stability margin Voltage dependent capabilities of synchronous machines have been considered m the model The new proposed scheme of generation rescheduling has consistently resulted into maximum voltage stability margin for various loading conditions Chapter 4 investigates the local bifurcations namely static and Hopf bifurcation in various sample power systems Effect of static bifurcation on power system stability has been demon strated by studying the associated unstable manifold of type-1 unstable equilibrium points on the stability boundary Center manifold theory has been used for studying the stability of Hopf bifurcation and determining critical values of parameters The effect of local bifurcations on the stability m parameter space has also been studied Chapter 5 presents the sample system study results on observance of various types of global bifurcations such as period doubling bifurcation, bifurcation of periodic orbit to torus, chaos via period doubling bifurcations &; via torus breakdown and boundary crisis Lyapunov exponents and dimension of the attractor have been determined Broad band frequency spectrum has been observed under chaotic oscillations Occurrence of voltage collapse due to subcntical Hopf bifurcation and boundary crisis has been demonstrated Chapter 6 demonstrates the application of FACTS devices such as PAR, SVC and CSC for control of bifurcation and chaos A first order delay model of these devices have been considered It has been found that the dynamic bifurcations and chaos can be eliminated with proper selection of control signals and the controller gam parameters Chapter 7 concludes the main findings m this thesis and gives suggestions for further work in this area References [1] IEEE system dynamic performance subcommittee, Voltage Stability of Power Systems Concepts, Analytical Tools and Industry Experience, IEEE document 90TH0358-2-PWR-1990 [2] P W Sauer and M A Pai, Power System Steady-State Stability and the Load Flow Jaco-bian, IEEE Trans on Power Systems, Vol 5, No 4, Nov 1990, pp 1374-1383 [3] V A. 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For more details click here Back

Title	***:***	***Nonlinwear Dynamical AnalysiS And Predictive Coding Of Speech***
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Abstract

Special signal coding has been an active field research for over a couple of decades and continues to be so in spit of the increasing proliferation of optical transmission media of relatively unlimited bandwidth. This is because of the continued and in fac t increasing use of band limited media such as satellite lines and radio channels and bit limited storage media such as CD - ROMs. Also the applications of space decoding have become numerous in recent times .a major effort has been given in the last ten y ears to the development of a class of analysis – by - synthesis coding schemes for low and medium bit ate speech coding. Most medium and low bit rate speech coders are based on the speech production model of a time verifying linear filter excited by a so urce. Such coders are usually designed to estimate the linear filter coefficients and the excitation sequence in a frame - by – frame manner such that the output of the filter approximates the speech signal in some sense. A large portion of the recent researc h effort has been directed to the design of appropriate excitation functions rather than to investigate alternatives to the linear filter from. However the above paradigm for speech coding may now be approaching a stage of saturation as far as improvem ents in terns of performance parameters are concerned. Further gains in speech coding are likely to accrue by incorporating deeper physiological aspects of the human speech production mechanism and characteristics of the speech single in the cider stru ctures. Towards this end we have done a dynamical analysis study of speech signals and explored some nonlinear representational form for productive coding of speech. The thesis documents our investigation of a nonlinear framework for speech signal coding .The complete study can be classified as an investigation of three related problems . The first problem is to chose a sufficiently general framework for .The choice of a deterministic framework rather than a stochastic lines is becau se of our interest in modeling the time waveform itself instead of its statistical moments .The motivation for using a detrrerministiuc state space framework is due to the recent advances in the understanding and characterization of the comp lex behaviour of deterministic chaos in dynamical systems . Viving complex time series behaviour as arising out of low dimensional chaos gives a new tool for analyzing and modeling it deterministically .In this framework the sp eech time series is embedded in a reconstruction state space as a reconstruct ional trajectory .We have done a detailed analysis of the reconstructed trajectories of unit articulation of speech namely phonemes in terms of dynam ical attributes such as dimension metric entropy and Lyapunov exponents .Just as a correlation analysis helps in a linear modeling exercise these dynamical attributes help in building nonlinear deterministic state space models .A s the second problem we study and compare with liner prediction the performance of some nonlinear state space based predictive models for speech .We have also implemented and carried out preliminary performance tests of a local state predication based low to medium delay speech coding scheme in the medium bit rate range . the

third problem addresses a related question of estimating the minimum rate at which information about a source can be trans mitted to the user subject to the condition that it can be reproduced with a specified ditortion function for stationary ergodic sources with memory . Bathe discrete and continuous alphabets sources are considered .Finally we u se this algorithm to compute the lower bound for quantized speech sources. In the following we give a chapter wise summary of the thesis. Chapter 1 begins with a contextual review of speech coding. Thereafter we build a case for th e study with a contextual review of speech coding. Thereafter we build a case for the study of nonlinear analysis and modeling of speech in terms of (a) observation from the speech production mechanism (b) observations from the speech signal (c) limitations of a linear model and (d) advances in nonlinear analysis and modeling techniques. We give a qualitative discussion of the notions of randomness determinateness and predictability in deterministic dynamics particularly in the light of chaos theory. A brief discussion of the three problems in the thesis is given next. These are (I) nonlinear dynamical analysis of speech signals (ii) state space predictive modeling of speech and (iii) computation of a lower bound of the rate distortio n function for stationary ergodic sources with memory .We also recorded in A historical note the recent investigations using tools from nonlinear dynamics for speech signal analysis and studies in non -linear predictive modeling and coding of speech. In chapters 2 and 3 we are concerned with nonlinear dynamical analysis study of speech signal. Chapter 2 begins with a discussion of the theorems that from the basis for reconstructing a state space trajectory from a scalar observable of a dynamical system evolution such that the dynamical invariants obtained from the reconstructed trajectory will be the same as those of the original dynamical system. We discuss two methods for optimal state space reconstruction based on sing ular b value decomposition and radiance criteria and use them to reconstruct speech trajectories and make observations .We also give results of the computation of the largest Lyapunov exponent of reconstructed trajectories of phenomena articulations. Lyapunov exponents space trajectory. The asymptotically categorize bounded trajectories into equilibrium points periodic and quasiperiodic data .From the results and comparison tests we conclude that and quasiperiodic data . From the resu lt s and comparison tests we conclude that reconstructed speech trajectories exhibit exponential divergence on the average. In chapter 3 we give results of the computation of two dynamical invariants namely the correlation dim ension and second order dynamical entropy of speech . The notion of dimension in dynamical systems is associated with the number of degrees of freedom that a system possesses. The dimension attribute of a time series is helpfu l in a deterministic state space modeling exercise because it givers the necessary and sufficient number of independent state space variables needed to model the data .A large dimensionality means that the trajectory is complex an d has numerous degrees of freedom in which case a random process model may be a better choice .as study of various phenomena categories shows that speech is largely a law dimensional signal . We have also computed the correlation dimension from a simplified statically model of particular vowel utterance .The dimension results are qualified with a study of the various sources of error affecting the estimates . The second dynamical invariant in switch we interested in chapter 3 is the metric entropy which quantifies the rate of loss of information about the imitational state of a dynamical system as it evolves in time .Its relevance in the average time duration for which a dynamical system or a time series model) can be predicted from a given initial condition .We have computed the second order dynamical entropy of speech which is a lower bound of the metric entropy . The positive values of the second order entropy and the largest Lyapunov exponent (chapter 2) for phoneme articulations both give evidence of the average divergence of nearby speech trajectories of speech in the reconstructed state space . Base on the dynamical analysis results we have investigated some nonlinear predication schemes for speech signal modeling and coding in a state space framework in chapter 4 and 5 .Chapter 4 begins with a review of the salient features o f the analysis – by - synthesis class of linear predication coders and in particular the CELP coding scheme .We give some model based analysis results which make a case for nonlinear modeling of speech .Thereafter we study the performanc e of some nonlinear representation forms for predictive modeling of speech in a state space framework .There are two basis scheme for predictive modeling of speech in a state space framework . There are two basic schemes in this framework .These are the global and local prediction schemes. In a global prediction scheme which we study in this chapter the function parameters are optimized over the entire state space .As this chapter the function pa rameters are (quadratic) polynomial representation form .The basis of comparison with short term Linear Predication (LP) in terms of segmental prediction gain is the number of coefficients in the two predictor model .We have princip ally considered two ordering schemes for selection of model upto a certain quadratic predictor . In the first method we exhaust all possible terms upto a certain time lag before considering terms which include signal depen dence for greater lags . The second method is based on orthogonal terms selection from a set of candidate terms . While the first method does not perform as well as a short term LP in terms of segmental predication gain the second method gives a modest improvement over LP FOR THE Same number of model coefficient s .In another study quadratic predictors exported recently the basis of comparison with LP is the time delay upto which signal correlations are considered rather than the number of model coefficients. In this case the performance of the former is significantly better in terms of sequential predication gain . In chapter 5 we investigate a Local State Predication (LSP) s cheme for speech . In this scheme the representation form is optimized over a local volume in state space where the prediction is to be done . the scheme is studied in terms of segmental predication gain plots of the predicti on error sequence their spectrum and autocorrelation function and is compared with the error sequence resulting from (I) short term LP and (ii) short term plus long term LP . In LSP an appropriately chosen neighborhood of a target point in the reconstructed space will contain trajectory points that are close to it in time as well as those which are approximately an integral number of pitch or formant periods away. Thus a LSP attempts to simulate the functions of b oth short terms and long terms linear predication simultaneously . Not that in the LSP scheme studied by us local neighborhood is chosen from an analysis farm length of previous data values .The performance of local linear predi cation scheme in the above terms can be broadly categorized as lying between short term and short term plus long terms LP (where both the prediction are done in forward adaptive mode) . We have done a preliminary study of a framework for low to medium delay speech coding in the medium bit rate range based on LSP.It is an analysis – by -synthesis coder operationally similar to CELP and tentatively named as a vector Excited Local State Prediction (VELSP) coder .The following points highlight the coding scheme and bring out the differences with CELP: (i) LSP is performed instead of L.P The LSP is performed using previous reproduced speech which is available to the decoder as well . (ii) A single exception codebook designed from empirical data is used instead of two separate codebooks as in CELP taking advantage of the predication property of LSP. (iii) A LSP based coder is naturally suited to low delay coding . (iv) S ince a LSP based coder is nonlinear we give a method to incorporated factor . We have implemented and studied a basic form of the coder structure at rates of 5.2 and 8.0 Kbits /s having theoretical coding delay of 7.5 ms,6.0ms and 4875ms respectively .While the segmental SNR performance is similar to CELP the reproduced speech has perceptible noise and becomes poor at the 5.2

Kbits/s rate .The preliminary investigation suggests that the VELSP c oding scheme can be a candidate for a more detailed study in terms of (a) improving the adaptive post filtering depending on a detailed study of the nature of reproductions error etc., and (b) reducing the computational complexity of the coder and decoder . In chapter 6 we give an application of the computation of a lower bound RL(FD) of the rate distortion function R(D) for stationary ergodic sources with memory . The relevance of the rate distortion function R(D) in the context of signal comparison is that out gives the minimum rate R at which information about a source can be transmitted subject to the constraint that it can be reproduced with an average and RL (D) = R1 (D)+ h - h(X) for continuous alphabet sources where R1(D) is the first order rate distortion function , H(X){h(x)} is the entropy {differential entropy } of the source X based on the marginal probability density of the source X and H {h} is the entropy rate {differential entropy rate } of the source X based on the marginal probability density of the source X and H{h} is the entropy rate {differential entropy rate } of the source X. The estimation of t he rates H{h} becomes difficult as statically dependencies for larger time frames are rates H{h} becomes difficult as statically dependencies for larger time frame are successively considered .We give procedures to estimate the second order entropy rate H2 (H2 $\leq$ H) and the second order differential entropy rate h2(h2 $\leq$ h) using a method I of generalized correlation sun which is conjectured to g ive better estimates than the histogen technique . The procedures are based on give better estimates than the histogram technique .The procedures are based on extensions of a method to estimate the metric entropy that has become standard in dynamical systems literature ion the last ten years .We give example s to show the dynamical of this estimation scheme .We also compute the lower bound of R(D) with respect to the mean square error distribution crite rion for quantized speech sources of resolution 6,8 and 10 bits/ sample . Finally we conclude in chapter 7 by summarizing the contributions of the thesis and nothing some directions for further investigations in these Areas

For more details click here Back

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Abstract

The overall objective of this thesis may be declared as a generalisation of signal and system theory. We contend that the fact that the theory in its current form relies on the use of highly sophisticated mathematical structures such as Hilbert spaces detracts considerably from the wider applicability and potential that its central concepts possess. We therefore proceed to consider a framework that is general enough to construct a more mathematically primitive formulation of signal and processor theory. An important decision made at this stage is that the formulation will be entirely carried out at the level of set theory. For this, we begin by defining, or rather postulating as axioms of the proposed theory, two sets to be called the domain D and the range C respectively. The domain and range sets will not be called upon to possess any more structure than that of mere sets by the theory itself, so that all the results obtained at this primitive level will continue to remain valid under situations where additional structure is imposed upon these two sets. A signal is then defined as nothing but a map from the domain to the range. Different signals then simply become different maps s : D ^ C, and the signal space S becomes the collection of all possible signals that may be thus defined. Next, a processor, conventionally understood as an input-output system that produces an 'output' signal for each impressed 'input' signal is directly abstracted as a map p : S i-> S that simply associates each s G S with some (possibly other) sf € S that is deemed the output corresponding to the input s. In the terminology of mathematics, the signals are functions, and the processors are operators.^ VI SYNOPSIS At this stage of the development, it becomes necessary to select some means of characterising-and thereby classifying-processors. Given the extreme generality of the formulation, the criterion used must be expressible purely in set-theoretic terms. We choose to characterise a processor in terms of the subsets SCSof the signal space that are closed under processing by the processor (we consider a subset S as closed under p if Vs €.& \p(s) £ S], The processor is associated, as a first level characterisation, with the" collection of all subsets with this property. It turns out that the collection so obtained forms a topological space with S as its ground set We call this topology on S the (first) preservance topology of p, after the fact that the topology consists of signal subsets 'preserved' by p. Along the same lines as the above, it is shown that a preservance topology may be defined for any p on the power set V(S) of S: however, for this, we will first need to define a new set theoretic relation we call subinelusion. This relation, indispensable for the definition of the higher preservance topologies (preservance topologies defined upon the power set of S, its power set, and so on) is shown to be a sort of diluted form of the conventional set inclusion relation C, and in fact reduces to it for the case of sets of unit rank Under the subinelusion relation C, we thus define for each p an entire sequence of preservance topologies (one for each positive integer) called respectively the first,- second,- third,- etc., preservance topologies of p. With this basic machinery in place, we embark upon a

project of generalisation of various well known concepts of signal processing theory, such as symmetry, linearity, invariance, etc. All these notions are formulated in purely set theoretic terms: the symmetry of a signal s with respect to a given domain transformation u (by a transformation on a set, we mean a bijective map upon it) is defined as its unalterability under that transformation, and the invariance of a processor p with respect to u as its commutation with the unique bijective processor that simulates u. Coming to processor linearity, we consider for our purposes that it consists only of what we conventionally call homogeneity (the property of additivity is not amenable to generalisation to a set theoretic framework): it turns out that homogeneity too, like invariance, can be modelled as a property of commutation of the processor in question with the unique bijective processor that simulates a certain range transformation. The important achievement about these generalisations SYNOPSIS VII of linearity and invariance, however, is their successful expression entirely in terms of certain topological properties, thus integrating the generalisations into our common mathematical framework of general topology. Introduced as a new analogue of the concept of signal symmetry is the concept of signal stasis, which is formally defined as the unalterability of a signal under a range transformation v. Several properties of symmetry and stasis are stated, such as the fact that the set of symmetries and the set of stases of a signal are both groups under composition. In addition, a study of the effects on a signal of simultaneous domain and range transformations will reveal the constancy of signal symmetry under arbitrary range transformations and the constancy of signal stasis under arbitrary domain transformations. Next, a new principle for the study of processors, called locality is introduced: it refers to the property by which the value of the output signal p(s) at a point d on the domain is affected only by the values of the input signal over a certain subset D of the domain. In other words, the output value at dr p(s)(d), would remain unchanged if (the values of) the input signal s were altered at points outside of Z>. The wide applicability of this principle and its physical significance are demonstrated: for example, it is shown that causality as defined in conventional theory may also be considered as an instance of locality. A function called the vista function that describes the locality properties of a processor is defined and used as the basis for further investigation. Where the vista function of a processor p possesses certain properties with respect to a certain domain subset D, we call the subset an isolation for p. The importance of isolations for parallel computing of the output signal is mentioned. The next study taken up is that of restricted signal processing systems. Restricted systems are shown to be entire signal processing systems in their own right, consisting of restricted signals and restricted processors. The restricted signals can be seen as projections of the primary signals into the restricted system: every primary signal is shown to be represented in the restricted system. Similarly, the restricted processors are shown to be projections of primary processors into the restricted system: but here it is found that only those primary processors for which the domain of the restricted system is an isolation are represented in the restricted system. A study of the linearity, viii SYNOPSIS invariance, symmetry and stasis properties of the entities in the restricted system, and the relation they bear with the corresponding properties of the entities in the primary system (of which they are the projections) is entered into. Finally, a set of results are obtained that relate the locality analysis of processors to the processors' topological characterisation: this integrates the locality theory with the mainstream topological theory. Next is attempted a generaEsation of a specific notion of stability we call range stability. Here, it is shown that the stability concept can be expressed even in the absence of a metric on the range set all that needs to be specified is a set of range subsets with certain closure properties to be called the neighbourhood profile. Entities we call tubes of signals are now defined: these are collections of signals deemed 'close' to one another under the particular neighbourhood profile chosen for the analysis. The closeness of signals under a metric will be shown to emerge as a special case of this general formulation. A processor is then defined to be stable if it takes input signals 'dose7 to one another to output signals also close to one another (under the same neighbourhood structure). Various properties and consequences are studied at length We generalise the principle of signal quantisation to a set theoretic leveL Signal quantisation is viewed as a consequence of a quantisation of the range set onto which the signals axe defined, where the range quantisation is effected by an idempotent map from the range to itself. The concept of a quantised system consisting of quantised signals and quantised processors is introduced; as with the case of locality7, it turns out that while all primary signals are represented in a quantised system, not all processors in the primary syscem are represented. The earlier theory of range-stability is now related to the question of quantisation by demonstrating that precisely those processors are represented in the quantised system that are stable in the range partition induced by the quantiser. Thus the construction of quantised systems is shown to be dependent upon the existence of processors that are stable under the quantisation. It is shown that the stability property and its connections with the signal quantisation problem can also be expressed in terms of the topological properties of the processors concerned. A very brief study attempting to connect SYNOPSIS IX various properties of the quantised entities with the primary entities of which they are the projections is taken up. Finally, a wider generalisation, based upon the cases studied, namely, invariance, linearity, locality and stability, is effected. This generalisation is shown to become possible because of the fact that at a purely topological level of discourse, these otherwise very different concepts have very similar descriptions, namely, as partition subtopologies of the second preservance of the processor in question, and the fact that partition subtopologies of this kind can occur under a very much larger class of situations of which those studied were mere instances. The final major subject dealt with in the thesis is that of signal representation and approximation. We set up a so called eigensystem which is a set of secondary signal processing systems into each of which all the signals and (some of the) processors in the primary system will be projected. The representation of any signal in an eigensystem will be called its spectrum and will be the set of its projections into each of the secondary systems making up the eigensystem. This spectrum is shown to actually be a primitive prototype of its conventional refined counterpart, and the eigensystem is likened to a basis of a vector space. The notions of subspace, of independence, of completeness, of filtering, etc., are all shown to be meaningful in the generalisation too. In all this, a convenient reinterpretation of the conventional concept of signal approximation is carried out in order to enable its inclusion in our primitive framework. It is shown that this is indeed a generalisation of the former, and that it does not require to possess any more than a set-theoretic structure to be applied. The classical theory of linear, time-invariant processing is now shown to derive froin the general fheory when the usual structural assumptions defining LTI systems are made for it In the process, an explanation, made out entirely in the language of our theory, of the central role played by complex exponentials in LTI theory also emerges. The thesis ends with a few final remarks suggesting the possibility of generalising the notion of an eigensystem even further; The concluding Section proposes several avenues for future research opened up by the formulation originated by the thesis

For more details click here Back

Title : ***Collision Resulution Algorithms For Finite User Blocked Random Access Channels***

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Supervisor(s) : ***Chatterjee PK& Rao PRK***

Abstract

Address the problems of optimal random access algorithm for a finite user mode by studying only blocked random access algorithms and focusing on their collision aspect. The common receiver model has been considered. The dynamic programming technique has been applied to derive an optimal collision resolution policy assuming known number of active users and an approximate way suggested to calculate the average collusion resolution length the performance of the algorithm has been studied by simulations after remaining the banging to be known with different initial probability distributions besides the access of users with infinite buffers and random access systems with multipacket reception capabilities have also been studied. It has been shown that the problem of optimal collision resolution can be considered within the framework of partially observable Monrovian precision process. The sequence of transmission epoch to be a Markov chain. The protocol has been compared with TDMA scheme in terms of thought put and resolution factor and the packet delay has been shown to consist of there components the expression for which have been derived. Transmitter based code assignment scheme with a multibit feedback broadcast by the common receives has been receiver model of the finite user random access system.

For more details click here Back

Title **:** ***Queueing Analysis Of ANon-Preemptive MMPP/D/1/K Priority System For Applications In ATM Networks***
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Abstract

In the Asynchronous Transfer Mode (ATM) networks, incorporation of service priority is desirable for certain real time process control, alarm and network control applications It has been found that the incorporation of service priority also improves the loss performance of the ATM nodal buffer and reduces the output port contention m an ATM switch with input buffers [1], [2] In the ATM context, the analysis of a queueing system with muliple service priorities has been reported only for simple input models in the literature, eg for Poisson and Bernoulli models For example, in [1], Gravey et al assume two priority classes for the traffic arriving at an ATM switch and use a non-preemptive M/D/1 priority model for obtaining the queueing delays for these classes However these models may not be completely satisfactory for the bursty sources encountered in ATM networks and more complex models, such as Markov Modulated Poisson Process (MMPP) models, may be more desirable Assuming the MMPP model and a FCFS discipline, several aspects of Call Admission Control (CAC) and nodal buffer design in ATM networks have already been considered m the literature [3] These approaches do not, however, account for multipriority traffic as results on the queueing model for such traffic are not available for MMPP sources The contribution made by this thesis is m presenting such a queueing model Specifically, we present in this thesis the queueing analysis of a multipriority, non-preemptive MMPP/D/1/K system with either infinite or finite buffers for the individual priorities, where the input processes may be MMPP in nature This analysis is carried out using the matrix anal3rtic approach Computational results are presented and the approach IS verified by comparing these numerical results with those obtained through simulationsSynopsis uiWe study a dual priority system first, under the following assumptions -1 The delay sensitive cells and the non-delay sensitive cells arriving at an ATM multiplexer are buffered at two separate queues Ql and Q2 The arrivals to Ql and Q2 are from two independent M (a:2), N (2:2) phase MMPPs2 A single server is shared between the two queues The cells at Q2 have non-preemptive priority over those at Ql for receiving service The priority is incorporated at call level The server is asynchronous, le a cell arriving when the queue is empty receives service immediately3 Cells from each priority class require a constant service time of D secWe use the words "cell" and "customer" interchangeably as the basic unit of information transferred in an ATM network is a cellUsing these assumptions the computation of the queue length density (QLD) at Ql and Q2 at the departure epochs of customers from the respective queues (viz the probability that the no of of customers in Qi is equal to 0, 1, at a departure instant from Qi for i=l, 2) is considered first For the application of the matrix geometric approach for the present problem, some generalizations of the approach used for the single priority system in [4],[5] are proposed and are as follows1

The mter-departure time of customers from the low priority queue depends on the phase of the arrival process to the higher priority queues and hence it should be treated as a vector random variable2 In the priority system, the time when the first customer arrives at an empty queue and the time when the busy period of the server starts need not be identical In view of this, the busy period of Qi (for i=l, 2) is defined to be the time that elapses since the beginning of the service for the first customer arriving at Qi and the time when Qi becomes empty againThe analysis approach can be summarized as follows We choose two Semi-Markov chains (SMC), one each corresponding to each priority class with the embedded points as the departure instants of customers from the respective queue The transition probability matrices of these SMCs are denoted as Q'(t) and Q"(t) respectively The stationary vectors of Q'(oo) and Q"(oo) give the requiredSynopsis ivQLDs at Ql and Q2 In view of the non-preemptive priority, the matrices Q*(oo)and Q"(co) are coupled and hence the invariant vectors have to be obtaineditherativelyQ'(t) and Q"(t) are expressed in terms of two sets of MNxMN matrix massfunctions [A'(t), B'(t)l and [A"(t), B"(tJ] respectively Evaluation of•¦m m** ""m m*'these matrix mass functions are considered next and it requires the knowledge of the busy period distribution of Q2, the counting functions associated with the MMPPs to Ql and Q2 (these functions quantify the probability of there being n arrivals in an interval of time along with the phase transition of the underlying modulating process), the probability that Ql, Q2 is empty at an arbitrary time instant and the QLD of Q2In order to compute the above matrix functions, evaluation of some of the characteristics of the busy periods of Ql and Q2 are considered Exploiting the fact that the service time/cell, is constant a recursive procedure for the computation of the busy period distribution is developed Evaluation of the counting functions associated with the MMPPs requires numerical integration of some differential-difference equations for a general service time distribution [4] Exploiting the fact of constant service time/cell an alternate, computationally efficient recursive procedure is proposed for computing these functions using infinite series expansion The convergence problem associated with this series at high traffic rates are overcome by computing them m two stages Expressions for the average number of customers served and the average duration of the busy period at Ql and Q2 are also obtainedNumerical computation of the QLDs are considered next Towards this end the following issues are considered m detail first1 Computation of the probability of finding zero, one cell at the departure instant of cells from Qi (for i=l, 2) using first passage time arguements2 Computation of the probability of Qi being empty at an arbitrary time t3 Evaluation of the moments of the queue lengths at Ql and Q24 Details and the relative advantages of the computation of the QLDs of Ql and Q2 using (i) Gaussian elimination method (u) Block Toeplitz inversion method (in) Recursive procedureAt high traffic loads at Ql, the computational and storage complexity required for the evaluation of the QLDs becomes prohibitively high Under this condition the practical buffer sizes used for Ql may not be large enough to be treated to be infinite Hence the computation of the QLDs of Ql and Q2 when QlSynopsis vis finite sized is considered next As an alternative, computation of the QLD of 02 and the moments of the queue length at Ql without evaluating the QLD of Ql is also consideredIt may be noted that the computational and storage complexity required for the evaluation of the QLD of Ql and Q2 under the priority system is increased2 2by a factor of 0(N) and 0(M) over that of the corresponding single priority system This is because in the priority system the phases of the MMPPs to both Ql and Q2 need to be tracked at all departure instants An approximate model which IS computationally and storage wise efficient is proposed next This model keeps track of the phase of only one of the MMPPs at a time Using this model, evaluation of the QLDs of Ql and Q2 when the inputs to both Ql and

Q2 are approximated by Poisson processes, is also considered Finally, some details on the evaluation of the busy period distribution and the QLDs using simulation is consideredResults on the computation of the QLDs of Ql and Q2 using the exact model (model I) and approximate model (model II) are presented next for a number of examples and compared with those obtained using simulation For numerical computations. We assume the traffic to Ql and Q2 to originate from Nl Type i on/off sources and N2 Type j on/off sources, respectively (i=j impliesidentical type of on/off sources to Ql and Q2) . An output link of 150 Mbps and a cell size of 53 b3rtes are also assumed The traffic to Ql and Q2 are approximated by two 2 phase MMPPs using the method proposed m [6] Knowing the MMPP model parameters, Q'(cx)) and Q"(oo) are then found and the QLDs are obtained iteratively Due to resource constraints, the evaluation of the QLD IS considered only for cases where the traffic offered to Q2 is less than or equal to 0 35 For cases where the high priority load is greater than 0 35 a finite capacity non-preemptive MMPP/D/1/K priority system is suggested for the evaluation When the total traffic offered to the server is close to the capacity of the server the computational and storage requirements become high and hence m these cases Ql buffer size is assumed to be finite Based on the examples considered the following conclusions are drawn1 The QLDs of Ql and Q2 obtained using the exact model agrees well with thoseobtained using simulation in all the examples considered2 It appears that model II should not be used if either of the two conditions, (a) or (b), are true - in these cases, the model I is recommended for computations Otherwise, model II may be preferred due to its simplicitySynopsis vi(a) if K/^2 ^^ significantly larger than 1, where X" denotes the arrival rate of the MMPP to Q2 in the ith phase and are labelled such that K>K(b) there is a particular phase pair of the two MMPPs which tends to overload the server and this phase pair is fairly likely to arise3 The QLDs of Ql computed using model II agrees with those of model I at low queue lengths even when the conditions (a) and (b) are true Because of this the probability of Qi being empty at an arbitrary time instant computed using model I and II turns out to be essentially the same4 The QLD of Q2 computed using all the three methods agree under all conditionsFinally, the results on the computation of the QLDs of Ql and Q2 obtained by assuming the traffic to be modelled as Poisson process, are presented for some typical examples In this case, the QLD of Ql agrees with that obtained using model II The QLD of Q2 differs from those of model I and II at higher queue lengthsNext, the expressions for the distribution of the virtual waiting time of a customer arriving at Q2 and its Laplace Steiltjes Transform are obtained Using these results, the average queueing delay at Q2 as well as that in Ql are obtained Extension of these results for the approximate model as well as the degenerate case of non-preemptive M/D/1 priority system are considered For the examples considered earlier for the evaluation of the QLDs, the average queueing delays at Ql and Q2 are computed using the exact as well as approximate models and are found to be m agreement with the results obtained using simulation For the M/D/1 system, the average queueing delays are computed and are found to be in agreement with the results obtained using an alternate approach given in [1] The expression for the LST of the virtual waiting time distribution also enables the computation of the percentile of the queueing delays at Q2 jNext, the computation of the QLDs of a non-preemptive MMPP/D/1 dual prior ity system with finite capacity at Q2 is considered The busy period distribution (BPD) of the finite capacity system differs from the infinite capacity case as follows1 Customers arriving when the buffer is full are denied service2 The maximum number of customers that can be admitted into the system during the service time of a customer depends on the empty space in Q2 BecauseSynopsis viiof this the distribution of the first passage times are not identical anddepends on the state of Q2 As in the infinite case, a recursive procedure Tor the computation of the BPD of Q2 is developed,using the fact that the ser-vice time/customer is constant Computation of the BPD for the finite capacity case requires considerable computational effort compared to that of the infinite capacity case This is because, m the present case to compute the BPD of Q2 of capacity >V, the BPD of capacity of 1, 2, . jV-1 should be computed lirst An indirect method for the efficient computation of the BPD of finite capacity Q2 is proposed Modifications of the equations of the infinite capacity system for the present case are discussedThe busy period distribution at Q2 is connputed for some examples using both the direct and indirect methods The latter inetliod is found to require 50% less computation time The BPD numerically computed is also compared with the simulation results and is found to match well For some typical examples of traffic from on/off sources, the QLDs at <51 clM Q2 are evaluated using the exact model as well as the approximate model a_iid compared with those obtained using simulation The conclusions draw^n for the infinite capacity is found to be valid for this case as well. Computation ol the average queuemg delays is also considered for the finite capacity caseFinally, the computation of the QLDs and thie queueing delays of a non-preemptive MMPP/D/1 priority system with more than two priority classes are considered The traffic from each priority class is assumed to arrive at separate queues and demaind the same service time of D sec /customer The computation of the QLD of a triple priority system is considered first and the extension of this result for higher number of priority classes are indicated The computation of the average queueing <lelay also proceeds in a similar fashionThe major contributions of this thesis are suamarized as follows1 Generalization of the matrix anal3?tic appr^oach for the study of a non-preemptive MMPP/D/1/K priority system ^v^ith cither finite or infinite buffer space under the assumption that the traffic from each priority class arrives at separate queues2 Evaluation of the queue length densities, moments of queue lengths and queueing delays at the queues correspondm^ to each priority class as wellSynopsis viiias the percentile of the queueing delay at the high priority queueDevelopment of efficient recursive procedures for the computation of thebusy period distribution of the server m each of the queues when thebuffer size is either infinite or fmiteDevelopment of an efficient procedure for the computation of the countingfunctions associated with the MMPPsSuggestion of two computationally and storage wise efficient approximatemodels for the priority system and investigation of the range over whichthey are accurate

For more details click here Back

Title : AC To DC BI-Directional PWM Voltage Source Converters
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Abstract

AC and DC converters have been extensively used in various industrial applications. The necessary bi-directional power conversion has been done using primarily line commutated thyrisor converters which have remained as a major work force for the industry over two decades. These converters have number of known disadvantages such as (i) low frequency harmonics developed at the source terminals, (ii) low power factor of operation especially at large firing angles, and (iii) low frequency harmonics or ripple in the output dc voltage. Use of higher pulse constructions (12-pulse or 24-pulse) was suggested in past to improve the converter performance. Further, forced commutation methods were developed to improve power factor and to reduce harmonics on ac and dc side. These methods employed some kind of Pulse Width Modulation (PWM) which had to be used at low frequencies corresponding to low switching frequency ratings of thyristors. This resulted in bulky commutation circuits and increase in losses causing increased volume, weight and cost of the forced commutated converters and hence limited their acceptance. Availability of self commutated devices like power transistors, MOSFETs. IGBTs and GTOs renewed interest in PWM ac to dc converters. These devices do not require any forced commutation circuits and hence offer considerable reduction in converter losses, volume, weight and cost. Due to high switching speeds of the first three devices mentioned above, harmonics on both ac and dc side of the converter can be considerably reduced. However, due to low voltage and current ratings, usage to these devices is restricted to low power applications. GTOs on the other hand, with their high voltage and current ratings, are fast revolutionizing PWM converter designs for medium to large power applications. This is inspite of their low switching frequency ratings (less than 2 kHz). PWM techniques, as applied to ac to dc power conversion and using self commutated devices, have given rise to two basic topologies for ac to dc converters. These are, namely, current source topology and voltage source topology. The converters are subsequently termed as Current Source Converters (CSCs) and Voltage Source Converters (VSCs). The work presented in this thesis deals with ac to dc bi-directional power conversion using GTO based Voltage Source Converters (VSCs). Specific areas covered are (1) reduction of harmonics in supply currents (2) two, three and multilevel converters (3) improvement in utility of Indirect Current Control (ICC) technique (4) multipulse converters for static VAR compensation and traction applications. Reported work, chapterwise, is given below. Chapter 1 gives past as well as recent development in the field of bi-directional ac to dc power conversion and introduces Voltage Source Converters (VSCs). It also gives the chosen objectives of thesis work which cover the above mentioned specific areas of interest and belief outline of the thesis. Chapter 2 gives review of developments mainly connected with

VSCs. It covers different types two level and three level converter configurations, harmonic elimination techniques, current control techniques and multipulse approach especially used in static VAR compensation schemes. Concept of using a parallel operating converter to selectively eliminate harmonics generated by a GTO based VSC controls bi-directional power flow. The parallel operating VSC does not handle nay active power but handles only harmonic power. Three different techniques are presented to illustrate selective elimination of harmonics produced by the GTO based VSC in supply currents. Since the parallel operating VSC needs to handle only harmonic power, it makes use of high switching frequency devices like power transistors, MOSFETs or IGBTs allowing considerable reduction in the harmonics. Harmonic elimination technique using superimposed harmonic signals on a fundamental modulating signal is presented in chapter 4. using the technique, harmonics are eliminated for a given range of modulation index Mi in a sinusoidal PWM controlled VSC. Necessary equations controlling amplitudes of the superimposed harmonic signals are derived based on Generalized Harmonic Elimination (GHE) technique given by Patel and Hoft. Chapter 5 deals with a harmonic elimination technique which is equally generalized in nature as the GHE Technique given by Patel and Hoft. The technique is hence called as Generalised Harmonic Elimination technique-II (GHE-II). The proposed technique does not require any initial guess for switching angles. Any suitable choice of switching angles could be used as a base. It also eliminates need to store the switching angles. Solutions to switching angles, while obtaining desired values for certain harmonics (low as well as high in number), are presented based on two level and three level switching waveforms. Possible improvements to realize harmonic values close to the desired values are also brought out clearly. Consolidation of single phase two level and three level VSCs is presented in chapter 6. equivalence is drawn between two series connected two level converters and a single three level converter alongwith proposed three level converters using center tapped and noncentre tapped output capacitor banks. Simulation of input supply current drawn by series connected two level converters and equivalent three level converters are also given. Further, equivalent economical configurations, producing multilevel input voltages waveforms, are suggested for series connected three level converters alongwith input supply current simulation. Experimental results demonstrating 360-degrees power angle range for two level and proposed three level converters are presented. Also presented are experimental results for series equivalence between two level converters and single three level converters as well as series connection of proposed three level converters at unity power factor operation. Chapter 7 deals with proposed three phase VSCs. Two economical three phase VSCs are suggested, one based on two level input voltage and other based on three level input voltage waveform. Both VSCs use λ-connected input voltage source and center tapped output capacitor bank. Experimental results demonstrating 360-degrees power angle range and open neutral construction are presented in both cases. Two methods which reduce complexity of the Indirect Current Control (ICC) technique are considered in Chapter 8. both methods use two converters in series. Current controlling voltage is thus obtained through summation of input switching voltages of the two converters. Reason to use two converters in series is to compensate either IXL or IZ voltage drop in the supply line separately. Method –1 which uses IXL voltage drop compensation was suggested earlier for Switch Mode Rectifier (SMR) application considering unidirectional power flow. Its suitability for, bi-directional power flow is confirmed here. Method-2, which is further proposed, uses IZ voltage drop compensation and has a better current controllability. Experimental results demonstrating controllable bi-directional power flow are presented for both methods. Chapter 9 presents multipulse waveform generation and its use in static VAR compensation and traction applications. Multipulse waveforms are generated using multilevel economical VSC configuration presented in chapter 6. voltage control is suggested using zero voltage "notch" angles. Five cases are investigated using two different 12-pulse waveforms to obtain a 24-pulses waveform with minimum harmonic. Based on this waveform. Economical schemes are proposed for static VAR compensation and traction applications. Comparison is also provided for a recently published/proposed scheme for a 100 MVA static VAR compensation application. Overall conclusions based on the work presented in the thesis are given in chapter 10 alongwith future scope of the work.

For more details click here Back

Title : Switch Mode Power Rectification
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Abstract

AC to DC power conversion has been mainly done using conventional diode rectifiers and line commutated thyristor converters. Conversion from fixed ac voltage to fixed dc voltage involving unidirectional power flow is required by power supplies. The power supplies are required in various industrial applications microprocessor based development systems, computers and computer based controls space equipments and military equipments. Such a power conversion has been done using primarily single phase and three phase diode rectifiers. These uncontrolled rectifiers have many disadvantages such as pulsed input currents causing distortion of the input supply voltage low power factor of operation ripple in the output dc voltage and low working efficiency. The passive current wave shaping techniques employing L, C or L-C filters, have been suggested for power factor improvement. These techniques could not gain wide acceptance because of the large size of input-output filter components required and only marginal improvement in power factor achieved. Recent advance in semiconductor devices, such as power transistors, GTOs and MOSFETs led to the development of new circuit configuration for ac to dc uncontrolled converters along with several active current wave shaping techniques. This thesis work investigates single phase and three phase ac to dc uncontrolled converters primarily using MOSFET as the switching element and some active current wave shaping techniques. An outline of the work reported in this thesis is given below. Chapter 1 introduces various aspects studies in this thesis and gives the idea about the thesis work in brief. Chapter 2 deals with the review of reported work in literature on single phase and three phase ac to dc uncontrolled converters using passive as well as active current wave shaping techniques. The passive current wave shaping techniques improve the performance of uncontrolled converters only to a limited extent. The active wave shaping techniques have considerable advantages in this respect. There fore the study of available power circuit configurations using these techniques, their component requirements and input-output filter requirements has been conducted. Reported active current wave shaping techniques are also studies. Three phase uncontrolled converters with passive and active current wave shaping techniques have also been reported in recent past. A review of these has also been done. Chapter 3 deals with development of new single phase switch mode Rectifiers (SMRs). Eight new single phase SMRs configuration are proposed analysed and their performance is compared with the standard SMR (single phase diode rectifier with a boost inductor a boost diode a boost switch and a capacitor bank on dc side). Comparative evaluation is done on the basis of number of components and power loss in the SMR. Experimental verification is provided for four of the SMR configurations. Latter part of this chapter deals with active current wave shaping techniques. Hysteresis current control (HCC) techniques is a

commonly used technique. It is reliable gives fast dynamic response and uses a simple closed loop control scheme. A detailed study of this technique is hence carried out along with experimental tests. Two current control techniques making use of HCC are proposed. Digitally simulated performance results are provided for these. Two new current control techniques making use of Multichannel SMR (MSMR) are introduced. Simulated results and experimental tests are provided for these techniques. It is noted that the HCC techniques has a drawback of random (variable) switching frequency. The switching frequency varies with load current and hysteresis. A such two applications of constant switching frequency operation using MSMR are considered. Simulated results are given for these. Comparative performance is evaluated for all the techniques considered. Chapter 4 deals with bang current control techniques using predicided switching frequency. The control technique avoids random switching frequency behaviour of HCC. The need for current sensing is also avoided. A fixed relation between switching frequency and load current is achieved, which helps in developing a simple closed loop control scheme. The switching frequency is constant for a given load current and varies in inverse proportion to the load current. Though the switching frequency varies with load current its variation with the load current is known. The nature of input current obtained is similar to that of bang HCC. The analysis and simulated performance results are provided. The feasibility of the proposed technique is experimentally verified. Chapter 5 deals with the SMR based on indirect current control (ICC) technique. The basis ICC technique is modified and a new SMR configurations suitable for modified ICC is proposed. This SMR compreises two converters connected is series namely an auxiliary converter and a main power converter. The auxiliary converter consists of two MOSFETs andtheir intrinsic antiparallel diodes while the main power converter is a single phase diode rectifier with dc side boost switch. The conventional sinusoidal pulse width modulation (SPWN) method is used which gives a constant switching frequency operation. In comparison with HCC technique the modified ICC technique has an advantage of elimination of the current feedback and hence the current transducers. The control circuit and power circuit for realizing the modified ICC technique are developed and the SMR is experimentally tested. Chapter 6 deals with discontinuous current conduction of single phase and three phase SMRs. A new current control criterion termed as equal area criterion (EAC) is developed. This criterion uses constant switching frequency operation. It gives same area for the discontinuous current pulse as that of the area under a sinusoidal reference input current in each switching cycle. This yields fundamental components of the input current close to the reference current. Single phase SMR can be operated by implementing EAC with variable turn on time of the switching device while three phase SMR needs to be operated with constant turn on time. This criterion gives a fixed relation between load current and input fundamental current. This relation also helps to obtain a simple closed loop control scheme. The discontinuous current operations, however results in high current stresses on switching devices which is its disadvantage. The analysis and digitally simulated results are provided for single phase and three phase SMRs. The performance is evaluated for low as well as high switching frequencies. The medium to high power applications by using devices such as power transistors or GTOs. The low to medium power applications can be realized with high switching frequency by using MOSFETs. The experimental test results are provided for the EAC based three phase SMR with constant turn on time at low as well as at high switching frequencies. Chapter 7 deals with the analysis and design of two new three phase SMRs, termed as SMR1 and SMR2. These SMRs operate with continuous current conduction and avoid the drawbacks of three phase SMR based on discontinuous current conduction. SMR1 comprises single three phase diode rectifier with six switching devices connected on input ac side. SMR2 comprises two numbers of three phase diode rectifiers each with a single boost switch connected on dc side. It can be series or parallel operated. The supply voltages are provided from a λ / λ and Δ/ λ double transformer construction. SMR1 allows 1800 conduction range while SMR2 allows 1200 conduction range of the rectifier diodes. Both the SMRs can operate with HCC. Other constant switching frequency control techniques (CSFCTs) can be considered though not studied. Simulated performance and experimental results using constant percentage HCC technique are provided for the SMR1, while analytical results are provided for SMR2.

For more details click here Back

Title : *Low Cost And Concurrent Algorithms For Adaptive Volterra And Linear Filters*

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Abstract

The present work addresses the problem of adaptively estimating the coefficients of a finite and discrete volterra or linear filter. It is assumed that the structure of the filter is known and the input signal and a noisy estimate of the desired signal is provided. There is a vast amount of literature [1, 12, 22, 35] for the adaptive estimation of the coefficient of a finite and discrete linear filter. However, the literature in the case of volterra filters has not yet reached the same level of maturity. The utility of the Volterra filter as nonlinear model for modeling various phenomena such as equilization, echo cancellation, adaptive noise cancellation system identification etc [21] has long been established. Even, then practical applications of adaptive volterra filters have been limited because of the absence of algorithms which can provide reasonable convergence rates at moderate computational costs. This provided the motivation for the present study. Recently there has been a significant contribution from Mathews et al [17]. They have reported a fast recursive least squares algorithms for volterra filters. The following observations provided the directions for the thesis: • The approaches currently used for adaptive filtering, for both linear and volterra filters, can be broadly classified into two families: least squares and stochastic gradient. The least squares family is characterized by high computational cost (particularly in the case of volterra filters where even fast least squares algorithms have relatively high computational cost) but rapid convergence, which is not significantly dependent on the properties of the input signal. The algorithms in the stochastic gradient family are relatively simple, have low computational but have slow convergence which is also sensitive to the Eigen value spread of the input signal. • The difference in the rates of convergence of the two broad families of adaptive filtering algorithms is large. Typically least squares algorithms converge an order of magnitude faster than the stochastic gradient algorithms. There is no family that can fill this huge gap in the difference between the convergences rate of the two families.• The typical computational cost of least squares algorithms is O(N2) while that of stochastic gradient algorithms is O(N) where N is the order of the filter. Although fast least squares algorithms, having a computational cost O(N) (in the case of linear filters) are known, they provide lesser savings in the case of volterra filters (in isfunction of the volterra filter under consideration) and in general, apart from having a larger and more complicated structure, their numerical stability is also suspect. • The problem of devising algorithms which have the potential of providing a trade-obetween convergence rate and computational cost has, as yet, not been addressed. Such algorithms will have the capacity to fill the large gap both in convergence rate and computational cost between the least squares and the stochastic gradient family.• Derivation of such algorithms will provide to be useful for the implementation of volterra filters in real time. Presently for real time applications of volterra filters, the stochastic gradient algorithms and least

squares algorithms have limited use becauof slow convergence and high computational cost respectively. What we need is algorithms whose cost is lower than that of least squares algorithms and which have a convergence rate that is better than that of stochastic gradient algorithms. One ay of achieving this is to derive an algorithms which is flexible enough to provide a trade-off-between its convergence rate and computational cost. Then depending on the problem at hand, we can choose some combination of cost and convergence rate which can fulfill our need on both the fronts. • A possible starting point for such an endeavor is provided by the fact that at every iteration apart from the instantaneous error and the input signal vector, stochastic gradient algorithms use only the information regarding the power of the input signal (to choose the step size) while least squares algorithms use the information provided by the complete correlation matrix of the input signal. In between these two extremes one can look for algorithms which use intermediate or partial information about the correlation matrix of the input signal for the purpose of updating their weight vector.• Such algorithms will be useful in the case of linear filters as well. Work Presented in the Dissertation The central theme of the thesis is the derivation and demonstration of the feasibility of a family of adaptive algorithms, for both volterra and linear filters which can provide new combinations of convergence rates and computational cost. It has been shown that the existing adaptive filtering algorithms can be viewed as special cases of this more general family leading to fresh insights into the inter-relationships of the currently known algorithms. The focus of the thesis does not include the issue of characterizing the general family in terms of its misadjustment performance, numerical stability in finite precision etc. We consider an arbitrary discrete volterra filters. The highest degree nonlinearity in the filter is taken as any positive integer r ($r \geq 2$). The number of terms in each degree of nonlinearity can be arbitrary. The number of terms which are shifted versions of each other within a degree can also be different. We give an algorithm to compute the overall order of the volterra filter i.e., the number of terms in the filter if the upper limits of the various summations defining the filter are specified. Then we present an RLS (Recursive least squares) algorithms to compute the coefficients of the filter adaptively. Utilizing whatever redundancy is present in a given filter in terms of how many terms are shifted versions of each other we give a fast RLS algorithm. An efficient method to update the coefficients of a transversal filter, if its order is updated is given. We decompose a volterra filter into disjoint and smaller filters such that each smaller filter can be treated as a linear filter in the sense that all the terms in its data vector are shifted versions of each other. Then we present an algorithms which is sub-optimal in the least square sense but which runs an exact least squares on each of the smaller filters. Each of the smaller filters can be operated in parallel. This algorithm is referred to as the parallel recursive least squares (PRLS) algorithm. A fast version of this algorithms is presented. The structure of the equation, solved by this algorithms solves is presented in relation to the normal equation which is solved by least squares algorithms at each step. Subsequently, the assumption that all the smaller filters can be treated as linear filters is relaxed. Then we compute a lower bound on the number of alternative ways in which the PRLS algorithm can be implemented given that the volterra filter has p nonlinearities. A nonlinearity is not defined in the conventional sense any sequence of data points that can be obtained by a shifting operation is defined as a nonlinearity. We compare all the alternatives in terms of their computational cost requirement and if concurrency is exploited, the computational time required per iteration. A cost and time function is derived to make these comparisons. Then we compare all the choices from the point of view of rate of convergence provided their misadjustments are not significantly different. We see that when the power of the input signal is low all the alternatives have the same rate of convergence and this is the same as that of the exact least squares. However, as the power of the input signal is increased the approximation involved in PRLS begins to show its effects. A trade-off feature comes into effect wherein the alternatives can be arranged hierarchically so that from one end to the other thee is a monotonic gradation in the rate of convergence and the computational cost. Generally an alternative with a higher rate of convergence has a higher computational cost. We show that at one end the PRLS degenerates into the conventional RLS and at the other end into a member of the family of stochastic gradient algorithms. In between it provides a whole range of algorithms which have their cost and rate of convergence intermediate to that of lest squares and the stochastic gradient families. We have done simulation studies of the various PRLS algorithms for different volterra filters and for a variety of input signal. The PRLS algorithm did not diverge in any of examples that were carried out. Heuristic arguments have been given to show why the PRLS algorithm in general converges. A method to implement the PRLS algorithm as a fast algorithm is given. Though the PRLS algorithm has been derived in the context of volterra filters we find that it is equally useful in the case of linear filters. We have shown that the results given earlier can be applied to linear filters. Since the structure of a linear filter is fixed, we have defined appropriate data and weight vectors, given the order of the filter to implement the PRLS algorithm. We treat the special case when the number of smaller filters into which the linear filters is decomposed is the same as N, the order of the parent filter. It has been shown that this algorithm can be interpreted as a stochastic gradient algorithm. We derive three new stochastic gradient algorithms based on the PRLS algorithm. These algorithms have reasonably good convergence rates even for ill-conditioned input signals. One of them has been shown to have advantages over LMS and MLMS in more than one respect. Then we return to Volterra filters again and show that within each nonlinearity a further subbranching can be done since each nonlinearity (as defined earlier) is essentially a linear filters. We refer to this algorithm as the nested PRLS algorithm. The utility of adaptive filtering algorithms based on QR decomposition has long been established. We have derived a PRLS algorithm using QR decomposition for linear and Volterra filters. This is referred to as the PQR algorithms. Then we extend this to a fast PQR algorithm. Since fast QR algorithms compute only the error of the filter, the derivation of the Fast PQR algorithm uses the structures of the unitary rotation matrix to first determine the a priori error and from it estimate the desired signals for the smaller filters. The fast PQR algorithm is derived using these desired signals. Finally we analyze the PRLS algorithm from the point of view of its potential to unify the family of least squares algorithms and the stochastic gradient algorithms. We derive the equation, the solution of which is provided by the PRLS algorithm. This equation is referred to as the generalized normal equation (GNE) because it also provides the equation solved by RLS (normal equation) and the equation which the LMS, solves, as special cases. In the literature the weight vector of the LMS algorithm is always expressed as a function of the weight vector at the previous iteration added to a correction term. However, using the GNE we can express the weight vector of the LMS in terms of the input and the desired signals (i.e., in a nonrecursive form). This also gives occasion to interpret a matrix appearing in the GNE as the generalized forgetting factor (GFF). In the case of LMS, GFF gives an explicit demonstration of the memory of the LMS algorithm. For the case of RLS this matrix reduces to, as required to a scalar which is less than or equal to one. We give recursive equation for the computation of this GFF. The usual condition for the convergence of the LMS algorithm have been re-derived starting from the GNE. We derive the weight vector in another form in which it is shown as the summation of the true weight vector and an error term. Taking the expected values, it has been shown that in general the estimate is biased. The properties of the correlation matrix corresponding to the PRLS algorithms have been derived. The properties of the matrix governing the propagation of the weight error vector have also been investigated. It has been shown that the norm of this matrix serves as a useful parameter to derive a new proof of the convergence of the LMS algorithm without making the usual assumptions of the stationarity and independence of the input signal. But for other cases this approach has been shown to be non-productive. An analysis of the new stochastic gradient algorithm, derived earlier, has been done. It has been shown that even for correlated input signals the algorithms is expected to be stable after a certain number of iterations. An expression for this number has been derived.

For more details click here Back

Title : ***New Decoding Algorithms For Reed-Muller And Cyclic Codes Based On Spectral Domain Deconvolution***

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Abstract

The main thrust of the thesis is to expand the range of codes decodable using spectral techniques . Two classes of sectral transformed chosen d for study are discrete Fourier transform (DFT) and Walsh - Handamard Transform (WHT) both important social cases pf GAENERALIZED Walsh Handamard Transforms .Using these classes of transforms decoding problems in the spectral domain are shown to be equivalent to appropriate deconvolution problems. Structural properties of the transforms and code s are used to obtain simple decoding (Deconvolution) algorithms. WHT techniques are applied for decoding of Reed - Muller (RM) codes of order 2 and 3 whereas DFT techniques are applied for decoding of different classes of cyclic codes. Design and implementation of good decoding algorithms has attracted lot of attention in the l9iterature .Since decoding speed puts a restriction on the overall through and a simpler decoder architecture reduces the cost of decoding smallest possible dec oding delay and simplest possible decoder architecture are the two important objective of a good decoding algorithm . Achieving these objective of reduced delay and implementation simplicity depends primarily on the mathematical structure of the code. In some cases the structural symmetries directly suggest a simple implementation scheme . Threshold decoding scheme for the class of Reed - Muller codes and Meggit decoder scheme for the class of cyclic codes may be seen as example s i n this category .In other cases it may be necessary to seek alternative representations of codes for obtaining good decoding means for developing efficient decoding algorithms; resulting algorithms are referred to as spectral or transform d omain decoding algorithms and constitute an important subset of algebraic decoding algorithms .As examples may be maintained DFT domain decoding algorithms for BHC codes [1] and WHT domain decoding algorithm for first order RM codes [2] . To detail the BHC decoding algorithm continues to be the most important algorithm among spectral decoding algorithms based on DFT domain techniques [1] .It is found hat the BHC bound .The concept of BHC decoding is applied for decoding of p other related class of codes such as alternate codes and Goppa codes . Another application of BHC decoding idea is in developing decoding algorithms for the class of 2 - D BHC codes obtained by generalizing the concept of BHC codes to two dimensions .Recently some attempts have been by generalizing the concept of BCH codes to two dimensions . Recently some attempts have been made to decode some non -BCH codes using spectral methods [3] .A careful examination of different DFT b ased spectral decoding algorithms reveals certain basic principles which can bused to derive all these decoding algorithms . The range of application of WHt techniques is limited to Reed - Muller codes. Presently decoding algorithms based on WH T methods are available only for first order RM codes .The attractiveness of WHT is that computation involves integer arithmetic and that decoding operations are relatively simpler. Taking the clue from the DFT based approach it is worth while to look for some common principle from which one can derive WHT based decoding algorithms for higher order RM codes also . It turns out that from the system theoretic

viewpoint decoding of RM and cyclic codes can be viewed as a ppropriate deconcvolution problems in with and transform domains respectively Therefore the main is to look for simple ways pof performing deconvoluation using the structural properties of the codes and the relevant transforms . The results obtained and the underlying approaches are summarized in the following pages. 1 WHT based Decoding Algorithms A mapping from binary {0.1} to real {, - 1} is necessary in order to apply WHT techniques. Then the EX - OR addition in bi nary field is mapped into point wise multiplication of error and the transmitted code world .In the transform domain the received spectrum .The error spectrum can accordingly be visualized as the impulse response of a linear which convolution ally distorts the transmitted codeword spectrum .The decoding problem can be formulated as deconvolution of the WHT of received signal to recover the codeword spectrum .It is found that WHT SPECTRUM OF RM codes can be ground into several hist ogen classes where a histogen class consists of all transformed RM codes can be grouped into several histogen classes where a histogen class consists of all transformed RM code vector having the same distribution of spectral values and t heir frequency (For histogram only absolute value are considered) It is shown that the maximum possible deviation of the WHT spectrum of the received signal from the WHT spectrum of the codeword is linearly to number of error occurred Th is information is useful to derive simple procedures for determining the histogram class of the codeword spectrum and consequently to recover the transmitted codeword from of the receives word. Let R(r,2 m) denote r th order RM code of leng th 2 m . The WHT spectrum of R(1,2 m) is known to have only one nonzero component with a value equal to ± 2 m .Therefore R(1,2 m) has single histogram class consisting of a single component with the absolute value 2 m and he remaining 2 m - 1 zero valued components .Our contribution is in extending WHT spectrum based histogram classes j th class containing 2 2j nonzero spectral locations having value jm −± 2 for j =0 to [m/2]. Proof of various re sults are based enantiomorphism group property of RM codes and a theorem concerning canonical representation of multivariable polynomials over a binary field .Unfortunately histogram characterization could not be extended to third and higher order RM codes primarily because of non - availability of appropriate canonical forms . Our next contrtribution is in using the histogram characterization of MR codes for their decoding .To proceed with the decoding we need to know how the histog ram of the MHT spectrum of the code is modified due to channel errors. This computation is in general quite tedious except in case of small errors . Fortunately it is found that the knowledge of range of variation of spectral values is u seful for decoding purpose. The maximum possible deviation of a spectral value is shown to be linearly related to number of error occurred . The possibility of recovering the histogen of the transmitted codeword spectrum from of the recei ved world spectrum is based on the following conditions: first the sign of a nonzero spectrum should not change due to error modification and second different spectral values within a spectral class are far enough so that their range of v ariation are distinct .An analysis of these conditions shown that it is still not possible to decode R(2,2 m) in a straightforward manner the exception being R(2,2 4) which id the first nontrivial single - error correcting R(2,2 4) code and utili ze it to develop a decoding algorithm for general R(2,2 m) . In order to obtain WHT domain decoding algorithm for general R(2,2 m) we utilize the superimposition property [2] of a RM code to decompose a larger RM code into smaller RM codes having efficient decoding algorithms which can be combinmned to provide a decoding algorithm correcting upto full error correcting capability for the larger code .In [4] Tokiwa et.al have an algorithm which we refer to as TSKN algorithm) based on the superimposition property which reduces the larger RM decoding into several smaller single - error - correcting and double

– error - ducting RM codes having an efficient decoding algorithm. They have also shown that TSKN algorithm has bett er delay compared to conventional majority logic decoding algorithms for RM codes . The decoding algorithm given in the based on a decomposition that reduces decoding of a R (2,2 m) to (m - 4) fold decoding of R (1,2 m - 1) and a single de coding of R (2,2 4) respected 2 m - 4 timesx (A for brevity the repeated code is represented as 2 m -4 *R(2,2 4) . A new WHT based decoding algorithm for decoding of 2 m - 4 *R(2,2 4) is developed on the lines of WHT based decoding algorithm for R(2,2 4) .This new WHT based algorithm and the well - known WHT based decoding algorithm for R(1,2 m) are combined to provide decoding algorithm for R(2,2 m) .We propose two variations of the decoding algorithm for R(2,2 m) one of which avoids the nee d for inverse WHT and also simplifies the process of recovering information bits . We show in a similar way that decoding R(3,2 m) reduce to decoding of (m - 5) FOLD r(2,2 m - 1) and a single decoding of 2 m - 5 *R(3,2 5) .WHT domain decoding algori thms for R(3,2 5) and 2m - 5 * R (3,2 5) are given and utilized for developing a decoding algorithm for R(3,2 m) by combining decoding algorithms for 2 m - 5 *R(3,2 5) and R (2,2 m). There are important differences in the working of the TSKN decod ing algorithms and the WHT based algorithm apart from different ways of decomposing a RM code .Both the method s at some stage have to do decoding of repeated codes .TSKN method attempts to decode each code block of the reported code an d then ascertain type validity of decoding . If the result is not valid it takes the next code block and repeats the process .The number of attempts varies one to a s the number of repeated code block. WHT based method attempts varies one t o as many as the number of repeated code blocks. WHT based method attempt to decodes the entire repeated code in a single step .In case there are ambiguities regarding the spectral pattern of the code it requires one more step to complete the decoding .Therefore the number of attempt s is at most 2 .This feature may simplify the decoder architecture compared to TSKN method .We give a decoder architecture based on the WHT domain decoding algorithms for second and third order RM codes .An estimate of the decoding delay with this architecture is given .Te delay is shown to be significantly less compared to TSKN algorithm which in turn has smaller delay compared to other decoding algorithms for higher order RM codes. 2D FT based decoding Algorithms Both 1 - D and 2 - D DFT based algorithms depends upon characterization of cyclic codes as having a specified set of codeword spectral components identically equal to zero .As a consequence the DFT spectrum of the recive3d vector at these locations is equal to the value of the error spectrum .This direct availability of partial information on error spectrum is in contrast to its non - availability in case of WHT By introducing an auxiliary vector is z ero at locations corresponding to error locations considering the point wise product of the error vector and the auxiliary vector and transforming the point wise product a “zero” output convolution the error spectrum is formed .Therefore DFT domain decoding can be formulated as zero - output deconvolution with partial knowledge of the error spectrum .The other convolving polynomial DFT of the auxiliary vector introduced earlier is also of importance in that its zero are related to the locations of the error and hence known a s the error - locator polynomial .In the DFT domain the deconvolution takes the form of problem of finding the connection polynomial of least order LFSR generating the entire error spectrum wit h he known error spectrum at the zero location of the code as the initial values of the LFSR .In view of this the error – locator polynomial is also referred to as the connection polynomial. We treat 1 - D and 2 - d decoding algorithms in a sing le framework .Analysis of a typical DFT domain decoding stage and the error evaluation stage the error spectrum computation stage the error location stage and the error location stage and the error evaluation stage .Any DFT based decoding approach centers around the error location stage which involves formulation and computation of error locator polynomials .Any particular decoding approach is typed by the nature of the disconsolation corresponding to the nature of the c onvolution of the transform domain vector of the corresponding error locator and error spectrum polynomials .For efficient computation the error location stage requires some structure on the zero location of the code ; this also influence the er ror evaluating stage .Error evaluation can be completed in various Ways and an appropriate choice is made so as to reduce the overall decoding complexity . We study two distinct DFT decoding the connect of an error locator is to associat e a position 'i' with ' a ' were a is the primitive n - the root of unity (n is the length of the code) .The basic of generalization from the standard 1 - D BCH case is the associatio n of a nonzero error position (I,j) with a 2 - d error – locator (ji ba) corresponding to row and column indices of the error position .Then the 2 - D convolution of the transform domain vectors of the corr esponding error - locator and error spectrum polynomials is zero and therefore the decoding problem is formulated as a 2 - D DFT zero - output DECONVOLUATION ,../The error - locator polynomials or recurrence polynomials can be interpreted as two variab le polynomials .This interpretation immediately suggests that all polynomials that satisfy a given array of 2 -D error spectrum constitute an ideal .Then the problem of LFSR synthesis is how to be equivalent to finding a "distinguished basis " of the ideal which is minimal in the senses of some total ordering for a 2 - D array .Sakata has given an algorithm for efficiently computing such a basis which is equivalent to finding an algorithm for 2 - D LFSR synthesis [5] .The error - e valuated stage consists of either solving for the roots of the system of connection polynomials for individuals error positions in the binary cases or performing 2 - d recursive extension of the known error spectrum to obtain the complete error s pectrum . We propose an alternative approach to obtain a DFT Based spectral decoding algorithms centered around a generalization of Blahut's 1 - D complexity theorem for a linear recurring sequence [6]. The complexity of a sequence is defined as the minimal order of LFSR generating the sequence. This notion of complexity has proved to be very useful in deriving the minimum distance bounds and in determining the weight structure of codes .We show that it can also play a crucial role in the formulation and design of decoding algorithms. This fact is more evident in the 2 - D decoding algorithms . Blahut's 1 - D complexity y theorem states that the minimal order of LFSR generating a time (or frequency) domain sequence is equal to the Hamming weight of transform sequence in frequency (time) domain vector .We showed that the notion of linear complexity is easily generated to a 2 - D sequence as follows .The minimal order of common LFSR generating all row (column) sub sequences is equal to number of nonzero columns (rows) in the transform sequence .The roots of the common row (column) connection polynomial gives the columns (row) corrupted by the error . The basic idea of our approach is to associat e anonzerro error position (I,j) with a 1 - D error - locator corresponding to either row (ia) or column (jb) index of the error position Then the 1 - D convolution of row column error locator vector with each of row (column) subsequences of the 2 - D error spectrum is zero A typical 2 - D decoding problem is then viewed as viewed as equivalent to solving several 1 - D disconsolation problems which for the DFT case are equivalent to synthesizing several 1 - D LFSRs corresponding to the row or column subsequences .Note that in contrast to 2 - D dconvolution approach LFSR synthesis is treated as 1 - D whereas the error spectrum is 2 - D in nature .In The error evaluation stage the recursive extension ha s to be done in both the column and row and column connection polynomials contains all the error positions. This makes 1 - D deconvolution approach is based on 1 - D LFSR synthesis this may imply milder constraints on the structure of zero loca tions for a given error -

For more details click here Back

Title **:** ***Novel Static Distance Realying Schemes For Transmission Line Protection***
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Supervisor(s) **:** ***Dubey GK & Singh LP***

Abstract

Fast and reliable protection is required for the safety of the equipments and security of the power system. Utmost care has been taken for the protection of transmission lines because these are subjected to the highest number of faults due to vast length its consequent exposure to atmospheric hazards and also outage of one line may cause out of step condition. A number of protection schemes using electromechanically or static relays have been developed. With the development of EHV/UHV systems for transfer of bulk power for longer distances the protection schemes have been subjected to challenging conditions. Electromechanical relays had been found unstable for EHV lines due to slower operation high burden and also these relays do not have independent contr ol over the resistive reach hence these are prone to palpation due to power swing. Static relay have been developed which can produce variety of threshold characteristics among these quadrilateral characteristic has been found most suitable for EHV lines . Emergence of UHV lines necessitated the development of ultra high - speed relays for improving stability. Recently traveling wave relays have been developed which can detect fault within 5 m sec. Among the available protection scheme these are most suit able for UHV lines but presently use of such relays is not very common. Recent survey of IEEE sub - committee showed that the carrier pilot scheme is most common for the protection of EHV/UHV LINES. Step distance relaying schemes without communicatio n channel are also used but these are common to provide back up protection only. However static measuring circuits are preferred most. Measurement of the impedance of line by static circuit is done indirectly by determination of phase angle us ing block average phase comparator. The circuit developed so far this purpose is not block average phase comparators in real sense. These have high operating time as the phase angle reaches the limiting value. Step distance protection is don e by relays using two or three zone detectors. From the point of view of reliability and economy three is a need of a single comparator three - step distance relay. For medium voltage lines mho or offset mho characteristics have been found suita ble. Polyphase relays have been preferred for the sake of economy. Polyphase relays developed so far the protection against phase to phase faults have source impedance dependent reach .The polyphase relays for ground fault protection either have narrow trip area or large trip area which make these relays unsuitable. Hence the is a need of development of relays with better performance. The objective of the present thesis has been (1) To present critical review of existing sche mes in terms of simplicity transient performance

number of comparators used in a relay and circuit capability in generating various types of pick up characteristics. (2) To develop a real block average phase comparison scheme. (3) To develop a fast and reliable three - step distance relay based on a single comparator three - zone detector. (4) To develop a novel polyphase relay with circular pick up characteristics for the detection of phase - to - phase faults and three phase fault and also the relay must ha ve the reach independent of the source impedance. (5) To develop a polypphase3 ground distance relay capable of generating major or offset mho characteristics and (6) To develop a three phase dynamic test bench with simple and economic design. The summary o f the work reported in this thesis is as follows: Chapter 1 gives motivation for choosing the present problem precise problem definition and brief historical account of the development of protection schemes for transmission lines .It also present curre nt trends and practices in protection of transmission lines. Chapter 2 presents critical review of the important relaying schemes developed so far medium voltage and EHV/UHV lines. It has been concluded from the review that so far no real block a verage phase comparator scheme has been developed .For step distance protection of transmission line on relay with a single comparator three zone detector has been developed. Chapter 3 presents a real block average phase comparator scheme. Also in this chapter a single unit three zone detector has been reported .In addition a three step distance relay based on only two phase comparators has been proposed designed and fabricated. Chapter 4 present a novel polyphase distance relay for the protection of medium voltage lined against all phase - to - phase faults. The relay provides identical offset mho characteristics for phase - to -phase faults and mho Characteristic for three - phase fault. The performance of the relay has been found to be better than existing scheme of the relay has been found to be better than existing schemes. The source impedance less affects the reach of the relay. Also in this chapter a polyphase ground distance relay has been presented .The relay procedures offset mh o or composite mho – offset mho characteristics for all ground faults .The reach of the relay is almost unaffected by different types of faults source impudence and prefault loading. Chapter 5 presents a three - phase dynamic test bench .T he bench has simple economic design .The control and measurement circuit has been developed using linear and digital Ics. The switching angle and relay operating time are measured digitally using seven segment displays and finally Chapter 6 concludes with a brief account of the work reported in this thesis and suggestions for future scope of work.

For more details click here Back

Title	***:***	***Polyphase And Frequency Hopping Sequences Obtained From Finite Rings***
Author(s)	***:***	***Udaya P***
Roll No	***:***	***8620462***
Supervisor(s)	***:***	***Siddiqui M U***

Abstract

POLYPHASE AND FREQUENCY HOPPING SEQUENCES OBTAINED FROM RINGAS This thesis is concerned with algebraic constriction of sequence obtained from residue clam finite rings and with their periodic correlation and linear complexity (LC) properties. Such sequences are of interest in polyphase and frequency looping code division multiple (CDMA) Communication systems. The finite rings considered in the study are Residue clam integer ring modulo M: ZM and residue class polynomial ring GF (p) [ )()]([:)(mod], xxxx wherewwPw pn is an nth degree polynomial over GF (p). The sequence in this thesis are viewed as polynomial mappings of trace functions of units elements of Galois extension rings he two Galois extension rings considered and Galois extension ring of degree of extension r and Galois extension ring of [kpn wp] the LC of sequences was complexity theorem for finite field case. For studying a correlation properties of sequences it b advantageous to a correlation transform of sequences over finite rings various families of sequences derived in the thesis are in to the following classes (1) Families derived form local rings (2) Families derived from semi-local rings.

For more details click here Back

Title	***:***	***A Discrete -Time HVDC System Model For Stability Analysis And Self-Tuning Control Design***
Author(s)	***:***	***Pandey Rajendre Kumar***
Roll No	***:***	***8710472***
Supervisor(s)	***:***	***Ghosh Arindam & Sachchidanand***

Abstract

High voltage direct current (HVDC) transmission made a modest beginning with the commissioning of a submarine link between the mainland of Sweden and island of Got land in 1954. ever sine then the HVDC transmission has grown at a tremendous pace and has no w emerged as a viable alternative to ac for bulk power transmission. In addition, the HVDC transmission has been also considered for asynchronous interconnection of ac systems and stability enhancement of the host ac systems. All these applications primarily exploit the controllability features inherently available with the dc transmission. Consequently, the HVDC converter controls assume paramount importance for successful and reliable operation of HVDC systems. HVDC simulators have been widely used in practice for the stability investigations and design of controllers. To supplement the use of HVDC simulator, analytical methods have also been reported in the literature. These analytical methods offer more flexibility in analyzing the various control alternatives. The outcome of these analytical studies can serve as a first step screening tool towards the choice of controller parameters. The final tuning of the control system can be done on an HVDC simulator which employs realistic controller models. This thesis also addresses itself to the development of analytical technique for the stability investigations and the design of converter controls. An HVDC systems is a large scale multivariable system which is both non linear and prone to external disturbance s. This makes the stability investigations extremely complex. In order to overcome these complexities to a certain extent, it is a standard practice to carry out stability investigations using linear zed perturbation models in which the system is linearized around a nominal operating point. To achieve this objective it is important that the behaviour of the various components of the HVDC system be appropriately represented in a linear domain. In this context, both continuous and discrete time system representations have been described in the literature. The basic difference in the two alternatives is in the approach employed for converter representation. In the development of continuous time system model, the converter is represented as a voltage source described by the average dc voltage equations of the converter. In discrete time representation the converter is modeled taking into consideration the discrete nature of its control which is effective only at discrete time instants corresponding to the value firing instants as determined by the firing pulse generation mechanisms. It may, however be acknowledged that the converter operations is discrete not only at the value firing instants but also at the instants corresponding to the completion of commutations process. Both these discrete

time instants need to be adequately taken into consideration while representing the converter for small signal stability investigations. In addition to this, the system model should appropriately take into account the dynamic interactions between the dc and the host ac systems. This is particularly important as the characteristics of the ac system significantly influence the controller design. The controller parameters obtained through small signal stability studies will provide sufficient damping and reject disturbance in the neighbourhood of the nominal operating conditions considered in the linearization process. To design a controller which can provide satisfactory performance over wide range of operating conditions, various nominal operating points must be chosen and the best controller parameters for them should be obtained. The final choice can, at best, be a compromise of the various controllers derived. Even then there is no guarantee that such s controller would meet the required performance criterion over the entire spectrum of operating conditions. To alleviate this problem and design a suitable controller which can meet the specified requirements an adequate control strategy or a robust control design strategy can be employed. In this thesis an attempt has been made to explore the possibility of employing a suitable multi-rate sampling based converter current controller. Based on the above considerations the objectives of this thesis are: (1) development of an integrated AC/DC/AC system model taking into account the discrete nature of converter operation corresponding to the time instants at which the value in different converter begin and cease conduction. (2) Design of self tuning controllers and the associated observer for a two terminal HVDC system utilizing the discrete time converter representation. The major contributions of the thesis are: (1) development of a novel HVDC converter model which is based on the theory of multi-rate sampling taking into account the discrete nature of the converter operation corresponding to the time instants at which the values begin and cease conditions. (2) Development of a two terminal integrated AC/DC/AC system model based on the theory of multi-rate sampling so as to appropriately inter face the two converters which results in four discrete time instants (two time instants corresponding to each converter) in the interval between two successive value firing instants in a converter. (3) Design of self tuning controllers for two terminal HVDC system utilizing the multi-rate sampling based converter model. In this context, the dynamic noise rejecting observer has also been designed. The various self tuning controllers considered are: Minimum Variance (MV), Polynomial Linear Quadratic Gaussian (PLQ G) and Pole Assignment (PA) controllers. A brief description of the work reported in the thesis is given below. The first Chapter introduces the various aspects studied in the thesis and reviews the literature in the area. Chapter two reports on the development of a novel discrete time converter model which is based on the average dc quantities. Under normal conditions, a six pulse converter operates in alternate two and three value conductions modes which areas due the firing of a value followed by cessation of another value at the end of commutation process. The transmission from two value to three value conduction mode and back to two value conduction model can be viewed as discrete events. Within two successive value firing instants (where the transit ion from two values to three value conduction takes place) which occur at every 60 o under steady state operation of a six pulse converter, the transition from three value to two value conduction model introduces an additional discrete time instant which depends on the commutation period. This situation is conveniently handled employing the theory of multi-rate sampling. The converter model is illustrated through the case study of a single converter system feeding an RL load which is represented in the state s pace framework along with the converter controller. The firing pulse generation scheme which is treated as a discrete time sub system is based on individual phase control. Employing the Eigen value analysis technique the stability boundary of the system is obtained in the controller parameter plane (gain versus time constant). The stability boundary is validated through nonlinear digital simulation. The stability boundary and root loci obtained for a samples system using the proposed converter model is compared with those obtained by modeling the same system based on modified – z transform technique. The results for both the cases are in good agreement. Chapter three describes the development of a two terminal HVDC system model in a state space framework. The beginning of value conduction and the end of commutation process at the rectifier and inverter terminals may not be coincident. Assuming a time delay between the conduction of a rectifier value and the conduction of an inverter value, four discrete time instants are obtained in an interval of 60o. this situation is conveniently handled employing the theory of multi-rate sampling which results in the proper interface of the rectifier and inverter terminals and leads to the development of a state space discrete time HVDC system model. The detailed representation of ac system is ignored here under the assumption that the host ac systems at the two ends are strong. However, the converter ac bus voltage phase angle separation between the two ends has been taken into account while deriving the system model. The HVDC system model is utilized for the stability analysis of sample systems. The stability domains are obtained in the controller parameter plane using eigen value analysis. The stability regions are further validated through detailed digital simulation. Employing the converter model derived in Chapter two, the development of a discrete time state space model of an integrated AC/DC/AC system is described in Chapter four. The development of the model proceeds systematically with the development of various components models which are interfaced through appropriate variables to arrive at the overall systems model. As the ac system is assumed to be weak, its detailed representation has been considered here to take into consideration the interaction between the ac and dc system dynamics. The development of system model is quite general and does not assume that the ac voltages at the rectifier and inverter ends are coincident.. both equidistant pulse control and individual phase control firing schemes have been considered. The system models illustrated through the case study of a sample system. Chapter five introduces multi-rate based self tuning control strategies for an HVDC converter. Such controllers can reject various disturbance arising due to the changes in system parameters and delays (operating conditions) which is not possible by a controller of the conventional type. The self tuning controllers are developed assuming that the firing pulses are generated u sing a digital firing scheme. In the digital implementation of the individual phase control firing scheme, the firing angle of the converter is loaded into a counter at the zero crossing of the commutation voltages. This requires that the current controller must provide the information about the firing angle well in advance. in the present work the direct current is obtained through a running average under these conditions, a control decisions has to be taken two steps in advance this philosophy is incorporated for the design of self tuning controllers Three different self tuning controllers (i.e. Minimum Variance (MV), Polynomial Linear Quadratic Gaussian (PLQG) and Pole Assignment (PA) controllers) are presented in this chapter. Out of these controllers, the first two use a two step ahead prediction model while the third one places the closed loop poles in a desired way. The controllers designed are tested through the digital computer simulations of a sample signal converter system feeding an RL - load under various conditions (small and large disturbances). Chapter six extends the philosophy of Chapter five for the design of self tuning controller for a two terminal HVDC system. The self tuning current controller, situated at the rectifier end, requires information about the current at the inverter terminal. If this information is available the self tuning controllers presented in Chapter five can easily be modified incorporating the current at inverter terminal as a disturbance. As the inverter is situated far away, it would seem that a measurement through the communication channel is the only option. However, as dc link transients are quite fast, a feed back over a communications channel may introduce time delay in the controller action which may be detrimental to the system stability. To alleviate this problem, a robust observer is also designed which requires the measurements of the rectifier terminal quantities only to estimate the inverter end current. This observer requires no knowledge of the inverter end voltage and can reject any disturbance acting on it. The proposed self tuning controller observer is verified through digital simulation. Chapter seven outlines the conclusions drawn from this work. A few suggestions about the future development are also presented here.

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