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Part of the Nuclear Engineering Commons, Operations Research, Systems Engineering and Industrial Engineering Commons, Other Engineering Commons, Power and Energy Commons, and the Risk Analysis Commons
Power System Operation & Energy Management deals operations and control continues to be utility mission-critical activities, focused on the reliability and security of the grid as well as on economic dispatch of the system. In spite of so many advancements in the power and energy sector over the last two decades, its survival to cater quality power with due consideration for planning, coordination, marketing, safety, stability, optimality and reliability is still believed to remain critical. Though it appears simple from the outside, yet the internal structure of large scale power systems is so complex that event management and decision making requires a formidable preliminary preparation, which gets still worsened in the presence of uncertainties and contingencies. These aspects have attracted several researchers to carryout continued research in this field and their valued contributions have been significantly helping the newcomers in understanding the evolutionary growth in this sector, starting from phenomena, tools, methodologies to strategies so as to ensure smooth, stable, safe, reliable and economic operation.

The Power System Operation & Energy Management had a great effect on Communication. Its utilities are operating under an unprecedented demand for accurate, real time data—enabling utilities to meet regulatory demands for open and timely external reporting to understand the situational awareness of the grid, and to answer critical operation and control issues.

The Conference sometimes is conducted in collaboration with other institutions. IRNet encourage and invite proposals from institutes within India to join hands to promote research in various areas of discipline. These conferences have not only promoted the international exchange and cooperation, but have also won favorable comments from national and international participants, thus enabled IRNet to reach out to a global network within three years time. The conference is first of its kind and gets granted with lot of blessings.

The conference designed to stimulate the young minds including Research Scholars, Academicians, and Practitioners to contribute their ideas, thoughts and nobility in these disciplines of Engineering. IRNet received a great response from all parts of country and abroad for the presentation and publication in the proceeding of the conference.

I sincerely thank all the authors for their invaluable contribution to this conference. I am indebted towards the reviewers and Board of Editors for their generous gifts of time, energy and effort. It’s my pleasure to welcome all the participants, delegates and organizer to this international conference on behalf of IRNet family members.

I wish all success to the paper presenters. The papers qualifying the review process will be published in the forthcoming IOAJ journal.

Convenor :-

Mr. Bikash Chandra Rout
A NOVEL APPROACH FOR ENHANCING ACCURACY IN WEIGHING SYSTEM USING A NEW SUPERREGENERATIVE ARCHITECTURE IN DSSS

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Abstract—This paper, deals with a new design high accuracy mass measuring system using a new Superregenerative architecture of directs sequence spared spectrum technique. use of conventional DSSS Superregenerative architecture to get accuracy in very efficient mass measuring is now wellknown ,but as because conventional Superregenerative architecture exhibit a major inconvenience :an excessive bandwith that makes it more vulnerable to noise and interference than other system.in this paper we present a new architecture that make use of Superregenerative principal to achieve non coherent detection of direct sequence spread spectrum. Here The signal coming out of load cell of mass measuring system is put down below the white noise level by using this new Super regenerative DSSS architecture, next amplified finally converted to digital form, subsequently, the digital signal is as input to the micro controller. Thus the data is transmitted accurately.

Keywords- Accuracy, Mass measuring, Noise, Superregenerative Reciever.

I. INTRODUCTION

To accurately indicate the mass of a process without extensive operator involvement, a mass indicator system relies upon a transducer, which accepts a mass, such as a capacitive or resistive transducer as input. It indicates the actual mass, or set point, and provides an output to a display element. And the whole system should be analyzed in selecting the proper transducer and the data acquisition system. Mass of a material is displayed digitally using a micro-controllers based mass measurement system. In the practice, the micro-controllers using general SS algorithms cannot meet higher accuracy. It is well known that SS systems offer certain desirable characteristics that are difficult to obtain via conventional technique. For instant, SS signal exhibit low spectral power density. And SS receiver achieve improve interference resistance

II. REVIEWS AND OVERVIEW

The main constraint in achieving higher accuracy is a superimposed noise on a useful signal from the mass measuring system. The common sources are: (1) constant values (2) the noises: electromagnetic pick-up, power harmonic, thermally unstable circuits and the gain programmable by software (through PW's output of the micro controller) (3) Mass sensor (i.e., Capacitive OR Resistive mass sensor). The common sources are amplified before transferred into the A/D converter, and therefore the noise is amplified and an error is introduced in the system. A signal processing module (SPM) acquires the electrical signal from the mass sensing device and estimates a value of mass.

The two main aims for improvement are increasing the Speed of

Sensing the mass variation and achieving good measurement accuracy. Improvement in SPM that provides any one or both aims brings significant benefit to the overall mass monitoring system. Owing to the physical characteristics, when the mass sensor detects the external mass of a material, it produces a weak current. The weak current is passed through AD7730 that has the PGA (Programmable Gain Amplifier), A/D converter and the two stage digital filter into digital signal. The micro-controller (8031) transfers the detected digital signal to the communication unit and the display unit according to the procedure set on the memory.

III. PRINCIPLE OF OPERATION

Fig 2(b) illustrates the new mode of operation presented in this paper for detecting direct sequence spread spectrum (DSSS) signals. The transmitted data are spread with a PN sequence, and the SRO is quenched synchronously with the received signal, so that a single sample of each chip pulse is taken. This
means quench frequency equals to the chip frequency. Since the duration of the sampled chip is closer to that of the sensitivity period, the bandwidth of the SS signal and that of the receiver become comparable. Moreover, the synchronous operation, which is in any case required to despread the DSSS signal, allows the use of special chip envelopes that concentrate the signal energy in the sensitivity periods. In particular, the receiver operates as a matched filter when the signal envelope matches the sensitivity curve.

IV. DESIGN METHOD

In Fig. 3, the amplitude spectra reveal the significant noise components of the Mass measurement system. The first peak is centered at approximately 2.2Hz, and the second and third peaks appear to be integer multiples of this noise. Hence we can assume that there are no significant harmonic components beyond 40Hz. Therefore any sampling rate selected must be greater than 80Hz. In order to reject the high-frequency random oscillatory component in the circuit, there is always a RC low pass filter in the output of the circuit. This kind of filters can restrain high frequency noise, but its effect on low frequency noise is few. Digital filters are ideal for the treatment of low frequency noise. They can be implemented in real time or as post-processed applications.

This paper, we present the comparison between the proposed digital filters in PC and the other filters which can achieve the main features the mass measuring system needs. Then, we present (1) Moving average filter algorithm [2], (2) LMS adaptive filter algorithm [3], (3) conventional FIR filter algorithm discusses the development of a microcontroller based measurement system. [4] Capacitive sensors are increasingly becoming common because they can be built with affordable technologies, and they have high impedance, which implies low power consumption [4, 5]. In data acquisition module Conventional SS receivers require proper pseudo-noise (PN) code synchronization in order to despread the received signal and detect data. In contrast to conventional SS receiver, in which synchronization circuitry entails a substantial in crease in receiver cost, size and power consumption (6), (7), the architectures presented here minimize overall complexity by employing a single signal processing path for data retrieval and synchronization, along with a reduced number of RF stages. As a result, very simple data acquisition ckt can be designed, which combines the advantages of superregeneration, low cost and low power consumption, with those SS techniques. The block diagram of the proposed DSSS superregenerative architectures shown in fig 4. They incorporate a local PN code generator that uses the quench oscillator as a clock, making the chip period equal to the quench period. In different ways, both architecture multiply the received signal by local PN code and perform integrate-and-dump filtering. The integrator output is sampled and held during each bit period, and used as decision variable. Finally, a synchronization loop that controls the frequency the quench VCO keeps a correct PN code alignment with Superregenerative oscillator (SRO). Fig. 5 show that the receiver bandwidth is much closer to that of the input signal than in narrow-band superregenerative receiver: the ratio between receiver and signal bandwidth ranges from 1 to 5, whereas for narrow-band receiver this ratio is at least (typically greater than) 10.

![Fig. 3](image_url)
extremely low frequency noise of the mass measurement System. Hence DSSS in the measuring system is proposed. The main difference is that we use a very important PN code to shift the noise to high frequency and we use different procedures to apply new DSSS, as shown in the of Fig. 4.

After shifting the noise to high frequency, we can use the filter with 9 taps to filter the noise.

With this novel method, the proposed new DSSS filter algorithm used in mass measuring system can meet the requirements of speed, stability, and precision.

The maximum Mass measured with this device is 70 kg. in step of 7.14% of full scale. The result are appreciably linear with R^2 (correlation coefficient) equals to0.999 as illustrated in figure.

V. CONCLUSION

In the Mass measurement systems conventional filtering method employed have limitation in improving the accuracy and in throughput rate. In this case, an alternative technique has been explored to find a solution. It will enable measurement accuracy to be 1/172000. The result shows that new technique of DSSS filter can be employed in a practical system. The basic materials for making mass sensor are simple glass tube and enameled copper wire and the filtering circuit involved semiconductor devices. From the experimental study, the repeatability, linearity, and resolution are satisfactory within the tolerable limit of industrial mass measurement.

REFERENCES


Abstract- Today the voltage stability has become a major concern for power system operation and controls from both consumers and Utilities. Even after the recent inventions in controls systems and power electronics, every Utility nowadays faces this problem and deals in own different way one of them is reactive power management. This method has found to be very effective in aspects of voltage regulation and stability too. In this paper I would try to take the attention of this reactive power management method, its presence in power systems, issues related with it and also with effects on various power system components and at last the major planning steps in brief to this problem to ensure reliability issues for the power Utilities and the consumers.

Keywords- voltage stability; power system operation and control; power reliability; power utilities

I. INTRODUCTION

In a modern electrical system, the power has been always stated either active or reactive power intentionally everywhere as both of them highlights their significance where they are supposed to be. Like running a motors, heating the conductor are the primary places where the active power is employed whereas the reactive power comes into picture where production of electric and magnetic fields in inductor and capacitors are related. One of the popular analogies used is the Beer mug analogy which quite simply explains the relation in the active, reactive and apparent power. Hence reactive power management can be defined as the control of voltage and power in order to satisfy the following objectives

1) Voltages at terminals of all connected load to existing power system should be within the accepting limits.

2) To enhance system stability for maximization of utilization of transmission systems; and voltage and reactive power control has very crucial role to achieve the goal of high system stability. For which every utility is bound to their customers.

3) The operation of transmission system with good efficiency is hugely depends upon the resistive and reactive losses taking place at system operational period, which can be practically made minimum using appropriate reactive power flow in the existing grid. Therefore we can stay that for every utility system to achieve high degree of constant system voltage profile even

i) ii) iii) in adverse system conditions or instabilities, reliability to customers and fulfilling the demands with minimization of losses during transmission and distribution especially. The reactive power management will carries a lot of weightages.

II. REACTIVE POWER ORIGIN

Reactive power production takes place when current and voltages are not in phase and it is measured in VAR (voltage ampere reactive) it is produced when current leads voltage and consumed when vice versa. Transmission line generates VAR under no or Low loaded condition and consumes under loaded one. Hence at any given instances power system can experience different voltage levels at various locations. It can also be stated that when generator introduces a reactive power it is said to be overexcited and under-excited when it is drawing the same from connected system or grid. So the reactive power output is a dependent function of generator field current provided by excitation system.

III. REACTIVE POWER IN THE PRESENT POWER SYSTEMS.

A) Need of reactive power

i) For any load connected to the grid for successful and satisfactory operation of it without causing problems like overheating etc., for reducing the transmission losses and maintaining system ability for withstanding and preventing the voltage collapse. That it can be simply put as fall and rise of system voltage with respective decrease and increase in reactive power. Leads to voltage collapse when system is trying for more load to serve than the rated load supported by system voltage.

ii) At such a low voltage cases, as voltage falls in order to continue to serve the constant power, current increases enormously leading to system consume the more reactive power this leads to further voltage drop. Now when the current exceeds the limit for given transmission line then
that particular transmission line goes offline which overloads the other operating lines causing the outages or cascading failures.

iii) If further voltage level dropping occurs then causes further reduction in reactive power from capacitor and line charging, again further voltage level falls which causes tripping of additional elements and falling voltage and load losses which is not desirable for any utility at any condition in order to achieve the high reliability for consumers point of view.

iv) This whole process continuously progresses into the uncontrollable declines of voltages in the system

v) Leading to inability to supply the reactive power required by system

vi) vii)

B) Presence of reactive power in an operational grid

I) As stated above the voltage profile and the reactive power control are the two different faces of single coin for supporting reliable operation and the multiple power transaction scheme held between two or more utilities on the commercial state

II) Following are the reasons which clearly indicates that why system voltage profile is so much dependent on reactive power

III) In the whole power network there are large no of equipment and the load connected at consumer ends are designed in such a way that they can satisfactorily work in the ranges of voltages which is ±5% of nominal voltage level.

IV) When the voltage falls many electrical load like light bulbs output decreases, overheating of induction motors and failure of electronic circuits; whereas at high voltage the chances of equipment damages are pretty high leading to reduction in life.

V) Another reason is that as reactive power consumes transmission and generation resources, for maximizing that real power we must lower down the reactive power associated with it, but this process will cause the lowering of the generation units real power production capacity.

VI) Also by feeding the reactive power to the existing line, it causes losses in that particular line and in order to balance those losses, we have to supply externally the additional power and energy.

VII) From above all it is clear that the transmission line itself acts as a nonlinear consumer of reactive power which depends upon loading of system as on light loads the reactive power is generated and on connection with heavy loads the reactive power is absorbed and this also accompanied by dependence of system on the transmission and generation configuration.

VIII) While taking into account of reliability consideration of any system the bulk power system is comprising of large no of equipment whose failure can occur with either due to ageing or by malfunctioning and hence system is designed by considering this single contingency situation too with continuous variation of reactive power demands as per the load requirement with time and the generation patterns. Both of these factors termed into the dynamic reactive power requirement hence loss of generator unit with forced outages and the transmission line compounds together for reduction in reactive power supply and the power flow get reconfigured as the system is now demanding for additional reactive power.

IX) There must be a specific part of system or portion from the grid which should respond quickly to the reactive power profile over the system should be responsible for keeping the nominal voltage all over the system. In other words just like a load flow study takes into account of the reserve capacity of plant or the spinning reserve from unit to take the load quickly during scheduled and forced outages, we must provide a same arrangement for reactive load too.

X) It has been found that the reactive loads are more responsible for fall of voltage and the corresponding
reactive losses associated with them than similar sizes of real loads.

IV. PURPOSE OF REACTIVE POWER

i) There are numerous techniques available in markets for controlling and maintaining same or constant system voltage throughout the grid such as making use of synchronous generator, static VAR compensators (SVC) and distributed energy resource equipment (DER)

ii) Synchronous generator are the 3-phase synchronous motor which either runs under excitation state or the overexcited state depending upon either to take up reactive power from system or to supply the same.

iii) Hence based on the requirement, reactive power is either injected to rise up the system voltage level or absorbing it to decrease the voltage level.

iv) Here location and magnitude of generator output can be said as the independent variable for function of the voltage support requirements and that also depends upon the customers loads and the configuration of DER in given power system.

v) The above requirement shows up a lot of variation in them as location, and it get even more volatile when generator outputs and the consumers loads are taken into account.

vi) For managing the reactive power and voltages, the system carries out the operation with three distinct and clear goals as vii) and viii)

1) The high priority is given for maintaining the constant voltages for complete transmission and distribution systems considering both current and contingency conditions.

2) Minimization of congestion of real power flows in the network.

3) And third is the minimization of the real power loss.

V. ISSUES RELATED WITH REACTIVE POWER

1) Reactive cannot be delivered over long distances like real power via transmission lines.

2) Hence wherever it demands for reactive power, the corresponding arrangements should be as near as possible with target location.

3) Many of the time reactive power to be fed is in tied with the real or active power ability of the line.

VI. EFFECT OF REACTIVE POWER

The effect is very wide and deeply related with system
is very important relationship implying similarly on
inductor too. Inductor absorbs more voltage at
higher voltages, and the device is needed most. In the
extreme case when voltage dip is observed as
obviously capacitor contributes less; this leads to
further fall in voltage again capacitors supplies even
lesser resulting voltage collapse and outages.

Inductors are used exclusively for absorption of
reactive power at rated voltage; they can be switched
on and off but offer no control for variation.

Capacitor banks are nothing of capacitors each of
rating around 200KVAR or less and can be connected
in series or parallel and offers limited amount of
variable control. Static VAR compensators SVC
consists of group of capacitors and inductors with fast
switching capacity. This switching time is very small
hence it is also called as sub cycle time (less than
1/60 or 1/50 seconds), hence suitable for continuous
range of control. This range is designed for span from
absorb to generate power. Some voltage drop is
observed as capacitors are used but very fast and
effective reactive power support. These lack short
term overload capacity of generators and synchronous
motors. Typically comes with harmonic filters to
remove harmonics injecting in power systems.

Static synchronous compensators A part of flexible
ac transmission system Static synchronous
compensators or STATCOMs Generates or absorbs
reactive power. STATCOMS are very similar to
SVC but makes use of power electronics than
conventional capacitors and inductors in association
with fast switches. They make use of power
electronics to synthesize reactive power hence the
output obtained is generally symmetric and capacity
is as much as absorption capacity. Very fast and
effective voltage control but not having short term
overload capacity of generators and synchronous
condensers. These are very useful to prevent
voltage collapse, as their response voltage opposes
the voltage square relationship of SVC and
capacitors. Distributed Generators In this case if
each generator has a capacity to supply and control
reactive power output then it would be very much
beneficiary system, otherwise the whole system
performance throughout all transmission and
distribution would get degraded.

Also making use of the energy storage devices and
require solid state inverters to interact with grid. This
gives an advance benefit of full reactive power
control just similar to that of STATCOM. On
transmission side Because of disturbed load
operation, there is phase shift in voltage and current
waveforms leading to synthesis of reactive power.

Some part is adjusted on customer side, and rest all
power loads the line. When reactive component of
current combines with the load current this results in
system voltage drop across network impedances.
Hence when reactive power is adjusted, this in turn
improves voltage drops at lines leading to
improvement at voltage level at customer side. A
transmission line without any load connected to it
acts as source of reactive power, that is leading
reactive power. Whereas a highly loaded line,
length wise inductors comes into picture, which are
charged because of current and introduce a lagging
reactive power. The reactive power occupies the
generator capacity and reduces the real power
production. In view with power factor It is very
simple as origin of reactive power is due to the phase
difference angle of Ø between voltage and current
waveform. Ideally this phase difference is supposed
to be zero and if not then power factor that is cos(Ø)
not equal to unity. Reactive power and the loads We
may be aware of reactive power is needed for
devices to run but reactive power severely makes
effects on appliances, motorized applications and
even on electrical infrastructure, as excess power
dissipation occurs due to flow of reactive current
through wires, switches and transformers. Also a
customer pays for the reactive power losses
developed by reactive current flowing in the homes as
these are losses typically in the form of heat energy
and hence cannot go back to grid.
VII. PLANNING STEPS TO BE TAKEN

Supply of reactive power should be kept as close as possible towards its consumption. Sufficient amount of static and dynamic voltage support along with reactive power reserves must be provided. Proper maintaining and functioning of reactive power devices. Load shedding system must be employed if any utility fails to keep the voltage in the prescribed range through reactive power reserves. Capacitors should be switched ON at low voltage networks for keeping voltage in acceptable limits during peak and contingency conditions. Load shedding should always be the last preference for

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AN EFFECTIVE APPROACH TO REDUCE COMPRESSION ARTIFACT IN IMAGES

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Abstract- Compression artifact is a type of noise which occurs due to the lossy data compression. In lossy data compression we will convert the huge data to a simplified form so that it can be easily stored within the desired space. But, during the time of decompression, the compressor will not be able to reproduce the original compressed data and as a result the quality of original data diminishes, which is known as the artifact. In this paper, we propose an efficient approach for reducing the compression artifact of images without removing the content of the image. We mainly focus on removing blocky noise and mosquito noise. In this paper, TV regularization decomposition, Gaussian filter and DEF technique are used to accomplish this task. This results in the improved quality of the image with better PSNR when compared with the existing methods.

Keywords- Compression artifact; Blocky noise; Mosquito noise; TV Regularization decomposition

I. INTRODUCTION

Nowadays, many high quality displays are widespread which have very high resolution. But the problem with this is that the compression artifacts of images are more visible when the quality of the display device increases. The compression artifacts are much more visible because of the narrow channel bandwidth of the mobile digital TV broadcast [1].

Image data compression is a very relevant issue in many applications. Data compression is done for the ease of storage and for reducing transmission costs while maintaining the quality of the image [2]. As a result, many efficient image coding techniques have been developed for various applications.

There are many techniques for the compression of images. Transform coding is one of the most widely used one [3]. Also, Discrete Cosine Transform (DCT) is one of the most common transforms. The main drawback of the DCT is the blocking effect. Before the coding process, when we divide the image into blocks, it results in discontinuities between the adjacent blocks which are known as the blocking effect.

A standard compression scheme for still images is the Joint Photographic Experts Group (JPEG) [2]. In JPEG compression, each block is independently quantized. As a result, image degradation is high for the reconstructed images from JPEG compression. When a high quantization parameter is used for high compression, the individual processing of each block induces blocking effects.

For reducing the compression artifact in the decoder, a lot of approaches are existing. The signal adaptive filtering method [2] was proposed to reduce the blocking effect of JPEG images in which the blocking effects were classified into grid noise, staircase noise and corner outlier. For reducing the blocky noise in the images, a method based on DEF (Deblocking Edge Filter) [4]-[7] and technique using wavelet transform [8]-[10] was proposed. A technique called Projection Onto Convex Sets (POCS) [3] was proposed in which it impose a number of constraints on the coded image in order to restore it to its original artifact free form. An approach using wavelet based sub band decomposition [9] was proposed for reducing blocking artifact in block coded images. An important method to reduce noise is the Total Variation (TV) regularization [11]-[15]. The TV regularization method [11] removes the noise from images with the help of a constrained optimization type of numerical algorithm.

Another approach for artifact free decomposition of JPEG images is adapted total variation method [15] which reduces the blocking artifacts without removing features and without smoothing images.

In this paper, a new noise removable method which utilizes the TV regularization method in a different way is proposed. First, we have the input image as the image with artifacts. Using the TV regularization decomposition method [16], this image is decomposed into structure component and texture component. The structure component comprises of smooth signals with only very little amount of noise and edges and the texture component comprises of noise.

The blocky noise, which occurs due to the quantization of low frequency coefficients, and the mosquito noise, which occurs due to the quantization of high frequency coefficients, are separated into the texture component. The structure component gives the details of the edge components and by using this
information we remove the noise. Through this method, blocky noise and mosquito noise can be efficiently reduced without any information loss.

Section II provides an overview of the existing method. Section III describes the detail of our proposed method with the detailed explanation of the block diagram. The simulation results are shown in section IV. Concluding remarks is given in section V.

II. EXISTING METHOD

For reducing compression artifact in the image many methods are existing. But in this paper we mainly consider only compression artifact reduction method proposed by T. Goto [17]. In this method, first, we have the input image as the image with artifacts. Using the TV regularization decomposition method [16], this image is decomposed into structure component and texture component.

The structure component comprises of smooth signals with only very little amount of noise and edges and the texture component comprises of noise.

The blocky noise, which occurs due to the quantization of low frequency coefficients, and the mosquito noise, which occurs due to the quantization of high frequency coefficients, is separated into the texture component.

The structure component gives the details of the edge components and this is passed through the sobel filter to extract only the edge components. The edge information is passed to the Gaussian filter where only the edge components are filtered to remove the mosquito noise. Now, the Gaussian filter output is passed through a Deblocking Edge Filter (DEF) [7] to remove the blocky noise. Finally, DEF output and structure components are added to get the final output image with reduced compression artifact.

The drawback of this method is that only the texture component is passed through DEF. So the little noise present in structure component is not filtered. Also, even though artifact is reduced in the output image some information contents are also removed causing the degradation of picture which results in decreased PSNR value.

III. PROPOSED METHOD

In our proposed method, a new noise removable method which utilizes the TV regularization method in a different way is proposed.

A. Block Diagram

Block diagram of the proposed system is shown in Fig. 1.

First, we have the input image as the image with artifact. Using the TV regularization decomposition method [16], this image is decomposed into structure component and texture component. The structure component comprises of smooth signals with only very little amount of noise and edges and the texture component comprises of noise. The blocky noise, which occurs due to the quantization of low frequency coefficients, and the mosquito noise, which occurs due to the quantization of high frequency coefficients, is separated into the texture component.

The structure component gives the details of the edge components and this is passed through the sobel filter to extract only the edge components. The dotted line indicates that only the edge information from sobel filter is fed to the Gaussian filter. The edge information is passed to the Gaussian filter where only the edge components are filtered to remove the mosquito noise. Now, the structure component and the filtered texture component are added with the help of an adder and then it is passed through a Deblocking Edge Filter (DEF) [7] to remove the blocky noise. So, in the output image, the amount of compression artifact is reduced.

B. Total Variation Regularization

The Total Variation Regularization technique [17] which is used for the artifact removal is explained below. At first, the method of TV regularization was formed as a criterion for regularization in order to solve inverse problems. This method is very effective for regularization of images. Consider the following function

$$E(s) = \int |\nabla s| dxdy + \alpha \int |s_i - s|^2 dxdy$$  \hspace{1cm} (1)

The above equation is known as the ROF model for the original TV regularization. It was proposed by Rudin, Osher, and Fatemi. The TV regularization is a process which is used to minimize the function given by (1). In (1), $|\nabla s| dxdy$ is a TV term and $\alpha |s_i - s|^2 dxdy$ represents the constraint condition.

The $\alpha$ denotes how much the texture component is constrained to the original input signal. With the help of projected iteration technique [12] we can solve the problem of ROF minimization. The first work of the
image decomposition model is the work done by the Meyer (2001) which is known as the Rudin-Osher-Fatemi (ROF) algorithm.

In this paper, by using TV regularization decomposition method, the input image \( i \), is decomposed into two sub components \( s \) and \( t \). The first component \( s \) can be called as the structure component, which is well-structured with only very little amount of noise and it models the homogeneous objects which are contained in the image.

The second component \( t \) can be called as the texture component which contains both the textures and noise.

C. Structure Component
The structure component is obtained by the decomposition of the input image using the TV regularization. It comprises of the edge with only very little amount of noise and the smooth signals. The structure component gives the details about edge components and by using this information the Gaussian filter and deblocking edge filter are controlled.

D. Texture Component
The texture component is obtained by the decomposition of the input image using the TV regularization. It comprises of noise and textures. All the mosquito noise and blocky noise are isolated into the texture component. Also, all the DCT noise is contained only in the texture component. Therefore, if the texture component is low pass filtered, the DCT based compressed noise can be removed.

If simple filtering is performed on texture component, it leads to the loss of small texture components causing the degradation of picture. The method of selective filtering is adopted to solve this problem.

E. Sobel Filter
In our proposed method, the structure component is fed to the sobel filter. It is used to filter out all the signals in the structure component and also for extracting the edge components a threshold value is setted in the filter. Using the edge information from this filter we control the Gaussian and the Deblocking Edge Filter.

F. Gaussian Filter
In our proposed method, the texture component is fed to the Gaussian filter. The edge information from the sobel filter is also fed to the Gaussian filter. According to the information about the edges, the Gaussian filter process only the edge components of the texture component thus removing the ringing or mosquito noise. Here Gaussian filter acts as a deringing filter.

G. Deblocking Edge Filter
The Deblocking Edge Filter [7] which is used for the removal of blocking artifact is explained below. The filter operates at the encoder and decoder side across 8×8 block edges. The data which is reconstructed is then clipped to 0 to 255 range. Now we apply the filtering process. An additional clipping process is added in the filtering to confirm that the pixel values are in the 0 to 255 range. Consider the reconstructed picture. Assume two blocks, block1 and block2, where block2 is below or to the right of block1. Consider four pixel values in the reconstructed picture. These values are on a vertical and horizontal line of the picture. The Deblocking Edge Filter operates using these four pixel values denoted as \( P \), \( Q \), \( R \) and \( S \) where \( P \) and \( Q \) are in the block1 and \( R \) and \( S \) are in the block2. The figure 2 shows the example of these pixel positions. We will modify the pixel values \( P \), \( Q \), \( R \) and \( S \) if we want to apply filtering across the edges.

IV. SIMULATION RESULTS
To demonstrate our proposed method, which reduces the compression artifact, simulations have been performed. For simulation, Intel(R) Core(TM) i3 CPU M 380 @ 2.53 Ghz processor with a memory of 3.00 GB memory and MATLAB 2010 software is used. We apply our proposed method to test images such as Lena, Cameraman, and Pepper. All of the images were having a resolution of 512×512.

In this paper, comparison results are presented only for the Lena and Cameraman image since the simulation results for the other images are similar to those of these images. Fig. 3 shows the original images which have a resolution of 512×512.
An Effective Approach to Reduce Compression Artifact in Images


For giving input image, we compressed the image with high compression ratio and generated the image with artifact which is shown in Fig. 4.

The structure component comprises of smooth signals and edges. This component gives the details of the edge components and is given by Fig. 7. The structure component which gives the details of the edge components is passed through sobel filter to extract only the edge components. Output of sobel filter is shown in Fig. 8. The texture component and the edge information from the sobel filter are fed to the Gaussian filter. According to the information about the edges, the Gaussian filter process only the edge components of the texture component thus removing the ringing or mosquito noise. Output of gaussian filter is shown in Fig. 9.

A. Existing Method

Compression artifact reduction based on total variation regularization method proposed by T. Goto [17] is treated as the existing method in this paper. The drawback of this method is that, only the texture component is passed through DEF. So, the little noise present in structure component is not filtered. Also, even though artifact is reduced in the output image some information contents are also removed causing the degradation of picture which results in decreased PSNR value. Fig. 5 shows the output of the existing method.

B. Proposed Method

In the proposed method, using the TV regularization decomposition method the input artifacted image is decomposed into structure and texture component. The blocky noise, which occurs due to the quantization of low frequency coefficients, and the mosquito noise, which occurs due to the quantization of high frequency coefficients, is separated into the texture component which is shown in Fig. 6.

For giving input image, we compressed the image with high compression ratio and generated the image with artifact which is shown in Fig. 4.
In this paper, an effective approach to reduce the compression artifact in the compressed images has been proposed. In the proposed method, using the TV regularization decomposition method, input image is decomposed into structure component and texture component. The structure component comprises of smooth signals and edges with only very little amount of noise and the texture component comprises of noise. The blocky noise, which occurs due to the quantization of low frequency coefficients, and the mosquito noise, which occurs due to the quantization of high frequency coefficients, is separated into the texture component. The structure component gives the details of the edge components and this is passed through the sobel filter to extract only the edge components. The edge information is fed to the Gaussian filter where only the edge components are filtered to remove the mosquito noise. Now, the structure component and the filtered texture component are added with the help of an adder and then it is passed through a Deblocking Edge Filter (DEF) to remove the blocky noise. So, in the output image, the amount of compression artifact is reduced.

In the experimental results, it is clear that the quality of the image in the proposed system is higher when compared with the existing system, even though artifact is reduced in the output image, some information contents are also removed causing the degradation of picture which results in decreased PSNR value. But with the proposed method it is possible to achieve artifact reduced image with better PSNR value.

V. CONCLUSION

REFERENCES


An Effective Approach to Reduce Compression Artifact in Images


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MOBILE TOPOLOGY ENHANCEMENT IN WIRELESS NETWORK

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Abstract- In this paper a virtual cluster-backbone structure is formed and topology control is performed to maintain connectivity. A cluster based backbone infrastructure is proposed for broadcasting in MANETs. To broadcast packet the backbone of the network takes advantage of the cluster structure and makes use of cluster heads to forward the broadcast packet. With cluster based backbone infrastructure analysis is done for stable, partial and fully mobile nodes and the QoS parameters are analyzed. Network simulator (NS 2.39) is the tool used to obtain the simulation results.

Keywords: MANET, Topology Control.

I. INTRODUCTION

The emergence of portable wireless communication and computation devices and advances in the communication has resulted in the growth of mobile wireless networks. MANETs do not have a fixed infrastructure and consist of wireless mobile nodes that perform various data communication. MANETs [1] have applications in rescue operations, battlefield communications, mobile conferences, etc. Topology Control (TC) is one of the most important techniques used in wireless networks to reduce energy consumption and radio interference [2]. The goal of this technique is to control the topology of the graph and to maintaining connectivity while reducing energy consumption.

Clustering is an important approach to manage MANETs. In dense, dynamic ad hoc networks, it is very difficult to construct an efficient network topology. Clustering improves the system performance by reducing the battery power, by decreasing the cluster size and thereby increasing the link stability of large MANETs. With clustered architecture, nodes organize themselves into interconnected clusters. Each cluster consists of a cluster head, normal nodes and one or more gateways. Gateway connects adjacent clusters. A gateway [3] may connect two clusters by acting as a member of both, or it may indirectly connect two clusters by acting as a member of one and forming a link to a member of another. By clustering the entire network, the size of the problem into can be decreased into small sized clusters.

The wide network coverage is provided by constructing Mobile Backbone Networks (MBN). For these networks there are two types of nodes. They are backbone nodes and regular nodes. When compared to regular nodes, backbone nodes have superior communication capability. The information is routed through mobile backbone nodes to regular nodes. The clusters communicate through backbone node so that transmission overhead is less. To improve the routing efficiency, a virtual backbone routing scheme [4] is used. This structure can be used to manage the network topology when a node joins in or leaves the network. The routing can also takes the advantages of the virtual structure built based on these backbones.

The algorithm proposed constructs backbone architecture on a clustered MANET. There are leaf nodes, cluster heads and backbone nodes. The leaf nodes cannot be selected as cluster heads. Backbone nodes can be selected manually or by means of an algorithm. The main contribution is that the backbone formation is fault tolerant. In this paper a virtual cluster-backbone structure is formed and topology control is performed to maintain connectivity.

II. RELATED WORKS

In [5] clustering of nodes is done based on transmission range based clustering (TRBC). TRBC algorithm provides Topology management by making use of coverage area of each node and power management based on mean transmission power. Topology is constructed by reducing the transmission range of nodes and thus reducing energy consumption of each node. Cluster head selection is done among the incoming nodes based on WCA algorithm. First distances are calculated for each node that enter into the cluster.

The maximum distance of the nodes generated is taken as the radius for coverage area. The transmission range is then calculated. The distance of the nodes generated is compared with the average transmission range. If the transmission range of any node is within the average transmission range, then the nodes are considered to be within the cluster. TRBC reduces the number of clusters and the link stability is increased. It is crucial to implement topology control.
In [6] Cooperative Communication (CC) is a technology where same data is transmitted by multiple nodes. Power is saved and transmission coverage is extended.

The main objective is to increase connectivity and to save transmission power. To increase connectivity Centralized topology controlled algorithm is used. To maintain the network connectivity and minimize the number of links, a Minimum Spanning Tree (MST) based topology control algorithm is used. MST is generated in a distributed fashion by DMST algorithm. Thereby, the node can determine which CC links should remain and then it broadcast data to the final CC links, which includes the IDs of source nodes’ and the destination clusters’, to every node in the same cluster.

Finally, an MST is constructed where each link is a CC link and each node is a cluster. Even when there are more disconnected networks, the path loss tends to be smaller. Cooperative Bridges scheme increases the connectivity while consuming transmission power compared to other existing topology control schemes.

In [7] the overall energy consumption is reduced and the system lifetime is maximized. Topology control protocols are used to eliminate the inefficient links in the network because fewer amounts of links reduce the complexity in finding the route to the destination. Here a Fair and Efficient Topology Control (FETC) is used. The protocol is divided in two phases. The first phase is the neighbor discovery phase where each node selects its neighboring nodes. The neighbor is selected based on node eligibility criterion. The second phase is to construct a symmetric graph based on already built graph in phase 1. The symmetry is obtained by adding the reverse edge to every asymmetric link. The advantage is that the network life time is increased but message complexity is 2n since it has to send 2 messages to determine the neighbor.

III. PROPOSED WORK

The topology of the proposed work is depicted in Fig.1. The model above shows a cluster backbone structure. The structure consists of 100 nodes which are randomly deployed. It includes leaf nodes, cluster heads and backbone nodes.

The nodes on the boundary are selected as leaf node and cluster heads cannot be selected from leaf nodes. Cluster head is selected based on node degree. Nodes with node degree more than two is selected as the cluster head. Node which is far away from the cluster head but within the transmission range is selected as the backbone node. The backbone node is used to connect cluster backbone and thereby maintains connectivity.

A. Virtual Cluster-Backbone Construction

The boundary nodes are selected as leaf nodes which cannot take part in cluster head selection. Nodes with largest node degree and transmission range are selected as the cluster head. The backbone selection algorithm is a self-running algorithm, which does not need any infrastructure. Nodes with backbone selection metric values larger than the threshold can be selected as the backbone node. The backbone selection metric are power, connectivity and immobility.

The metric connectivity is defined as the measure of connectivity among neighbors. Let Na be the neighbors of node a, including the node itself. Depend on requirements we can set different base B. If Na is larger than B, then the connectivity metric is set up as 1. Otherwise, it is set as the Na/B, which should be bigger than 0 and smaller than 1. The power status is the measure of power capability that can support a node to function properly. The power is expressed as a fraction of 1. Higher the value, the more powerful the node is. The immobility metric is the moving speed of a node and is expressed as a fraction of 100. Lower the immobility value more stable the node is. With these three metrics, we define the selection formula as:

\[ \text{Metric} = \left( \frac{k_1 \times C}{k_2 \times P} \right) / \left( k_3 \times S \right) \]

where the ‘k’ variable. The selection process selects backbone nodes from normal nodes. In the beginning, all nodes are treated as normal nodes. Then every node runs the backbone selection algorithm and get the value of node state. Nodes with N=1 are selected as the backbone nodes.

Then select a root node U of one cluster and a destination node V from another cluster. Add an edge E from U through cluster heads and backbone node till it reaches destination V. Then form at least 3 such
cluster-backbone paths from \( U_i \) to \( V_i \) such that \( v_1, v_2 \) and \( v_3 \in B_s \).

**B. Topology Maintenance**

If any link \( l_i \) breaks alternate the route using \( B_{s1} \), where \( B_{s1} \) is the cluster-backbone structure towards destination. If \( B_{s1} \) fails then use \( B_{s2} \), where \( B_{s2} \) is the next structure with the shortest path. Continue the steps until it reaches the destination.

**C. Algorithm:**

1. Input: Unconnected network graph \( G \) // Initialization
2. Output: virtual cluster backbone structure
3. Node with largest node degree is taken as cluster head.
4. Calculate transmission range of cluster head
   \[ Tr = \left( \frac{ndd}{ndc} \right) \left( \frac{\text{coverage area}}{1} \right) \]
5. \( N = 0 \) for normal node and \( N = 1 \) for backbone node
6. Calculate selection metrics:
   \[ \text{Metric} = \left( k_1 \times C \right) \times \left( k_2 \times P \right) \div \left( k_3 \times S \right) \]
7. If \( \text{Metric} \geq L \) threshold and (node lies within the transmission range calculated) then
8. \( N = 1 \).
9. Select a root node \( U_i \) and destination node \( V_i \) from clusters \( C_i \) (i=1, 2, 3…n).
10. Insert edge \( E_i \) to \( U_i \) upto \( V_i \) through cluster heads and backbone node (i= 1,2,3…n) # virtual cluster-backbone formation
11. if more than three \( v_i \)'s are in same coverage area then
12. Construct a virtual backbone structure, where \( v_1, v_2 \) and \( v_3 \in B_s \),
13. else go to step 9.
14. if link \( l_i \) is disconnected then // Alternate routing 1
15. alternate the route using \( B_{s1} \), where \( B_{s1} \) is the cluster-backbone structure towards destination.
16. if \( B_{s1} \) fails then // Alternate routing 2
17. use \( B_{s2} \), where \( B_{s2} \) is the next structure with the shortest path.
18. continue until reaches the destination.
19. else wait for connectivity or go to step 15.
20. end.

**IV. PERFORMANCE ANALYSIS AND SIMULATION RESULTS**

In the proposed work nodes are randomly deployed. A Mobile Ad hoc network is considered here. As per the algorithm virtual cluster-backbone structure are formed and topology control is implemented. The performance of the network is evaluated in terms of Throughput, Packet Delivery Ratio (PDR), and Delay parameters. Mesh structure created, Hop count, Overhead, Scalability defined as follows,

a). Packet Delivery Ratio (PDR) is defined as the ratio of the total number of successfully transmitted data packets to the total number of data packets sent from the source to the destination.
b). Throughput is the average rate of successfully transmitted data packets over the communication channel’s capacities.
c). End-to-end delay refers to the time taken for a packet to be transmitted across a network from source to destination.
d). Number of Dropped Packets is the difference between total number of packets sent from source to destination and the total number of successfully transmitted data packets.
e). Hop Count is the maximum number of hops required to reach from a source node to destination.

In the simulation three cases are considered:

1. Static nodes
2. Partially mobile nodes
3. Fully mobile nodes

**Throughput** or network throughput is the average rate of successful message delivery over a communication channel. This data is delivered over a physical or logical link, or pass through a certain network node. The throughput is usually measured in bits per second (bit/s or bps), and sometimes in data packets per second or data packets per timeslot.

The system throughput or aggregate throughput is the sum of the data rates that are delivered to all terminals in a network. Throughput over analog channels is defined entirely by the modulation scheme, the signal to noise ratio, and the available bandwidth. Fig. 2 shows the throughput \( V_s \) time graph. Here the throughput increase with time for Stable, partial and fully mobile nodes.

**Packet delivery fractions (PDR)** are the ratio of the data packets delivered to the destinations to those generated by the CBR sources. The PDR shows how successful a protocol performs delivering packets from source to destination. The higher for the value give use the better results. This metric characterizes both the completeness and correctness of the routing protocol also reliability of routing protocol by giving its effectiveness. This ratio directly affects the maximum throughput that the network can support. Packet delivery ratio increases with an increase in

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**Fig. 2 Throughput**
time for stable nodes. Fig 5.3 shows the PDR $V_t$ time graph where PDR decreases with time. When nodes are partially mobile PDR is less when compared to stable. PDR further decreases when they are fully mobile.

Average end-to-end delay is an average end-to-end delay of data packets. It also caused by queuing for transmission at the node and buffering data for detouring. It includes processing delay, queuing delay, propagation delay, transmission delay. As number of loops increases interference, congestion etc. will be more. So packets need to be re-routed will be queued and therefore encounter longer delays. The delay of a network is the time it takes for a bit of data to travel across the network from one node or end point to another. It is typically measured in fractions of seconds. Delay differs slightly, depending on the location of the specific pair of communicating nodes. This metric describes the packet delivery time: the lower the end-to-end delay the better the application performance.

Number of Dropped Packets is the difference between total number of packets sent from source to destination and the total number of successfully transmitted data packets. Fig 5.5 shows the dropped packets $V_t$ time graph where it increases with respect to time delay, traffic and many other factors.

Hop Count is the maximum number of hops required to reach from a source node to destination. Hop count refers to the intermediate devices (like routers) through which data must pass between source and destination, rather than flowing directly over a single wire. Each router along the data path constitutes a hop as the data is moved from one network to another. Hop count is therefore a basic measurement of distance in a network. The hop count for mobile network was more than fixed network. Nodes move, the links and routes get disrupted. So the hops required to reach the destination will also increase. Hop counts are normally used to find faults in a network, or to discover if routing is correct. Hop count is plotted against number of nodes in Fig.5.6 As number of mobile nodes increases the hop count will also increase as the number of intermediate nodes will increase as number of hosts increases.

V. CONCLUSION

In this paper a virtual cluster is constructed. The main contribution is that the cluster-backbone formation is fault tolerant. Cluster-backbone architecture is a network control structure that reduces interference in a multiple access broadcast environment by forming distinct clusters of nodes in which transmissions can be scheduled in a contention free manner. With backbone-cluster architecture, nodes autonomously organize themselves into interconnected clusters. The communication between clusters is done through backbone node so that transmission overhead is less
structure built based on these backbones. A virtual cluster-backbone structure is formed and topology control and the goal of this technique is to control the topology of the graph representing the communication links between network nodes with the purpose of maintaining connectivity. The main contribution is that the backbone formation is fault tolerant. Instead of transmitting with the maximal power, each node in a wireless multi-hop network executes the topology control algorithm to adjust its transmission power, so that the proper network topology can be defined. The nodes in the structure are considered to be static, partial and fully mobile and the QoS parameters are analyzed. From the simulation results it can be analyzed that the algorithm outperforms other algorithms in terms of better throughput, packet delivery ratio and lesser delay. The proposed algorithm ensures better connectivity in a highly mobile environment. Thus it is adaptive to dynamic nature of the network and is effective in maintaining the connectivity and the topology even if the nodes are mobile.

REFERENCES


SEPARATION OF MECG AND FECG USING THE CCA-EMD ALGORITHM

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Abstract- Joint analysis of multi set data arises in many applications such as while working with multi-subject or multi-condition medical data, hyper spectral data, or transformed data in multiple bins. Hence, algorithms that can achieve source separation jointly on those multiple datasets while fully utilizing the multivariate nature of the whole data are desirable in many applications where blind source separation (BSS) has been successfully applied. Here in this paper two BSS methods, CCA and EMD have been combined. In CCA-EMD, the goal is thus to achieve a separation of given multi set data such that the estimated sources are aligned for each dataset. Denoising the bipolar biomedical signals so far is a common method using the already existing processes ranging from wavelet filtering to use of adaptive filtering , BSS , JBSS and ICA and the combination of all the above. But actual signals are monopolar and hence new techniques are to be employed. This paper proposes an algorithm to solve the denoising of monopolar signals especially concerning the biomedical signals. The simulation results on a electrocardiogram (ECG) data separation problem are studied and their performance is measured. Comparison of CCA-EMD with the JBSS algorithm is given here to prove that the present method gives signals that have been retained by the new method. The proposed method removed the noises and separated the signals successfully. The CCA-EMD algorithm performed considerably better than the JBSS method.

Keywords- CCA, EMD, JBSS, ECG

I. INTRODUCTION

Electro-hysterogram is a noisy signal recorded by bipolar electrodes placed on the surface of the abdominal skin of the pregnant woman. During labor, poor progress and quantitative assessment of abdominal activity guides the doctor to choose a uterine contraction induction or augmentation, a C-section, or other therapies. Furthermore, monitoring the fetal heart response to the uterine activity (cardiotography) is widely used as a screening test for timely recognition of fetal distress (e.g. asphyxia) [2] [3]. For all these, need of EHG becomes an useful observation tool. So far, denoising was applied on bipolar EHG signals which are stationary. But in reality these biomedical signals are non stationary and thus monopolar EHG signals have to be considered. Combination of already existing methods is used to denote the EHG monopolar signals. During recordings, EHG is corrupted by fetal movements, the muscle contractions within the uterus, the electronic and electromagnetic noise as well ECG of mother and fetus. The EHG is a signal that has very low frequency and has very low amplitude as compared to various sources of contaminating noise.

Wavelet filtering [6] and JBSS (Joint BSS) have been used successfully on bipolar EHG for removing maternal and fetal ECG as well as electric noises. Most wavelet filtering techniques assume that the noise is of low amplitude and stationary when compared to the main signal of interest [1]. However, in monopolar EHG, the noise is non-stationary and usually of higher amplitude than the signal of interest. In addition, many sources of noise signals have their main frequency components closer to the frequency of the signal of interest. The EHG has most of its energy in the frequency band of 0.1 to 1.5 Hz, as compared to maternal cardiac frequency (typically 1.2 Hz), maternal respiration (typically 0.2 Hz), and fetal respiration (typically 1 Hz). And as the noise is in the same frequency bands and is of high amplitude and is sporadic, it cannot be rejected by wavelet filters or the filters used in the JBSS process. The drawbacks in using bipolar electrodes in a system are that a meaningless or biased analysis of the direction of propagation is obtained, because the direction of propagation along the line between the two bipolar electrodes is privileged. Measuring bipolar signals of electric phenomena are based on rejecting the part of the signal that is measured on both electrodes and keeping only the part that is dissimilar to the two electrodes. Assumptions underlying this technique are that the “local” electrical activity is not similar to the two electrodes and that the common part is afield. Thus, making it, not relevant. It is anyway highly uncertain that this holds for the case of EHG. So there are chances that important information is lost by using bipolar signals only. Hence the need to go for monopolar signals.

Traditional methods of Blind source separation (BSS) methods such as the Independent Component Analysis (ICA) and Principal Component Analysis (PCA) are being used in biomedical signal processing which involves the analysis of multiple variate time series data such as EMG [5], [8].


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Also there are methods of extracting the fetal ECG from the EHG based on the BAF approach (Blind Adaptive Filtering). Diseases like arrhythmia can be found out using a moving window [3]. This method has out-rated the ICA methods.

Lately, a new method for muscle artifact elimination in scalp EEG has been developed that doesn’t have the disadvantages of ICA [4]. The process is based on the combination of statistical canonical correlation analysis (CCA) method applied as a BSS technique, called as BSS_CCA. This method has shown considerably better performance than ICA in some applications [4], [5].

In this paper, the aim of CCA is to extract the uterine bursts. It is based on the hypothesis that the bursts have a higher autocorrelation coefficient than noise. The bursts observed after applying CCA only contain fewer artifacts than the original signals but still contain the artifacts that have high autocorrelation.

The empirical mode decomposition (EMD) was chosen as a second step to remove this noise from the extracted burst. The EMD technique was introduced by Huang et al. [9] to analyze non-stationary and nonlinear signals. The EMD has become a very important tool to analyze biomedical signals. The use of EMD for analyzing esophageal manometric data in gastro esophageal reflux disease indicated better performance in removing different kind of artifacts (respiratory, motion, etc.) from electrogastrogram signals. The EMD approach also proved to be efficient in removing artifacts from ECG signals [12]. Here it was demonstrated how reconstruction using the Hilbert–Huang transform can successfully be applied to contaminated ECG data for the purposes of removing unwanted ocular artifacts. More recently, Wu and Huang have introduced a noise assisted version of the EMD method, called ensemble empirical mode decomposition (EEMD) [6]. This method has shown better performance than EMD as it extracts the intrinsic mode functions (IMFs) in a manner so that the mode mixing disadvantage of the EMD method is corrected.

Wavelet Transform (WT) denoising methods could also be good candidates for the second step of the processing presented here. However, in this first attempt EMD was chosen rather than WT methods for two reasons: 1) EMD is a data-driven algorithm. It decomposes the signal in a natural way without prior knowledge about the signal of interest embedded in the noise, which is not the case the wavelet transform; 2) EMD demonstrated better performance than WT when combined with ICA in denoising EEG signals [9]. The aim of this paper is to combine the use of BSS_CCA and EMD algorithms to design a new tool, called CCA-EMD, to remove the main interferences embedded in monopolar abdominal EHG recordings. The method is applied to real signals recorded on women during pregnancy and labor.

II. MATERIALS AND METHODS

A. Measurements

The FECG measurements are taken from [14], [15]. These are already recorded ECG signals from a pregnant woman whose ECG is mixed with the fetal ECG signals within the womb.

From the figure below it is seen that the mixed ECG signals taken from the womb are passed through the CCA algorithm to obtain separated mother and fetal signals but which are corrupted. Later EMD process removes the external noise caused by the electronic devices to get clear MECG and FECG.

![Figure 1. Block Diagram showing the CCA-EMD process](image)

Fig.1 shows the mixed ECG signal plot. The sampling rate is 250 Hz and T=2500 samples are recorded. From fig.1, it is seen that the strong and slow heart beat signal of the mother in all these eight channels, and the weak and fast one of the fetus in the first several channels. The task is to separate out the weak fetal ECG signal. The first 5 channels shown in the fig. are the signals taken from the abdomen whereas the last 2 signals have been taken from the thoracic region of the pregnant lady.

![Figure 1. Mixed ECG signal in MATLAB format](image)

B. Canonical Correlation Analysis (CCA)

CCA is a process of solving the BSS problem of contrast functions. In ICA (Independent Component Analysis), the estimated sources are considered to be
non Gaussian as possible. But in CCA, the sources are forced to be maximally auto correlated and mutually uncorrelated while the mixing matrix is taken to be a square matrix. In CCA the linear relationship between the two multi dimensional variables are found by finding two sets of basis vectors such that the correlation between the projections of the variables onto these basis vectors are maximized [1]. On considering the observed data matrix X(t) and its delayed version Y(t), their corresponding canonical correlations will be found based on a basis vector or a Canonical Variate. The correlations between the successively extracted canonical variates would be gradually smaller and smaller. The Correlation Coefficients are proportion of correlation between the canonical variates accounted for by the particular variable, X(t) and Y(t).

The canonical correlation between X and Y can be calculated by solving these equations

\[ C_{xx}^{-1}C_{xy}C_{yy}^{-1}C_{yy}w_x = \rho^2 w_x \]  
\[ C_{yy}^{-1}C_{yy}C_{xx}^{-1}C_{xx}w_y = \rho^2 w_y \]  

where \( \rho \) is the canonical correlation coefficient, as the square root of the Eigen value and Eigen vectors as \( w_x, w_y \). The CCA gives the source signals that are uncorrelated with each other. They would be also maximally autocorrelated and ordered in decreasing autocorrelation. Application of CCA includes Speaker Recognition and Image Retrieval.

D. Empirical Mode Decomposition (EMD)

EMD is a nonlinear and data driven technique used on non stationary signal decomposition. Here the main idea is to represent a signal as a sum of components, each of them being a Zero-mean AM-FM function. Decomposing a complicated set of data into a finite number of Intrinsic Mode Functions (IMF), that follows a well behaved Hilbert Transforms. In EMD, Partial Signal Reconstruction by appropriate selection of IMFs is performed and exclusion of those IMFs that contribute to the noise contamination and feature distortion of the signal of choice.

Processing Procedure:
1. Respiration signals were monitored in X,Y axes by measuring the acceleration
2. Application of the EMD on each axis signal
3. Application of the spectral criteria on each IMF of 2-axes respiratory signal
4. Evaluation of the EMD based technique was aided by metrics computation (Cross Correlation Coefficients)

E. CCA-EMD

De-noising by EMD is in general carried out by partial signal reconstruction, which is based on the fact that noise components lie in the first few IMFs. BSS is the best way to extract the uterine bursts and based on the hypothesis that the sources of uterine bursts have higher autocorrelation than the sources corresponding to the artifacts, CCA method was chosen as a way to extract the uterine bursts and in the same time eliminate all the low autocorrelated noise. The sources of device noise (electronic artifacts) are highly autocorrelated and therefore it is not possible to remove it by using only the CCA method. It is shown that EMD shows better performance in removing this kind of noise [12]. For this and other reasons, EMD was chosen as the complementary tool to remove the residual electronic noise. Thus this combination is called the CCA-EMD algorithm.

III. RESULTS

Fig. 2 and 3 shows the mixing matrices plotted for applying the BSS approach on the Mixed ECG signals. ECG signals of the original and delayed signals are plotted. Initially the size of the signal is obtained from where it was determined if the signal is a 1-D or 2-D signal. Corresponding mixing matrices A and B are found using the BSS-CCA combined approach.

Fig. 4 and 5, here the correlation coefficients are calculated and their corresponding variates are found. This helping in easily recognizing the maternal and fetal signals. The plots are made on the basis of checking out the dimensions and they are maximized. The covariates found are then inverted. These are multiplied with the original and delayed signals to obtain the corresponding signals.
Separation of MECG & FECG using The CCA-EMD Algorithm

IV. RESULTS AND DISCUSSION

B. JBSS (Joint BSS)
In this algorithm, separation of the FECG and MECG signal from the recorded ECG signal so that there is a correct diagnosis obtained during pregnancy. In the JBSS algorithm, the noise has been removed using the Butterworth filter. Then the process of pre-whitening was done for uniform distribution. After that construction of cumulant matrix was done to convert the original matrix into one dimensional matrix which was followed by joint diagonalization. Finally the separation was performed to get the independent mother and fetal signal.

C. JBSS vs CCA-EMD
In Fig 7 and 8 shows the ECG plots for JBSS and CCA-EMD. The maternal and fetal signals can be identified by the difference in the frequency rates. As it is seen, in a time span of 1sec, there are two peaks showing a fetal signal which is less in the maternal signal. We see that CCA-EMD process shows better separation results.

Figure 3. Delayed ECG signals

Figure 4. CCA_BSS on Original Signals
Once the CCA process is over, the signals are passed to the EMD algorithm where the corresponding IMFs are found using the step by step procedure just like mentioned before. In Fig.6 the repeated IMF findings shall in turn reduce the amount of noise and thus make it visible for us to identify the mother and fetal signals which are plotted independently. After the EMD is found, the signal to interference ratio is found and compared with JBSS.
D. Parameter Comparison

<table>
<thead>
<tr>
<th>Methods</th>
<th>SIRf</th>
<th>SIRm</th>
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<tbody>
<tr>
<td>CCA-EMD</td>
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<td>77</td>
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<tr>
<td>JBSS</td>
<td>62</td>
<td>67</td>
</tr>
</tbody>
</table>

The above table shows that the signal-to-interference ratio for CCA-EMD is better in comparison to the JBSS measurements. The value of SIR in CCA-EMD for the maternal signal is 10 times better than the JBSS SIR value. Similarly the SIR values for the fetal signals while CCA-EMD used was 12 times improved than the JBSS case.

V. CONCLUSION

This work includes the separation of the FECG and MECG signal from the recorded ECG signal so that a correct diagnosis can be given during pregnancy. The algorithm used is CCA-EMD (Canonical Correlation Analysis and Empirical Mode Decomposition).

Here first the uterine bursts are found by the BSS-CCA process and the noise is removed using EMD technique. This is a fast and economical method to separate as well as remove noise from the EHG signals. The EMD threshold value could be computed or visually selected to remove the IMFs in the GRAPH of the signals for the MECG and FECG.

At the end of the CCA-EMD process a noise free separated maternal and fetal ECG signal is obtained. The significance of this work is a better approach in findings out the fetal related heart disorders as well as growth process while still in the womb. Experiments are carried out in stages of pregnancy and during labor. The SIR values have been calculated and found out to be that the SIR values on application of CCA-EMD for mother and fetus are 10 and 12 times more than the values of compared methods.

Detecting the fetal heart beat helps in understanding the chances of various heart problems which could be diagnosed as soon as the baby is born or while still inside the womb. Diseases like hypoxia, Down’s syndrome, arrhythmia can also be found out through the heart waves.

REFERENCES


Separation of MECG & FECG using The CCA-EMD Algorithm


DESIGN OF ULTRAWIDEBAND MONOPOLE ANTENNA WITH MINIMUM GROUND PLANE EFFECT

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1,2Department of Electronics and Communication, Karunya University, Coimbatore, India

Abstract- A Planar ultra wideband antenna design is analyzed for increased impedance matching in the Ultrawideband (UWB) range (3.1GHz to 10.6GHz). Also the effect of the ground plane is minimized by cutting slot on the ground plane. Impedance matching of Ultrawideband (UWB) antenna can be improved by introducing simple microstrip transitions between the 50-ohm feed line and the printed disc. In this paper a dual step feed is proposed between the feed line and radiator. It also offers a very simple geometry suitable for low cost fabrication and straightforward printed circuit board integration. Here triangle slot is provided on the ground plane in order to reduce the ground plane effect. The radiator used here is elliptical disc.

Keywords- UltraWideBand ; impedance matching ; monopole elliptical disc antenna

I. INTRODUCTION

Ultra Wide Band technology is a promising technology mainly used for short range and high speed wireless communications. The frequency band 3.1GHz to 10.6GHz is allocated by the Federal Communications Commission (FCC) for UWB technology in 2002 [1]. One of the main component of UWB system is the front-end antenna unit. The main applications of Ultra Wide Band antennas are mine detection, transient radars, indoor wireless radio, medical imaging systems and unexploded ordnance location and identifications. Various types of antenna have been designed for UWB applications, but the planar monopole antenna have been mostly used. This is because of its low cost, low profile, ease of fabrication and compact size. Various types of disc like circular, elliptical, square, rectangular, hexagonal etc. are used [8] among these elliptical and hexagonal shaped discs are provided good performance [5], [7]. Since using printed structure, it can easily integrate into UWB devices with very low cost. In order to achieve a broad impedance bandwidth, radiator and ground plane shapes as well as the feeding structure should be optimized.

In this paper we mainly discussed how to improve the impedance matching of monopole antenna and then how to reduce the ground plane effects on antenna performance. Here elliptical disc monopole antenna is discussed in order to improve the impedance matching and radiation efficiencies. The good impedance matching can be mainly achieved by providing steps between feed line and disc [2]. Commonly used feeding techniques for antennas are microstrip line feed, coplanar wave guide feeds and slotted structure. In this paper use microstrip line feed since it offers low cost for antenna design, and provide good impedance matching in terms of 50-ohm impedance matching and radiation behavior. The performance of the antenna like impedance matching, radiation behavior and bandwidth are mainly affect the ground plane size and shape. Here only partial ground plane is used and the effect of ground plane on antenna performance can be minimized by cutting slots on the ground plane [3], [4], [6], [9]. Here a triangle slot is cut on the upper edge of the ground plane.

II. ANTENNA DESIGN

The top view of proposed elliptical disc monopole antenna is shown in the Fig1. The antenna is printed on a dielectric substrate with thickness 0.83mm and relative permittivity of 3.38. Antenna consists of an elliptical disc radiator and a microstrip line feed. A dual step transition is provided between the feed line and elliptical disc. The partial ground plane is used and a triangular slot with length ‘la’ and height ‘lb’ is cut on the upper side of the ground plane as shown in the Fig 2.
By optimizing the values of width and length of microstrip line, ground plane size and elliptical disc diameters a good impedance matching can be obtained. Normally ground plane influences the impedance matching and radiation behavior of the antenna. By cutting triangular slot on a ground plane with optimized values this ground plane effect can be reduced. The optimized values of antenna dimensions are shown in the Table 1.

### Table 1. Dimensions of the Monopole Antenna

<table>
<thead>
<tr>
<th>Dimensions</th>
<th>Millimeter (mm)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Length of the Substrate (L)</td>
<td>35</td>
</tr>
<tr>
<td>Width of the Substrate (W)</td>
<td>25</td>
</tr>
<tr>
<td>Major Axis of the disc (a)</td>
<td>8</td>
</tr>
<tr>
<td>Minor Axis of the disc (b)</td>
<td>6</td>
</tr>
<tr>
<td>Length of the Feed line (L3)</td>
<td>8</td>
</tr>
<tr>
<td>Width of the Feed line (W3)</td>
<td>2.4</td>
</tr>
<tr>
<td>Length of the First microstrip line (L2)</td>
<td>5</td>
</tr>
<tr>
<td>Width of the First microstrip line (W2)</td>
<td>2</td>
</tr>
<tr>
<td>Length of the 2nd microstrip line (L1)</td>
<td>3</td>
</tr>
<tr>
<td>Width of the 2nd microstrip line (W1)</td>
<td>1.6</td>
</tr>
<tr>
<td>Length of the partial ground plane (lg)</td>
<td>13.6</td>
</tr>
<tr>
<td>Length of the triangular slot (la)</td>
<td>3.6</td>
</tr>
<tr>
<td>Height of the triangular slot (lb)</td>
<td>2.6</td>
</tr>
</tbody>
</table>

### III. SIMULATED RESULTS

The elliptical disc monopole antenna is designed and simulated by FEKO software and antenna parameters are analyzed and discussed in the following sections.

#### A Return Loss

Return loss is an important parameter in the antenna performance; it gives how much power is lost while radiating the antenna. If there is any discontinuity in the transmission line more loss occurs. Return loss (RL) can be expressed in terms of power as given as in (1):

\[
RL (\text{dB}) = 10\log_{10} \left( \frac{P_i}{P_r} \right) \quad (1)
\]

Pi is the incident power and Pr is the reflected power. Return loss is mainly expressed in terms of scattering parameters or ‘S’ parameter. It describes the response of the N-port network. The first number in the subscript of S parameter represents responding port and second number represent incident port. In the case of S11 both responding and incident ports are same; hence it represents how much power is returned back i.e. its return loss. In order to achieve good impedance matching antenna should be matched with transmission line i.e. reduce the power reflected from the antenna and maximize power delivered to the antenna. The S11 parameter of elliptical disc monopole antenna is shown in the Fig 3. From the graph it is clear that the first resonance is occurring at 3.2 GHz and corresponding S11 value is -48 dB, it indicates good impedance match i.e. good matching between the antenna and transmission line. The frequency at which first resonance value occurred is mainly depends on the size of the elliptical disc. Second resonance value occurred at 7.5 GHz and corresponding S11 value is -40.5 GHz.

#### B Minimizing the Effect of Ground plane

When antenna is fed by microstrip line feed vertical current mode will exited on the radiator. Excitation of first vertical mode depends on dimensions of the radiator, it determines lowest usable frequency and interaction between first and higher order modes decide the highest achievable frequency. The size and shape of the ground plane mainly influence the impedance matching. In order to reduce the effect a triangular slot is cut on the upper side of the ground plane. Fig 4 indicates the S11 parameter of the elliptical disc monopole antenna without slot in the ground plane. In this case if there is any change in ground plane it influences the S11 parameter. From this graph it is clear that when we change the width of the ground plane, frequency at which resonance occurred is changed, if there is no slot in the ground plane. Fig 5 indicates the S11 comparison of antenna for varying the ground plane width with slots in the ground plane. From the graph it is clear that there is no so much difference in the
resonance frequency when changing width of the ground plane

C. Radiation Pattern

Radiation pattern gives radiated power variance as a function of direction away from the antenna. This power variation can be calculated in far field or near field. The region close to the antenna is called near field or induction field and region far from the antenna is called far field region or radiation field. Antenna patterns are normally measured in the far field region. The 3-D radiation pattern of the elliptical monopole antenna is given in Fig 6, 7. Since the ground plane is partial antenna radiates below the ground plane also. Fig 6 shows the 3D radiation pattern of antenna at first resonance frequency i.e. 3.2 GHz and Fig 7 shows the 3D radiation pattern of antenna at second resonance frequency i.e. 7.5 GHz. Radiations pattern can be drawn in polar plot also as shown in the Fig 8. It gives value of E-field in dB at each angle.

D. Current Distribution

The total current distribution on the antenna is the superposition of the characteristic currents with appropriate weighting coefficients. The current distribution of elliptical monopole antenna is varied at each frequency. At resonance frequency maximum current is distributed across the antenna compared with other frequencies. Comparison between the current distribution of resonance frequencies and other frequencies as shown in Fig 9. Resonance frequencies are 3.2 GHz and 7.5 GHz, in these frequencies more current is distributed compared with other frequencies like 5 GHz and 11 GHz.
IV. CONCLUSION

Elliptical disc monopole antenna for UWB application were designed and analyzed. By using a dual step feed between the feed line and elliptical disc, better impedance matching can be obtained in the UWB range. Further improvement can be obtained by providing additional steps between the feed line and disc. Effect of ground plane is reduced by cutting slot on the upper side of the ground plane.

REFERENCES


DESIGN OF WIDEBAND E-SHAPED MICROSTRIP ARRAY
ANTENNA FOR WIRELESS APPLICATIONS

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Abstract- The microstrip antenna offers a wide range of applications because of the advantage that it is a low profile, narrow bandwidth, compact and high gain antenna. The limitation of narrow bandwidth and poor impedance matching was investigated in this paper. New E-shaped design for microstrip arrays was proposed. As a commercial simulation tool, CADFEKO software, a 3D electromagnetic simulator was used. This two-dimensional wideband array is designed for wireless applications, mainly for frequency range around 2.45 GHz ISM band.

Keywords- Antenna array; Wireless Local Area Network (WLAN); impedance matching; VSWR

I. INTRODUCTION

The WLAN technology that uses a single antenna element has a broad beamwidth and can provide low directive gain [1] which is not practical for applications which require higher gain such as long-distance communications and those needs narrower beamwidth such as radar systems. One possible solution is to increase the electrical size of the antenna so that the antenna can achieve higher gain and narrow beamwidth, but it is inefficient and costly. An alternative way without enlarging the antenna element is to construct an array of antenna elements, which forms a group of radiating elements assembled in such a way as to obtain a high directivity, or by adjusting the array parameters such as geometrical configuration, element spacing, excitation and so on. The ways to improve bandwidth include, introduction of additional stacked or coupled patches [7]-[8], but it makes the element configuration larger. Another method is to use aperture fed patches. But, it makes the design complex. Using PIFA’s introduces walls or vias in their topology, which was more expensive to produce. So, the best alternative is to use specially shaped patches, the E-shaped or slotted patches in the design of the antenna. Wide bandwidth achievement and size reduction are becoming major design considerations for practical applications of microstrip antennas due to the rapid demand. But, usually achieving one characteristics result in degradation of the other.

In this paper we mainly discussed how to improve the impedance matching and bandwidth of the microstrip array antenna. Here, an array design using the E-shaped antenna design with slots different from [9] is discussed in order to improve the impedance matching and radiation properties. Better impedance matching and therefore high gain can be achieved by using these four E-shaped patches. Feeding techniques for microstrip antennas include microstrip line feeding, co-axial feeding, aperture coupled feed and proximity coupled feed. In this paper we use co-axial feed. The advantage of this feeding scheme is that the feed provides good impedance matching and also it is easy to fabricate and has low spurious radiation. The performance of the antenna like impedance matching, radiation behavior and bandwidth are mainly affected by the ground plane size, shape and distance between the ground plane and the substrate.

Antenna design is performed on CADFEKO 5.5. FEKO can be used for various types of electromagnetic field analyses involving objects of arbitrary shapes. FEKO is a software Suite intended for the analysis of a wide range of electromagnetic problems. Applications include EMC analysis, antenna design, microstrip antennas and circuits, dielectric media, scattering analysis and many more. The kernel provides a comprehensive set of powerful computational methods and has been extended for the analysis of thin dielectric sheets.

II. ANTENNA DESIGN

Figure 1. Proposed E-shaped patch design geometry

In this paper we mainly discussed how to improve the impedance matching and bandwidth of the microstrip array antenna. Here, an array design using the E-shaped antenna design with slots different from [9] is discussed in order to improve the impedance matching and radiation properties. Better impedance matching and therefore high gain can be achieved by using these four E-shaped patches. Feeding techniques for microstrip
multiple homogeneous dielectric bodies and planar stratified media.

The geometry of the proposed microstrip antenna array is shown in the Fig1. The proposed Antenna consists of four E- shaped patches, of which three of them is formed by A,B C and D, E forms the third as shown in figure 1. All the dimensions of the antenna is in mm. The values of dimensions a, b is 2 mm and x is 4 mm. It uses standard FR4 substrate with thickness 0.8 mm, dielectric loss factor = 0.014 and relative permittivity = 4.7. The ground plane is perfect electric conductor formed below the substrate. The ground plane is of width 154 mm and length 206 mm. The distance between the substrate and the ground plane as in [9] is not used, instead the ground plane is placed closely under the substrate. It uses an integrated feeding using a 50 ohm coaxial feed.

III. SIMULATION RESULTS

A Impedance Matching

The theory of maximum power transfer states that for the transfer of maximum power from a source with fixed internal impedance to the load, the impedance of the load must be the same of the source which is called Jacobi’s law ,can be represented with the help of eqn(1) as,

$$Z_s = Z_L \quad \text{*}$$

(1)

Where $Z_s$ = impedance of the source.

$Z_L$ = impedance of the load.

( * ) indicates the complex conjugate.

Most microwave applications are designed with an input impedance of 50 ohms, so matching the antenna to 50 ohms is our desire. The impedance of the microstrip patch antenna does not depend on the substrate dielectric constant, $\varepsilon_r$ or its height .The s-parameter graph and voltage standing wave ratio graph is unleashed for the impedance matching performance of the antenna.

B Return Loss

Return loss is an important parameter when testing an antenna. It is related to impedance matching and the maximum transfer of power theory. It is also a measure of the effectiveness of an antenna to deliver power from source to antenna. The return loss (RL) is defined by the ratio of the incident power of the antenna $P_i$ to the power reflected back from the antenna of the source $P_{\text{ref}}$ ; the mathematical expression is:

$$RL = 10 \log \frac{P_i}{P_{\text{ref}}} \quad \text{(dB)}$$

(2)

For good power transfer, the ratio $P_i/P_{\text{ref}}$ is high. Another definition of return loss we can get from this equation is the difference in dB between the power sent towards the antenna and the power reflected from it. It is always positive when the antenna is passive and negative when it is active. The return loss

is obtained from the s-parameter graph as shown in figure 2. The designed antenna resonates at 2.17 GHz, 2.3 GHz, 2.77 GHz and 2.87 GHz. The return loss at these frequencies are 39dB, 36dB, 18dB and 26dB which indicates that the designed antenna provides better impedance matching between the antenna and transmission line.
The Radiation pattern is a 3D graph showing the distribution of field strength or power strength of EM wave at all points which are at equal distance from the antenna. It defines the variation of the power radiated by an antenna as a function of the direction away from the antenna. The 3-D radiation pattern of the microstrip array antenna is given in Fig 3. The polar plot for the antenna is shown in figure 4.

**D Voltage Standing Wave Ratio (VSWR)**

The VSWR plot for coaxial feed antenna is shown in Figure 5. Ideally, VSWR must lie in the range of 1-2 which has been achieved for all the frequency range from 2-3 GHz.

**IV. CONCLUSION**

In this paper, new E-shaped design for microstrip arrays have been developed. The designed antenna achieves good impedance matching with an S11 of -39 dB at resonance frequency of 2.17 GHz and achieves wide-bandwidth. It also achieves desired VSWR for all the design frequency. Since, the microstrip lines connecting the patches couples electromagnetically energy to the microstrip array antenna, the feeding avoids vias or pins to connect the array and feedlines. This is achieved by the use of integrated feed design, which is suitable for large array antenna design. With these features, the proposed antenna can be very much suitable for wireless applications.

**REFERENCES**


INVESTIGATION OF RAMAN FIBER AMPLIFIER IN C AND L BAND AND IT'S COMPARISON WITH ERBIUM DOPED FIBER AMPLIFIER

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Abstract- Raman fiber amplifier (RFA) is constructed with 1480 nm pump source based on Raman shift amplification mechanism in the C and L band. EDFA is suitable to operate at the conventional (C) band from about 1530 to 1565 nm. EDFA by itself has a very low gain at the L band and also has larger noise figure than C band EDFA[1]. In this paper, we have investigated that in L band it is better to adopt RFA rather than L band EDFA.

Keywords- EDFA; Raman fiber amplifier; Gain; Noise Figure; Pump Power.

I. INTRODUCTION

If an amplifier is required for a CWDM based system considerable bandwidth is needed. Coarse Wavelength Division Multiplexing (CWDM) is an attractive solution for medium capacity communication systems. Conventional Erbium Doped Fiber Amplifiers (EDFAs) have a bandwidth smaller than 40 nm, while a Raman amplifier can provide gain over more than 100 nm, therefore EDFAs are not suitable for CWDM systems [2].

The distributed Raman amplifier is nothing but an arrangement of long transmission fiber and few high-power light sources. To extend the optical bandwidth and increase the number of WDM channels, L-band optical amplifiers are used [3].

A typical L-band EDFA has a very low-gain and has larger noise figure than C-band EDFA. The disadvantage of L band EDFA can be overcome using L band RFA.

In this paper, the study of comparison between L band EDFA and L band RFA for gain and noise figure has been carried out. From the mathematical analysis, it can be found that the optimized gain for C band EDFA is $G=44.4\,\text{dB}$, for the pump power $P_p=40\,\text{mW}$ and the noise figure is below $6.875\,\text{dB}$ while for L-Band EDFA for the same $40\,\text{mW}$ pump power the noise figure is greater than $6.875\,\text{dB}$ [3]. EDFA by itself has a very low-gain at the L-band, most realizations of L-band EDFA implement a long length of erbium-doped fiber (EDF) to pump up its gain [1].

In this paper we find the optimized gain is in the range of 9.8 dB - 10.6 dB, and the noise figure is below -2.9 dB for the pump power $P_p=800\,\text{mW}(29\,\text{dBm})$ for the L band RFA. Though EDFA gain is more in L-Band, the Raman amplifier gain is stable in L-Band and also RFA gives high bandwidth.

II. FUNDAMENTALS OF EDFA AND RFA

A. Erbium Doped Fiber Amplifier

EDFA is an optical or infrared (IR) repeater that amplifies a modulated laser beam directly, without any opto-electronic and electro-optical conversion. It works on the principle of stimulated emission and its active medium is a piece of silica fiber heavily doped with ions of rare-earth element erbium, typically in levels.

When the signal-carrying laser beams pass through this short length fiber, external energy is applied usually at IR wavelengths. This is called pumping, it excites the atoms in the erbium-doped section of optical fiber, increasing the intensity of the laser beams [3].

B. Raman Fiber Amplifier

A phenomenon observed in the scattering of light as light passes through a transparent medium; the light undergoes a change in frequency and a random alteration in phase due to a change in rotational or vibrational energy of the scattering molecules. This is known as Raman effect or Raman scattering. It is the inelastic scattering of a photon [4].

For high enough pump powers, the scattered light can grow rapidly with most of the pump energy converted into scattered light. This process is called stimulated Raman scattering(SRS), and it is the gain mechanism in Raman amplification [5]. Fibers used for Raman amplifiers are not doped with rare earth ions. In distributed or composite Raman amplifiers, any ordinary single-mode fiber could be used, and in practice an extended length of the optical transmission fiber is used to achieve amplification [6].

Distributed fiber Raman amplifiers are known to offer a large improvement of optical signal-to-noise ratio (OSNR) with respect to usual erbium doped fiber amplifiers [7].
### III. MODELING OF RAMAN AMPLIFICATION

In the case of continuous-wave (CW) and quasi-CW conditions, the nonlinear interaction between the pump and Stokes waves of SRS, is governed by the following set of two coupled equations for the forward-pumping case [5] [8]:

\[
\frac{dP_S}{dz} = g_R P_p P_S - \alpha_s P_S \quad (1)
\]

\[
\frac{dP_p}{dz} = \alpha_p \frac{\omega_p}{\omega_s} g_R P_p P_S - \alpha_p P_p \quad (2)
\]

where \(P_s, P_p, \omega_s, \omega_p\) are the power and frequency of signal and pump respectively and \(g_R\) is the Raman Gain coefficient. \(\alpha_s\) and \(\alpha_p\) account for fiber losses at the Stokes and pump wavelengths, respectively. In many practical situations, pump power is so large compared with the signal power that pump depletion can be neglected by setting \(g_R = 0\) in (2), which is then easily solved [5]:

\[
P_p(z) = P_0 \exp(-\alpha_p z) \quad (3)
\]

where \(P_0\) is the input pump power at \(z = 0\), we substitute this solution in (1), we get,

\[
\frac{dP_s}{dz} = g_R P_0 \exp(-\alpha_p z) P_S - \alpha_s P_S \quad (4)
\]

This equation can be easily integrated to obtain,

\[
P_s(L) = P_s(0) \exp(-\alpha_p L) \quad (5)
\]

where \(G(L)\) is the net signal gain, \(L\) is the amplifier length, and \(L_{eff}\) is an effective length defined as [5],

\[
L_{eff} = \frac{1 - \exp(-\alpha_p L)}{\alpha_p} \quad (6)
\]

Thus gain of RFA is simply defined by ratio,

\[
G_R = \frac{P_s(L)}{P_s(0)} \quad (7)
\]

From (5),

\[
P_s(L) = G(L) P_s(0) \quad (8)
\]

where \(G_R\) represents the total amplifier gain distributed over a length \(L_{eff}\) [5]. One of the most important parameters for Raman amplification in any applications is the Raman effective gain coefficient, which is defined as,

\[
\frac{g_R}{A_{eff}} = \frac{g_R}{L_{eff}} \quad (9)
\]

The effective Raman gain coefficient depends not only on the Raman gain coefficients itself but also on the effective area of the fiber \((A_{eff})\) [8]. The Raman gain coefficient is defined as in [15]:

\[
g_R = \frac{8\pi n^2 \alpha_p S}{\hbar \omega_s n^2 (N_0 + 1) D_{02}} \quad (10)
\]

Where \(N_0\) is the Bose occupation factor, \(n\) is the refractive index, \(S\) is the scattering efficiency, and \(\hbar\) is the reduced Planck’s constant.

**Do** is the FWHM of the spontaneous lineshape. Thus gain of RFA is given by [8] [13],

\[
G = \exp\left(\frac{g_R P_0 L_{eff}}{A_{eff}}\right) \quad (11)
\]

A noise figure in the optical domain is defined as the optical signal to noise ratio (OSNR) at the input of an optical amplifier (OA) divided by the OSNR at the OA output [9]. Thus, the noise figure may be simply estimated by measuring the Raman gain \(G_R\) and the amplified spontaneous emission power \(P_{ASE}\). Thus the NF is expressed as:

\[
NF = \frac{1}{G_R} \frac{2 P_{ASE}}{G_R \nu B_0} \quad (12)
\]

where \(\nu B_0\) is the shot noise contribution must be taken into account as an input reference and heuristically added at the OA output. Spontaneous Raman scattering adds to the amplified signal and appears as a noise because of random phases associated with all spontaneously generated photons. However, when the loss rates at the pump, \(\alpha_p\) and signal, \(\alpha_s\) are equal (\(\alpha = \alpha_p = \alpha_s\)), the ASE noise power will be evaluated analytically as [5] [10]:

\[
P_{ASE} = \nu B_0 \eta_T \left[ G_R \frac{1}{G_R} \left( \exp(\alpha L) - \frac{1}{G_R} \right) \right] \quad (13)
\]

where \(\eta_T\) is thermal equilibrium photon number and \(B_0\) is the bandwidth of the optical filter. \(h\) is the photon energy.

### IV. EDFA IN L BAND

The gain of EDFA is given as [3]:

\[
G_{EDFA} = \frac{\sigma n_{T} C P}{2 \nu B + \nu C + \nu B_{p}} \quad (14)
\]

Where \(\sigma, c, \nu, n, p, W_p\) are the cross section for induced emission, velocity of light, reciprocal of lifetime of charge carrier, total population density of Er ions, photon density and pump rate of particles respectively. Noise figure (NF) of EDFA is defined as in[11]:

\[
NF = \frac{P_{ASE}}{h v \Delta v G} + \frac{1}{G} \quad (15)
\]

The noise figure (NF) can be very simply written in terms of amplified spontaneous emission power (\(P_{ASE}\)) exiting the fiber in a bandwidth \(\Delta v\). Since the noise power is given by[11]:

\[
P_{ASE} = 2 n_{sp} \hbar v h v . (G - 1) \quad (16)
\]

Where \(G\) is the EDFA gain, \(h\) is the Planck’s constant, \(v\) is the frequency of light and \(n_{sp}\) is the inversion factor.

Fig. 1 and Fig. 2 shows the gain and NF Vs. wavelength of EDFA for different values of pump powers. We find the gain at 1575nm is 44.5dB and the NF is 6.875dB, for the pump power 40mW and
Investigation of Raman Fiber Amplifier in C and L Band and its Comparison with Erbium Doped Fiber Amplifier

the signal power 1µW. Beyond this wavelength or in the case of L band the gain starts to decrease and the NF starts to increase, due to low gain efficiency in L band [12]. Relative disadvantages of L band EDFA includes a lower efficiency, leading to high pump power requirements, and a low gain per unit length, leading to undesirably long fibers.

V. RFA IN C AND L BAND

By using the same pump configuration as that of EDFA i.e. co- or forward pumping we find the higher gain and low noise. For RFA no special doping is required, amplification takes place throughout the length of transmission fiber hence also know as distributed amplifier. The limitations of

L band EDFA is overcome by using Raman amplification. In L band RFA, Raman amplification requires very long fibers in the order of several kilometers and pump lasers with high optical power. Based on the analytical equations of Raman amplification using MATLAB's results of the gain flattened L-band RFA, we have investigated the effects of the Raman gain coefficient on the relative gain flatness and the effective gain bandwidth (Fig.3 and Fig.5).

C. Raman In C Band

Fig. 3 and Fig. 4 shows the gain and NF Vs. wavelength of RFA for different values of pump powers (Pp=500, 600, 700 and 800 mW). We find the gain at 1550nm is 9.8 dB and the NF is -2 dB, for the pump power 800mW. RFA has flat gain even in C-band, but since EDFA has low power requirements in C-Band, EDFA is preferred over RFA.

D. Raman In L Band

We find the optimum gain at 1600nm is 10.4dB (Fig. 5) and the NF is -2.98 dB (Fig. 6), for the pump power 800mW.

Figure 1. L-band EDFA Gain versus Wavelength

Figure 2. L-band EDFA Noise Figure versus Wavelength

Figure 3. C and L-band RFA Gain versus Wavelength

Figure 4. C and L-band RFA Noise Figure versus Wavelength

Figure 5. L-band RFA Gain versus Wavelength
Typically, the pumps will be located at wavelengths approximately 100 nm below the signal wavelengths to be amplified, i.e. below 1580 nm for amplification in the L-band [14]. Hence the pump wavelength used for Raman amplification is 1480 nm, this corresponds to a flattened or stable gain in the L-band between 1565 and 1625 nm.

As per the analysis the average gain in RFA is at 10.5 dB with a flatness of +/- 0.5 dB is achieved over the bandwidth of 100 nm by using 1480 nm pump laser.

When Raman pumping is used, small $A_{\text{eff}}$ fibers increase the Raman efficiency and therefore give higher gains for set pump powers [8] [13]. The Raman gain also depends on the fiber length and the attenuation at the pump wavelength through the effective length $L_{\text{eff}}$ (6). The maximum gain is obtained when $L_{\text{eff}} = 1/\alpha$. From (8) it is also clear that by increasing the effective length of the fiber the Raman gain will also be increased.

VI. CONCLUSION

In L-band RFA is better than EDFA to use for telecommunication because of improved gain efficiency and increased length of fiber. Raman gain, $G_R$ using the Raman gain efficiencies, permits to cancel out any inaccuracy related to spectrally dependent attenuation of components, connectors, splices and fiber. Hence compared to L-band EDFA, for the same pump laser, 1480 nm, and maximum pump power (P$_p$= 800 mW), RFA has optimized gain of 10.8 dB and NF below -2.9 dB at the signal wavelength 1600 nm and the deviation is +/- 0.5 dB, i.e. a flattened gain in L-Band.

VII. ACKNOWLEDGMENT

The authors wish to thank Prof. Shobha Krishnan whose stimulating suggestions helped us in all the time of research.

REFERENCES


LIDAR ENABLED WIND TURBINE LOAD MITIGATION USING FX-RSL FEED-FORWARD ALGORITHM

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Abstract - A wind turbine is a device that converts kinetic energy from the wind, into mechanical energy; energy known as wind energy or wind power. The turbines are used for an increasingly important source of wind power-produced commercial electricity. The utilization of wind turbines can be a great way to capture the energy of the wind in a bid to convert this into useable electricity. Harnessing the winds energy with a wind turbine can provide a source of clean and renewable electricity for large or small industries. Wind energy is undoubtedly one of the cleanest forms of producing power from a renewable source. There is no pollution, there is no burning of fossil fuels, and unless something very drastic happens, you don’t run out of wind. But it’s not like we can erect a wind turbine anywhere and it will start generating power. There are lots of factors that can make an impact on the amount of energy we can generate out of wind, such as wind speed, height or altitude & the rotor size. Recent researches have been done to regulate the rotor speed & reduction of the component loads with the help of feed forward controllers. Wind speeds measured by light detection & ranging system (LIDAR) will give information of wind variations at various distances and so are used along with FX-RLS feed forward controllers for better tracking & better load reduction when the wind turbine is running at beyond its operating point.

I. INTRODUCTION:

According to the recent researches, it has been proved that efforts could be made to improvise the wind turbines efficiencies by providing the feed forward control systems which will propagate the wind speeds operating range. The utilization of wind turbines can be a great way to capture the energy of the wind in a bid to convert this into useable electricity which provides a source of clean and renewable electricity for all industries.

There are lots of factors that can make an impact on the amount of energy we can generate out of wind, such as wind speed, height or altitude & the rotor size. Wind turbines operate at larger loads and therefore are subjected to fatigue. Because of the variation of the wind flow, the turbines may go beyond the operating speeds at times. These can be avoided by introducing the controllers for the wind turbines.

Since the wind turbines are non-linear because of the variations of wind speeds, non-linear adaptive controllers are used. For example, EEC (extreme event control) algorithm is used to prevent the rotor from operating beyond the allocated speed range. Recent researches have been done to regulate the rotor speed & reduction of the component loads with the help of feed forward controllers. Two types of controllers are taken into account –

1. Adaptive feed forward controller based on a filtered-x recursive least square algorithm (FX-RLS).
2. Non-adaptive feed forward controller based on a zero phase error tracking control (ZPETC).

Results show that, combining the PI feedback control with ZPETC feed forward control improves the blade loads but affects the tower loads which may lead to fatigue. Whereas FX-RLS gives better output & better reduction of blade & tower bending moments with only a smaller energy loss.

II. BLOCK DIAGRAM:

Because of the variation of the wind flow, the turbines may go beyond the operating speeds at times. As the Wind turbines operate at larger loads and therefore they are subjected to fatigue. These can be avoided by introducing the controllers for the wind turbines. Wind speeds measured by light detection & ranging system (LIDAR) and will give information of wind variations at various distances. If wind speed exceeds the rated wind speed then control algorithm is use to regulate the rotor speed of wind turbine and also to reduce the component load. Hence this LIDAR information about wind speed is used along with feed forward controllers for better tracking & better load reduction when the wind turbine is running at beyond its operating point. Since the wind turbines are non-linear because of the variations of wind speeds, non-linear adaptive controllers are used. LIDAR information with feed forward controller and feedback controller is used for adjusting or for controlling the pitch angle of blade. Pitch control means that the blades can pivot upon their own
longitudinal axis. And this pitch angle used for controlling turbine rotor speed.

Software used for simulation is MATLAB. Initially by assuming some turbine specifications like Hub height, Rotor diameter, maximum pitch rate, rated rotor speed, rated power etc. wind is modelled. For wind modelling, as we know wind is non-linear in nature so have to model it non-linearly by adding Gaussian noise to it. And by taking delay of 1 sec between wind measured at LIDAR and wind measured at turbine hub, wind is modelled. And both the winds are show on the same graph by different colours.

In the above graph two graphs are denoted by different colours in which red graph indicated the wind measured by LIDAR and blue graph indicate the wind measured at turbine hub. And there is tiny gap is present between this two graphs which is nothing but a delay of 1 sec which we assumed initially while modelling the wind.

REFERENCES:


A COMPARATIVE ANALYSIS OF AUTOMATIC GENERATION CONTROL FOR TWO AREA CONTROL SYSTEM USING INTEGRAL, PI AND PID CONTROLLER FOR OPTIMIZATION OF DYNAMIC ERROR

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Abstract- To synchronize between generation side and load side and to avoid steady state error in a power system, controllers viz. proportional, integral, and derivative and the combination of them are employed. This paper presents a comparative study using different types of controllers and shows the best suitable controller that can minimize the steady-state error at minimal time span. Simulation results demonstrate that the system responses, in case of a two-area load frequency control, are more prone to minimal system recovery time when Proportional-Integral-Derivative controller is used.

Keywords- power generation, industrial automatic controllers, load frequency control.

I. INTRODUCTION

Industrial applications nowadays have become increasingly complex and demanding, especially in the field of power systems, where generation of power is as important as control and regulation of power in order to match up with the load requirements. In the present trend the two main components of power viz. Active power and Reactive power are essential for maintenance of system stability and system voltage [4]. Take the case of any steam power plant where to raise the active power steam is fed to the turbine which converts heat energy to mechanical energy and fetched to the generator which converts mechanical energy to electrical energy i.e. power output. So more the steam input, more will be the frequency, more the rotor speed of any turbo generator system, with which the generator angle δ will rise and accordingly active power will be increased and will be fetched to the load [3]. This mechanism can be separately controlled and adjusted through a closed loop control system known as load frequency control but used in conjunction with another separate closed loop control system for adjustment of reactive power for which the output voltage of generator is compared with a reference voltage, and the difference is fetched to a controller which senses whether there is any requirement for adjustment of generator field current so as to adjust the generator voltage and reactive power generation is adjusted which is essential for maintenance of system voltage [9]. So as evident manual control is not feasible for this kind of operation as the load demand also keeps on varying on the system which necessitates for automatic control. Therefore this entire process known as automatic generation and voltage control (AGC) is feasible in large interconnected systems where for each generator a separate AGC regulation equipment is installed. In electric power generation, system disturbances caused by load fluctuations result in changes of the desired frequency value, which being a control error give rise to transients in the system and eventually there can be a complete breakdown of the system but if sustained can burn out lines, motors, generators [3]. With the help of any suitable controller this steady state error can be fetched back to its input which can then control the speed changer setting and thereby causes an adjustment in the power generation (increases or decreases) to match up with load and thereby reduce this change in frequency to zero [4].

II. TWO AREA LOAD FREQUENCY CONTROL

Power system as nonlinear system is linearized around an operating point. The system investigated in this paper, consisting of two control area connected by a tie-line is shown in Fig. 1. Each control area is containing non-reheat turbine type thermal unit [1]. Each area is assumed to have only one equivalent generator and is equipped with governor- turbine system. The load flow controller shown in Fig. 1 is based upon tie-line bias control, where each area tends to reduce the Area Control Error (ACE) to zero. The ACE given in Eq.(1) & Eq.(2) for each area consists of a linear combination of frequency and tie-line power deviation [2].

\[ ACE_1(s) = \Delta P_{tie,1}(s) + b_1 \Delta F_1(s) \]

Eq.(1) [3]
A Comparative Analysis of Automatic Generation Control For Two Area Control System Using Integral, PI and PID Controller for Optimization of Dynamic Error

**III. CIRCUIT DESCRIPTION**

Here two load frequency control areas are interconnected by means of a tie-line. Aim is to reduce area control error to zero in minimum time as possible. The Area control error here comprises of $(\Delta F + \Delta P_{tie})$ for both the areas. Load demand change is kept different for both the areas in order to see the incremental change in tie line power and that this error is also reduced to zero in minimum time. In the conventional methods when employed it is seen that the area control errors have been reduced to zero but taking a longer time of about 20 secs. (with respect to reference [5] as evident from Fig. 3, 4, 5, 6). But if for such a long period this kind of transient lasts then it causes a long power shutdown as more the time it stays more is the duration of transients. Here in control area-1 $\Delta P_{d1}$ (incremental load demand change) = 2 p.u. in step time of 1 second and in area-2 $\Delta P_{d2}$ = 4 p.u. in step time of 1 second. Change in frequency will now be obtained in both the areas as well as incremental change in tie line power $\Delta P_{tie,1}$ and $\Delta P_{tie,2}$ according to equations [Eq.3],[Eq.4],[Eq.5],[Eq.6]. Both the control errors are fetched to the controllers. Both the ACEs are integrated for step disturbances and accordingly through proportional, integral and derivative gains of the PID controllers the errors $\Delta F_1$ and $\Delta F_2$ are reduced to zero as evident from graphs [Fig.2]-[Fig.13]. And similarly $\Delta P_{tie,1}$ & $\Delta P_{tie,2}$ are reduced to zero but only when errors $\Delta F_1$, $\Delta F_2$ become equal. When change in load demand takes place, there exists a difference between change in generation and change in demand. When, these two changes are made equal with help of controller then change in frequency gradually reduces to zero. The effectiveness of a controller depends upon how this dynamic error becomes zero [6]. Here, incremental changes in tie-line powers takes place due to unequal load disturbances for both the areas when considered in practical situation. These errors are also reduce to zero with the help of suitable controllers and which can mainly happen when the difference between the incremental changes in frequencies of both the areas is reduced to zero.
IV. OBSERVATIONS

Fig. 2. $\Delta F_1$ reduced to zero by Integral controller

Fig. 3. $\Delta F_2$ reduced to zero by Integral controller

Fig. 4. $\Delta P_{tie,1}$ reduced to zero by Integral controller

Fig. 5. $\Delta P_{tie,2}$ reduced to zero by Integral controller

Fig. 6. $\Delta F_1$ reduced to zero by PI controller

Fig. 7. $\Delta F_2$ reduced to zero by PI controller

Fig. 8. $\Delta P_{tie,1}$ reduced to zero by PI controller

Fig. 9. $\Delta P_{tie,2}$ reduced to zero by PI controller
From this chart it is seen that which controller takes how much time to reach steady state condition. It is obvious that PID is much faster than PI or I controller.
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