Reduction Of Power Line Humming And High Frequency Noise From Electrocardiogram Signals

Mohammed Nabil Abdalazim Mursi, Dr. Mohamed H. M. Nerma

ABSTRACT: With the latest advancements in electronics, several techniques are used for removal of unwanted entities from signals especially that are implied in the most complicated applications. The removal of power line interference from most sensitive medical monitoring equipments can also be achieved by implementing various useful techniques. The power line interference (50/60 Hz) is the main source of noise in most of bio-electric signals. The thesis report presents the removal of power line interference and other single frequency tones from ECG signal using the advanced adaptive filtering technique with least mean square (LMS) algorithm. The thesis is based on digital signal processing (DSP) techniques with MATLAB package. The MATLAB package will be used in the thesis work which is a powerful tool for the interactive design in most of the scientific applications and complex engineering calculations. In addition so as to achieve the goal of thesis, the removal of harmonics (hum) and high frequency noise from ECG signal by using general notch rejection filters is investigated and implemented.

Index Terms: Power line interference, adaptive filtering, LMS, DSP, hum, high frequency noise, ECG, general notch rejection filters.

I.INTRODUCTION

The Electrocardiograph (ECG) signal is an electrical signal generated by the heart's beats and can be used to examine some of the functions of the heart. The ECG signal can be distorted with noise of 50/60 Hz and by some other sources. The noise from electric power system is a major source of noise during the recording or monitoring of ECG [1]. Different noises have different frequencies, the noise with low frequency is the main problem with ECG signal and sometimes high frequency noises can also cause interference in ECG i.e. mobile phone. If the physical or mathematical variable changes rapidly then it can be high frequency and if it changes slowly then it would be low frequency. If the variable does not change at all then it is said that it has zero frequency. The frequency is measured in cycle/second or in "Hertz". For example the electric power used in daily life in The United States is 60 Hz and 50 Hz in the rest of world. Most of the electronic devices such as ECG, transmitter, receiver, computer etc get power from power line. The 50 Hz alternative current (AC) is reduced in voltage, rectified and then filtered to obtain low voltage direct current (DC). This is used to give power to those electronic devices [2]. Numbers of adaptive filter solution had been proposed for noise cancellation in ECG. The adaptive filter remove or reduces the mean squared error between primary input (ECG signal) and the reference input (noise with ECG signal) [3]. While recording ECG signal, the critical problem is unwanted noise from power line interference. There are different noises which affect ECG signal but 50/60 Hz interference from power line distribution is most critical and also 1 Hz power line interference due to patient's movement.

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Various methods were developed for the removal of power line interference from last two decade. The suitable and prime methods were based on ECG filtering. There have been different filtering solutions, which were introduced for the removal of power line AC interference. The crucial problem of power line interference was found in ECG signal. The reduction of power line interference of 50/60 Hz from ECG signal is the main purpose of the thesis. Different filtering solutions have been studied to find out the best solution for the reduction of power line interference from ECG signal. Digital filter has been selected to overcome this problem; there are few filtering solutions which were examined before to manipulate the power line interference from signal which can be divided into following categories [4].

- Low Pass Filters
- General Notch-Rejection Filters
- Adaptive Filters
- Global filters

In the thesis two filtering solutions has been chosen for the removal of power line interference, its respective harmonics and high frequency noise from Original ECG signal. The removal of power line interference (50Hz) from ECG signal can be reduced by adaptive filtering while the harmonics and high frequency noise can be reduced by implementing general notch rejection filters and low pass filters respectively.

A. Heart Mechanism and Purpose of ECG Diagnosis

The heart is a muscular organ; it pumps the blood throughout the body and collects blood circulating back from the body [5]. Electrical impulses are the main source of generation of regular normal heartbeat. The heart muscle must be activated electrically before the beginning of its mechanical function. When the electrical abnormalities of the heart occur then the heart cannot pump blood properly and supply enough to the body and brain. This can cause unconsciousness within seconds and death within minutes [6]. An ECG recording is important for clinical diagnosis and treatment; it is a graphical recording of electrical impulses generated by heart. The ECG is needed to be done when chest pain occurred such as heart attack, shortness of breath, faster heartbeats, irregular heartbeats, high blood pressure and high cholesterol in order to check the heart's electrical activity [5]. An image of the human heart and an example ECG signal is shown in Figure 1:

Figure 1: *Heart Electricity [7]*

B. Introduction to ECG Signal

ECG is the graphical recording of the electrical activity of the heart, it is by far the most easily recognized biological signal, it is also the most commonly used for clinical diagnosis and more important is the fact that the ECG wave shape is altered by cardiovascular diseases and abnormalities, each portion of the ECG waveform carries information that is relevant to the clinician in arriving at a proper diagnosis. The Electrocardiograph signal taken from a patient was corrupted by an external noise, so that we need a proper way to get a noise free ECG signal, simple ECG wave form is shown in Figure 2, an ECG signal is a combination of P,T,U wave, and a QRS complex [8], the complete wave form is called an electrocardiogram with labels P, Q, R, S, and T indicating its distinctive features. The P wave arises from the depolarization of the atrium. The QRS complex arises from depolarization of the ventricles. The T wave arises from re-polarization of the ventricle muscle.

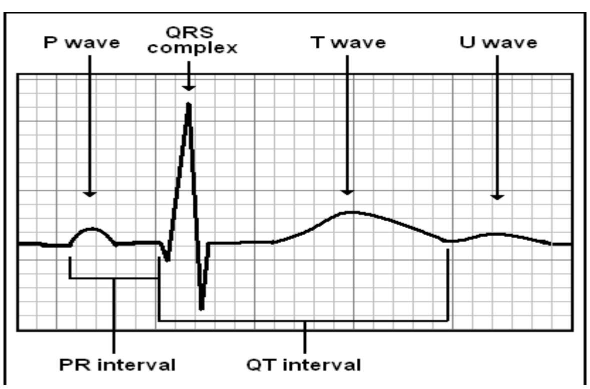


Figure 2: *ECG Signal*

II.NOISE IN ECG

Adaptive filters are used to eliminate the power line interference (50 Hz) and they are proposed to obtain the impulse response of the normal QRS complex [3]. In Figure 2, an uncorrupted ECG signal shows an original signal graph for ECG signal which demonstrate the diagnosis of heart activities for heart patient. Consequently, it is analysis that how to remove the power line interference of 50/60 Hz which is a problem for biomedical signal measurement. Electromagnetic interference (EMI) from 50/60 Hz power line noise is present in cable holding ECG signal [9]. Several solutions for the reduction of power line interference have been expressed. The main source of interference is AC power line interference. The interference is caused by magnetic fields as well as by the electric fields. When special signal recording techniques are applied, which minimize the interference therefore some AC noise remains as a consequence of unbalanced input impedances. Also we have harmonics or Hum that is a component frequency of the signal that is an integer multiple of the fundamental frequency, i.e. if the fundamental frequency is f, the harmonics have frequencies 2f, 3f, 4f, . . . etc. The harmonics have the property that they are all periodic at the fundamental frequency; therefore the sum of harmonics is also periodic at that frequency. Harmonic frequencies are equally spaced by the width of the fundamental frequency and can be found by repeatedly adding that frequency. For example, if the fundamental frequency (first harmonic) is 50 Hz, the frequencies of the next harmonics are: 100 Hz (2nd harmonic), 150 Hz (3rd harmonic), 200 Hz (4th harmonic) etc. Figure 3 describes in more detail [10].

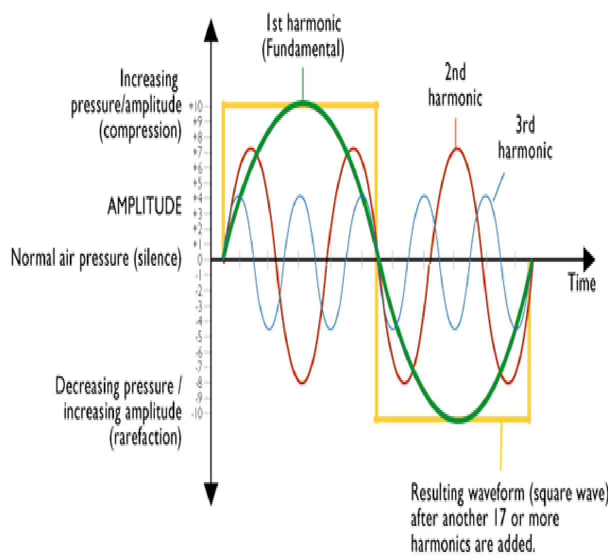


Figure 3: *Harmonic Wave [10]*

III.FILTERING TECHNIQUES

A. Finite Impulse Response (FIR)

A finite Impulse Response (FIR) filter are type of digital filters [11] and consists of weighting sequence (impulse response) among non-recursive digital filters which is finite in length. [12] FIR filters are non recursive digital filters [11] has been selected for this thesis due to their good characteristics and can be used to implement in any sort of frequency response digitally. The series of multipliers, delays and adders are used for FIR filters' implementation for filter's output. The output of the non recursive digital filter is formed from the weighted linear combination of current input and previous value of the input [13]. Finite impulse response (FIR) filter structure [14] is presented in Figure 4, which describes the relationship between input and output sequences which also describe the basic structure and diagram of FIR filter having a length of N (where N is filter order) and the input samples are operated by the delays of results. All the delayed samples are multiplied by suitable coefficient as the hk is the coefficient value for multiplication for output at time n [14]. The selection of FIR filter is due to coefficient sensitivity, round off noise, stability and suitable for high speed applications [15]. FIR and IIR filters are two different classes of digital filters, these digitals filters can be implemented for different application. The selection of any type of digital filter is based on the practical implementation of required application. The FIR filter is mostly applied for adaptive filtering and the main choice of FIR filter was its stability and robustness.

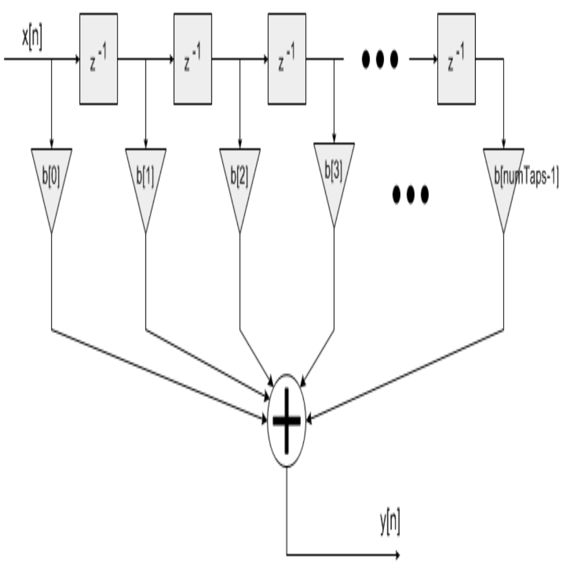


Figure 4: *Finite Impulse Response filter*

B. Least Mean Square (LMS) Algorithm

The LMS algorithm is extensively used in different application of adaptive filtering due to low computational complexity, stability and unbiased convergence [16]. In any signal's processes there can be error occurred in the required output. There must be suitable algorithm needed to manipulate this problem. The least mean square (LMS) algorithm is introduced to minimize the error between a given preferred signal and output of the linear filter by adjusting recursively the parameters of a linear filter [17]. The more suitable and basic algorithm for the adaptive filtering is LMS, which is also famous for the stability of the system [18]. LMS is the most important algorithms in whole family of algorithms, which has been developed for minimizing the error [17]. This algorithm is used for the better condition of input signal to attain the faster convergence [16] and can address a range of problem settings, computational restrictions and minimization criteria [17]. Every application needs its simple and easy solution, the LMS algorithm has been selected for this thesis due to its simplicity, robustness and ease of implementation. Simplicity and robustness are its major future and it is the widely used in adaptive filtering algorithm for different applications [19]. The different areas where LMS algorithm is used include adaptive signal processing, system identification and adaptive control [18]. Due to its simplicity and robustness, it has made the standard for the adaptive filtering and is much famous for the different application as compared to other linear adaptive algorithms [20]. Least mean square algorithm has lots of benefits in different applications; it has been productively used in many applications due to the following performance aspects [17].

> LMS has the ability to reject noisy data due to minute step size parameter μ.
> LMS demonstrate slowly time varying system.
> LMS algorithm does not get stuck at undesired local minima.
> LMS is computationally simple memory competent.

In general LMS adaptive filter removes noise or obtains a desired signal by adapting the filter coefficient with least-square algorithm based on given filter [21]. The performance of the LMS algorithm is very high and it is simple in implementation for the removal of low frequency noise. The suitable value for step size parameter μ can be selected according to the application's requirement. LMS is used for the simplification of gradient vector computation [16]. The overview of the structure and operation of the LMS algorithm can be discussed according to LMS algorithm's properties and its processes [20]. The main property of LMS algorithm is its convergence behaviour in a stationary environment [16]. LMS is a linear adaptive filtering algorithm and it consists of two basic processes:

- **Filtering Process**

Filtering process is used to calculate the output of linear filter and to generate an estimated error by comparing this output with desired response [20].

- **An Adaptive Process**

An adaptive process is used for the automatic adjustment of the filter's parameters in accordance with the estimated error [20].

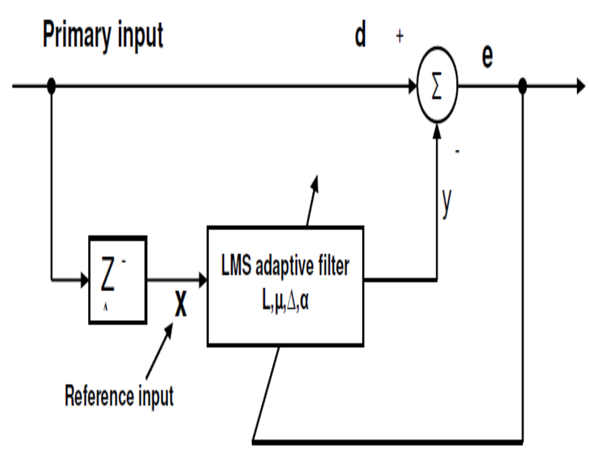


Figure 5: *LMS Adaptive Algorithm*

In the Figure 5, the overview of least mean square (LMS) algorithm is shown. The primary input has been taken, where 'X' is the reference input. The error signal occurs for the desired output, there LMS adaptive filter has employed to manipulate the error. The error signal manipulated by the adaptive algorithm is described in equation 1:

$$e_n = d_n - y_n \qquad (1)$$

The equation above shows the desired signal and the filter output, where d(n) is the desired signal and y(n) is the filter output. For the minimization of error signal the input vector x(n) and e(n) are employed. Here it needs to work according to the criterion that is supposed to minimize. The input vector is used to update the adaptive coefficients according to that criterion [22]. The criterion used here is the mean-square error (MSE) shown in equation 2 below:

$$e = Ee_n^2 \qquad (2)$$

C. Notch Filter
It is well known or simplest filter to remove the power line interface notch filter it computes the Fourier transform of the signal delete undesired component and the inverse Fourier transform [23]. The notch filters were constructed in analogue form traditionally. However, analogue notch filters have several problems such as frequency response accuracy, difficult realization and nonadjustable notch frequencies. For these disadvantages, the digital notch filter is developed. If we classify the digital notch filter by the length of impulse response, then it can be put into two sections: (1) finite impulse response (FIR) (2) infinite impulse response (IIR). The digital FIR notch filter is always stable and it provides linear phase response. On the other hand, the digital IIR notch filter is potentially unstable and do not provide linear phase response. In general, IIR filter structures can be designed with a much lower order then their FIR counterparts for meeting equivalent magnitude specifications. So, a digital FIR notch filter need long filter length to reach the same requirement of the magnitude response. Because the signal delay is proportional to the filter length, it is often intolerable for many applications. The digital notch filter can also be classified according to the number of frequencies the filter can reject : (1) Fixed notch filters (2) Tuneable notch filters (3) Adaptive notch filters (ANFs). The digital notch filter can reject a specific annoying frequency and keep other broadband signals intact. This kind of notch filter is called the single notch filter which only diminishes a prescribed frequency. At times, more than one interfering frequency exists, so the multiple notch filter is required to get rid of more than one prescribed frequency. The simplest way to construct a multiple notch filter is to cascade single notch filters. Tuneable notch filters are similar to fixed notch filters that have a range of frequencies that they can be set to and then fixed at that frequency. If we encounter with signals which are variable frequency and depend on events over time, i.e. we don't know the notch frequencies in advance, then adaptive notch filters (ANFs) are utilized in this kind of situation. They can automatically adjust their frequency response depending upon circumstances [24]. A notch filter highly attenuates/ eliminates a particular frequency component from the input signal spectrum while leaving the amplitude of the other frequencies relatively unchanged. A notch filter is, thus, essentially a bandstop filter with a very narrow stopband and two passbands. The amplitude response, H1(w), of a typical notch filter (designated as NF1) is shown in Figure 6 and is characterized by the notch frequency, wd in radians and 3- dB rejection bandwidth, BW. For an ideal notch filter, BW should be zero, the passband magnitude should be unity (zero dB) and the attenuation at the notch frequency should be infinite [24].

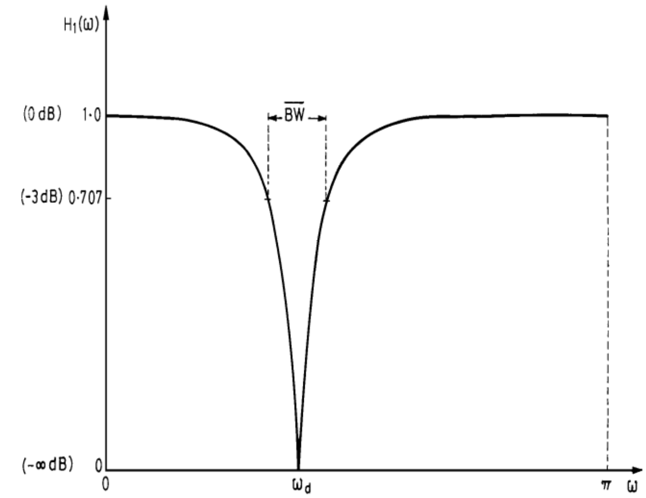


Figure 6: *The Amplitude Response H1(w) of Notch Filter: NF1.*

The magnitude response jH2(w)j is of the same type as that shown in Figure 7. We review methodologies for approximating notch filters of both the types [24].

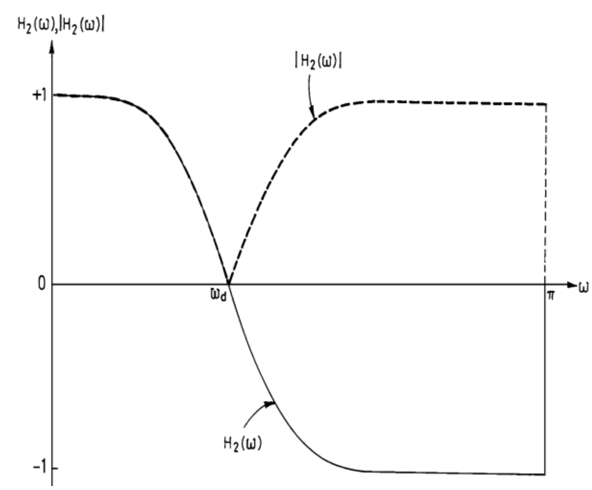


Figure 7: *The Response H2(w) and jH2(w)j of Notch Filter: NF2*

We may, alternatively, have the amplitude response, H2(w), of a notch filter (designated as NF2) as shown in Figure 7. H2(w) has 180 degrees phase shift beyond the notch frequency (wd) [24].

IV. Noise Reduction
Noise reduction is the process of removing noise from a signal. All recording devices, either analogue or digital, have traits which make them susceptible to noise. Noise can be random or white noise with no coherence, or coherent noise introduced by the device's mechanism or processing algorithms. In electronic recording devices, a

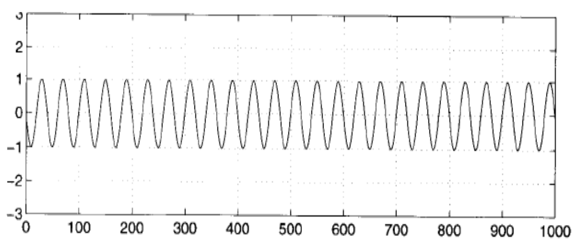
major form of noise is hiss caused by random electrons that, heavily influenced by heat, stray from their designated path. These stray electrons influence the voltage of the output signal and thus create detectable noise [25].

V. Noise Cancellation

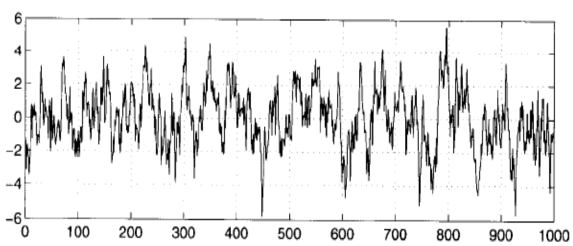
The noise cancellation is severe problem in signals, the process d_n in noise cancellation is estimated from a noise corrupted observation.

$$x_n = d_n + v1_n \tag{3}$$

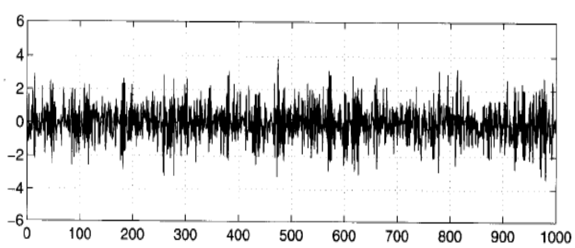
In the equation 3 there must be some information about d_n or $v1_n$ for the separation of signal from noise. The reference signal may be used to estimate the noise $v1_n$, and this estimate may then be subtracted from x_n to form an estimate of d_n [19]. In the Figure 8, (a) Represent the information signal, (b) Represent the noise signal, (c) Represent the mixed signal of information and noise signals, (d)Represent the original signal after filtration. The principle of noise cancellation is to obtain noise signal and subtract it from the corrupted signal. The required signal can be found after subtracting the corrupted signal from noise [26]. Figure 8 represents the information signal (a) and the noise signal (b), both the signals are mixed together and then mixed signal (c) is displayed. To remove the noise signal (b) from information signal (a), there must be some filtration method which is based on the application requirement. The original signal (d) represents the output signal after filtration.



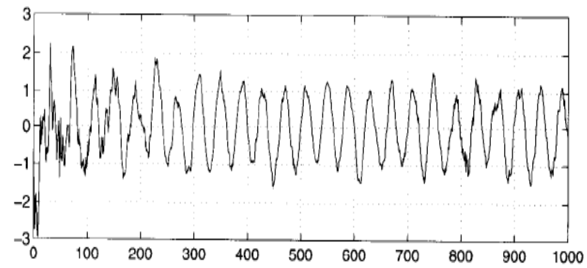
(a)



(b)



(c)



(d)

Figure 8: *Noise Cancellation Process, a) Information Signal, b) Noise Signal, c) Mixed Signal and d) Original Signal.*

VI. Adaptive Noise Cancellation

Adaptive noise cancellation can be considered as an outgrowth of the interference cancellation. One of the adaptive noise cancellation applications was to remove 50/60 Hz noise from ECG signal [27].

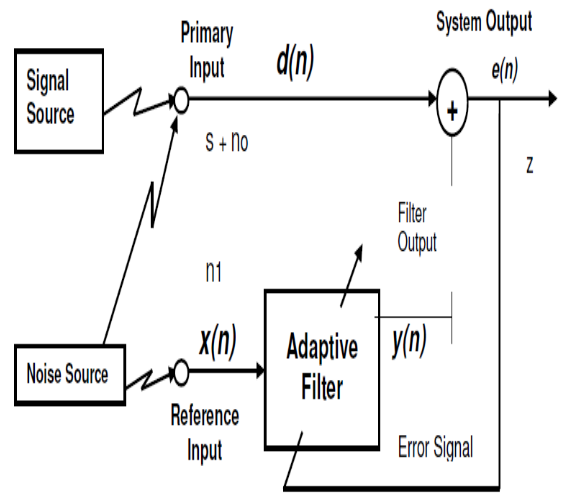


Figure 9: *Block Diagram of Adaptive Noise Canceller*

At the primary input signal's' is measured together with noise is assumed to be the sum of an information signal and sinusoidal interference. For monitoring of the noise 'n1' a reference input supplies a correlated version of the sinusoidal interference. The correlation of the noises 'no' and 'n1' is assumed to be high and have same origin so that influence of the useful signal's' is negligible at the reference input [28]. The filter uses the reference input to provide an estimate of sinusoidal interfering signal contained in the primary input. The adaptive filter forms as estimate of 'no' thus by subtracting the adaptive filter output from the primary input signal. So the information signal with noise is cancelled at the output by adaptive noise cancellation method [29]. There are two important characteristics of LMS algorithm i.e. canceller behaves as an adaptive notch filter, which is tuneable and the notch in the frequency response can be made very sharp by choosing the small value of the step size parameter μ [20]. The noise cancellation is required to remove unnecessary noise from the given signal. The term cancellation principle is used to detect the noise and subtract that noise from the corrupted signal. Its feasibility depends upon the availability of a noise signal originating signal. The concept of noise cancellation in its simplest form is described in Figure 9.

VII. Interference Cancellation by Adaptive Filtering

There are numbers of interferences in different applications in different fields; in the field of biomedical engineering interference cancellation is severe problem for biomedical signals such as ECG. In our case power line interference 50 Hz is core problem for ECG signal. Adaptive filtering is the best solution to reduce power line interference from biomedical signals. In Figure 10 the adaptive filtering for interference cancellation is shown. The unknown interference can be cancelled or reduced by this class of adaptive filtering from primary signal. The desired output by adaptive filtering is concerned to its primary signal [30].

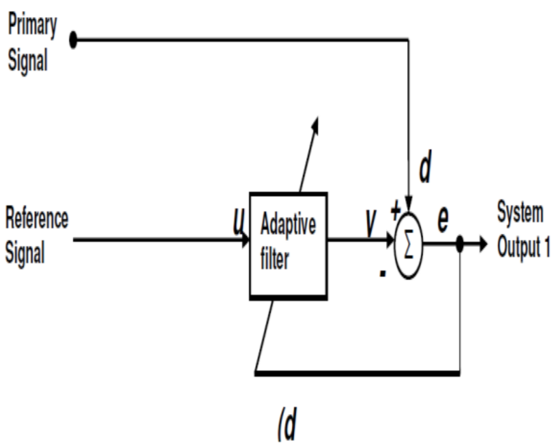


Figure 10: *Interference Cancellation by Adaptive Filtering*

According to Figure 10, adaptive interference cancellation method the reference signal is employed as an input to the adaptive filter. The sensor (s) which supplying the primary signal can derive a reference signal in such a manner that the information bearing signal becomes essentially undetectable or weak [30].

VIII. Simulation Design

The MATLAB environment provides an accurate simulation of the application in real world, the more details and parameters are defined the more accurate the simulation will be, thus providing strong results to back the conclusion. In this thesis the objective is to simulate an ECG signal getting corrupted by different causes and later filtered to provide a high quality noise reduced signal. First, in order to accomplish that (thesis objective) an ECG signal, that has an already defined function in the MATLAB, was generated. Second, a 50Hz noise signal was also generated. Thirdly, both the ECG original signal and the 50Hz noise signal were mixed together to provide the power line interference desired to corrupt the signal. Fourthly, an error signal has been calculated by the difference of original ECG and 50 Hz noise signal. Fifthly, the LMS filtering function was defined and applied on the mixed signal in order to retrieve the original ECG signal. The above steps were applied four times with different filter taps and step sizes to verify the best parameters that are to be used for the filtering process. Also part of the thesis objective was the reduction of high frequency noise and humming through low pass filter and general notch filter respectively. This was achieved through first the generation of the ECG original signal, second was the generation of Hum signal, third was the generation of high frequency noise that was assigned to be 500 Hz. The Fourth step was to mix the original ECG signal, the Hum and the high frequency noise all together resulting in the corrupt environment required for the simulation. The Fifth step was to apply the general notch filter to the mixed signal that resulted in an output signal containing the ECG signal mixed with the high frequency noise. The sixth and final step was to pass this signal through a low pass filter in order to retrieve the original ECG signal. Figure 11 explains the process of 50Hz noise reduction from ECG signal using LMS filter.

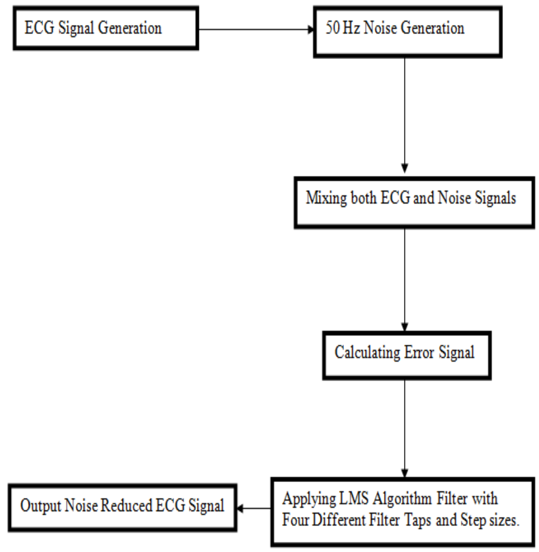


Figure 11: *Block Diagram of Power Line Interference Reduction from ECG Signal*

Figure 12 explains the process of harmonics and high frequency reduction from ECG signal using general notch rejection filter and low pass filter respectively.

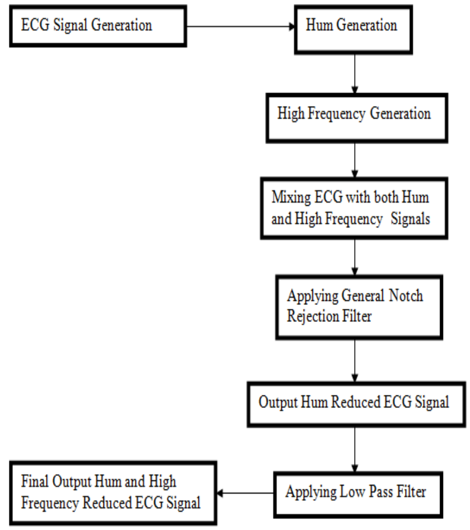


Figure 12: *Block Diagram of Hum and High Frequency Reduction from ECG Signal*

IX. SIMULATION RESULTS & DISCUSSION

A. Removal of Power Line Interference from ECG Signal by LMS Algorithm

The ECG signal has been taken and LMS adaptive filter algorithm has been developed. The ECG signal of 50 Hz is displayed in MATLAB environment as ECG Signal and then the noise of 50 Hz is generated and then mixed with the ECG Signal, which is displayed as Mixed signal. The adaptive filter is implemented by using LMS algorithm, FIR filter has been designed. The ECG Signal, 50 Hz Noise signal, Mixed signal, Error signal and Adaptive LMS filtered output signal have been displayed. The output is nearly same as the ECG input signal.

1. ECG Signal Simulation - Case 1

In Case 1 the Filter Tap=16 and the step size μ = 0.005, when applying these parameters in the MATLAB environment it results in the Figure 13.

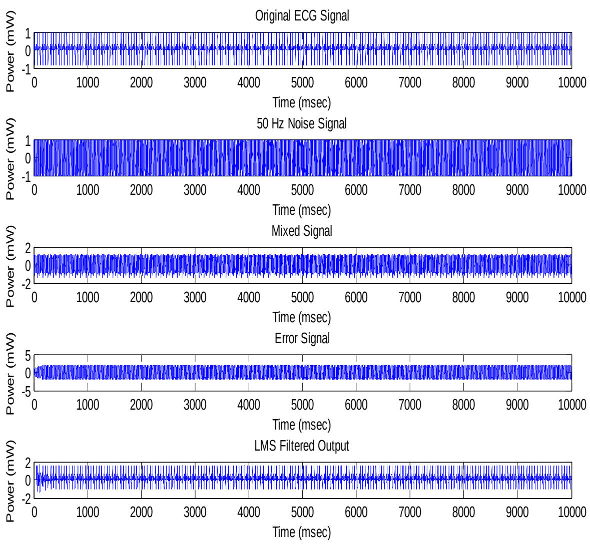


Figure 13: *ECG Signal Simulation - Case 1*

In the Figure 13 ECG Signal Simulation-Case 1, the value of the filter tap and μ has been taken 16 and 0.005 respectively to generate the graph. The figure shows five different plots, first, second, third, fourth and fifth plots are taken as to be ECG signal, 50 Hz noise signal, mixed signal, error signal and filtered output signal respectively. The ECG signal of 50 Hz and noise signal of 50 Hz (which is supposed to be power line interference) are generated and then mixed together in the third plot of the figure. The fourth plot of the figure shows the error signal which has been calculated by the difference of original ECG and 50 Hz noise signal. The fifth plot shows the LMS filtered output of the mixed signal which is nearly same to the input ECG signal. It is noticed that the system shows adaptation after subtracting the noise of 50 Hz from ECG signal.

2. ECG Signal Simulation - Case 2

In Case 2 the Filter Tap=16 and the step size μ = 0.009, when applying these parameters in the MATLAB environment it results in the Figure 14.

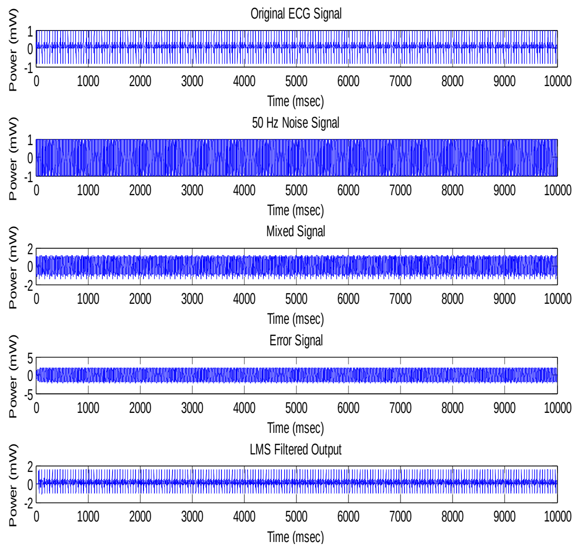


Figure 14: *ECG Signal Simulation - Case 2*

In the Figure 14 ECG Signal Simulation-Case 2, the value of filter tap and μ are taken as 16 and 0.009, which shows different results as compared to the ECG Signal Simulation-1. The figure shows five different plots, first, second, third, fourth and fifth plots are taken as to be ECG signal, 50 Hz noise signal, mixed signal, error signal and filtered output signal respectively. The ECG signal of 50 Hz and noise signal of 50 Hz (which is supposed to be power line interference) are generated and then mixed together in the third plot of the figure. The fourth plot of the figure shows the error signal which has been calculated by the difference of original ECG and 50 Hz noise signal. The fifth plot shows the LMS filtered output of the mixed signal which is nearly same to the input ECG signal. It is concluded that the rate of convergence is changed by changing the value of μ. The slower the convergence the higher the accuracy and vice versa.

3. ECG Signal Simulation - Case 3

In Case 3 the Filter Tap=32 and the step size μ = 0.005, when applying these parameters in the MATLAB environment it results in the Figure 15.

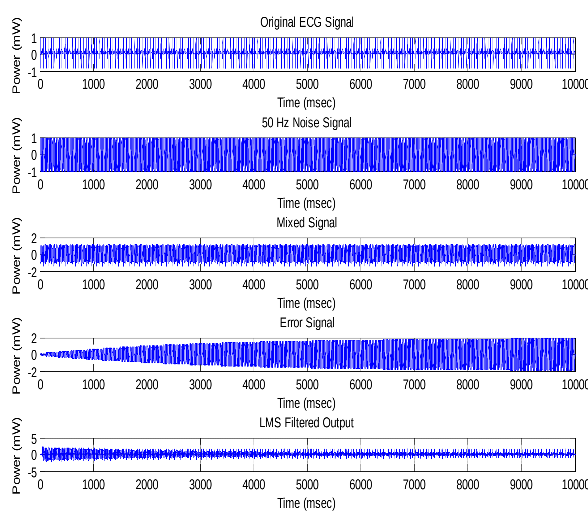


Figure 15: *ECG Signal Simulation - Case 3*

In the Figure 15 ECG Signal Simulation-Case 3, the value of filter tab has been changed from 16 to 32 and μ is taken as 0.005. The figure shows five different plots, first, second, third, fourth and fifth plots are taken as to be ECG signal, 50 Hz noise signal, mixed signal, error signal and filtered output signal respectively. The ECG signal of 50 Hz and noise signal of 50 Hz (which is supposed to be power line interference) are generated and then mixed together in the third plot of the figure. The fourth plot of the figure shows the error signal which has been calculated by the difference of original ECG and 50 Hz noise signal. The fifth plot shows the LMS filtered output of the mixed signal which is nearly same to the input ECG signal. This shows the ECG signal (50 Hz) and noise signals (50 Hz) are mixed together and the filtered out by using LMS adaptive filter. This figure can be compare with the ECG Signal Simulation-4 where the value of filter tab and μ are taken as 32 and 0.009 respectively.

4. ECG Signal Simulation - Case 4

In Case 4 the Filter Tap=32 and the step size μ = 0.009, when applying these parameters in the MATLAB environment it results in the Figure 16.

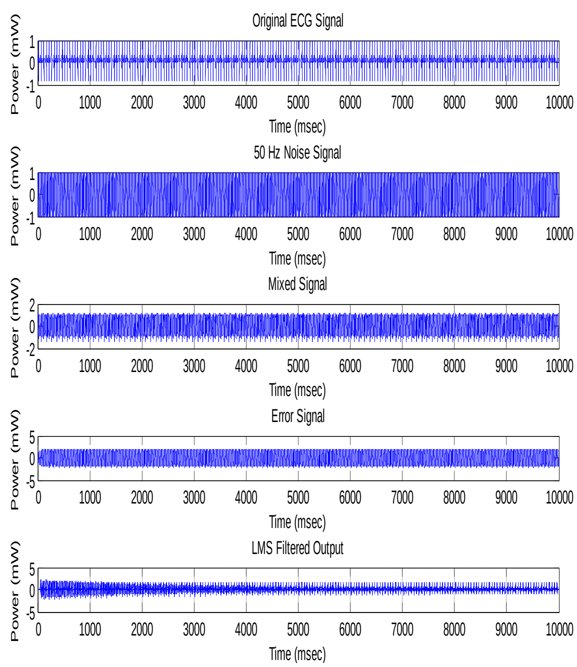


Figure 16: *ECG Signal Simulation - Case 4*

In the Figure 16 above, simulation graph the value of μ is 0.009 which shows the small change in the coefficient and the convergence of filter act as slowly. The figure shows five different plots, first, second, third, fourth and fifth plots are taken as to be ECG signal, 50 Hz noise signal, mixed signal, error signal and filtered output signal respectively. The ECG signal of 50 Hz and noise signal of 50 Hz (which is supposed to be power line interference) are generated and then mixed together in the third plot of the figure. The fourth plot of the Figure 16 the error signal which has been calculated by the difference of original ECG and 50 Hz noise signal. The fifth plot shows the LMS filtered output of the mixed signal which is nearly same to the input ECG signal. So it is concluded that with large step-size the filter convergence takes place fast. It is also concluded that with the large value of μ, the filter convergence act as fast. By changing the results and convergence rates, finally it is concluded that the LMS adaptation took place properly and it performed adaptation.

B. Removing of Harmonics and High Frequency Noise from Original ECG Signal

To the verification of noise reduction from original ECG signal, another method has been developed for the implementation and analysis of different noise reduction technique in ECG signal. The test for the simulation of ECG signal has been taken by adding different noises. Humming which is also known as low frequency noise has been taken and added in the original ECG signal. The major work is concerned with reducing undesired frequency harmonics while saving the original information in the signal as much as possible during this elimination process. Biomedical signals play an important role for monitoring of patient's ECG. High frequency noise can also be added with ECG signal while diagnosis of patient in hospital or somewhere else. This high frequency noise can be added due to the use of Mobile phone near patient or some other sources. The original ECG signal mixed with noise is passed through notch and low pass filter, which filter out hum and high frequency noise portion from the ECG signal. With this process the unwanted noise signal can be reduced from ECG signal, and then original ECG signal with much less noise can be obtained. The graph of original ECG signal with hum (low frequency noise) and high frequency noise is displayed in Figure below.

1. Simulation for Harmonics and High Frequency Noise and its Reduction from Original ECG Signal

In Case 5, several parameters were added to simulate a most corrupted MATLAB environment to the ECG input signal, it results in the Figure 17.

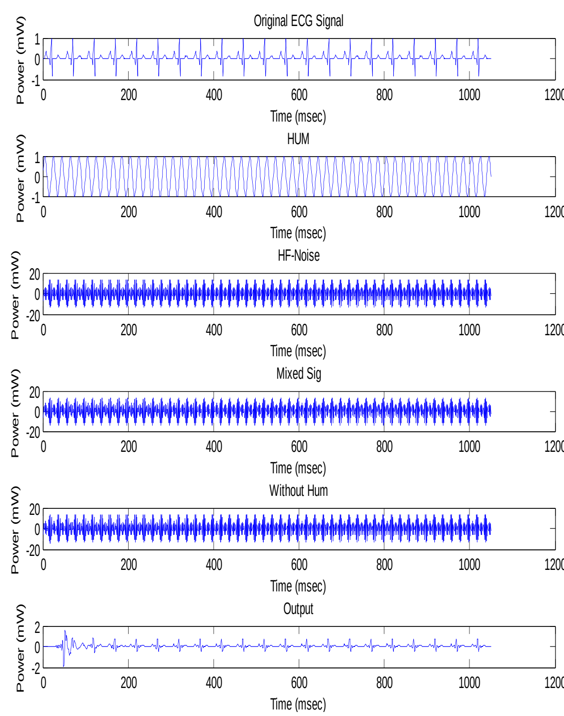


Figure 17: *Original ECG Signal with HUM and High Frequency Noise Simulation by Notch Filter*

Figure 17 shows six plots. In the first plot of the figure the original ECG signal has been displayed and the hum and high frequency noise are displayed in second and third plots respectively. Then both noises humming and high frequency noise has been added to the original ECG signal, which is shown in the fourth plot as mixed signal. Notch rejection filter is employed for the removal of humming (50 Hz). As there are two different noises in this case i.e. humming and high frequency noise, both noises after passing through the notch filter, the signal with high frequency noise remain with the original ECG signal in the fifth plot of the figure. Finally the corrupted signal is passed through low pass filter to retrieve the original ECG signal. The high frequency noise is then filtered out by using low pass filter and the desired output of the ECG signal has then been achieved at the final output. By comparing the graph of input original ECG signal with the output, it is concluded that the overall result of method and technique developed in this thesis is achieved. The goal for the noise reduction of different noise frequencies has been performed satisfactory. After analysing the filtering technique for humming and high frequency noise from ECG signal, it is concluded that implementation and analysis has been performed more or less the same and it also shows the method and filtering technique developed for the reduction of harmonics and high frequency noise from ECG signal performed its operation and meet the required results.

C. Signal to Noise Ratio (SNR)

After all the simulations are implemented, it is a major need to measure the degree of accomplishment in order to evaluate the whole process in all Five Cases applied. This is done through measuring the signal to noise ratio in all Five Cases before and after the different techniques implementations as shown in Table 1.

Table 1: *Signal to Noise Ratio Comparison between All Five Cases*

******	Case 1	Case2	Case3	Case 4	Case 5
SNR before filtering in dB	0.561	0.561	0.561	0.561	0.0142
SNR after filtering in dB	2.934	2.891	2.893	2.840	4.8262

In the Table 1: Signal to Noise Ratio Comparison Between All Five Cases, we can notice that all the filtering techniques have achieved high percentages of noise reduction in all Five cases, however, after observation it has also been noticed that in Case 1 the result is best among the first four cases, the reason behind that is the small values assigned for the filter tap and the step size (μ). It is concluded that the less the values of the filter tap and the step size (μ) the higher noise reduction and better results achieved and the opposite is true.

X. Conclusion

This thesis is devoted to the problems and solutions on reduction of Power Line Interference and other Frequency Tones from Signals. The thesis has proposed a solution for the power line interference, its respective harmonics and high frequency noise interferences from original ECG signal. The results that have been obtained were a fulfillment of the assigned objectives by the thesis. The test for the simulation of ECG signal has been implemented. The signal is corrupted by power line interference of 50 Hz. It is observed that the frequency of the power line interference is 50 Hz which is then mixed with original ECG signal, it is also examined that the mixed signal is displayed on the plot. After passing through LMS algorithm the filtered output is nearly same as the input signal with some acceptable distortion range. The ECG signal graphs described in the simulation results verify the adaptation of the LMS adaptive algorithm by changing various parameters like step size, convergence value (μ) and filter taps have various effects on the output graphs. The result shows that LMS is an effective algorithm used for the adaptive filter in the noise reduction implementation. By increasing the filter order it shows a convergence rate but makes the results more precise and by decreasing the step size value it creates the slower convergence but improves the stability and accuracy. The recovered signal closely resembles to the original simulated signal subtracting the noise. It can be seen that the implementation of the algorithm functions as correctly and efficiently. By comparing the graphs of the input signal of ECG and output signal, it is noticed that the simulation program performs satisfactorily and that noise reduction from original ECG signal is achieved. The overall performance of LMS algorithm for power line interference is implemented. Furthermore the general notch rejection filters method also performs the required operation while filtering the noise from original ECG signal. This technique for the investigation, implementation and analysis of reduction of harmonics and high frequency noise from original ECG signal performed satisfactory. It is concluded that the low frequency noise (hum) and high frequency noise can be reduced from original ECG signal by the implementation of general notch rejection filters method and the low pass filter; the desired result can be achieved accurately.

XI. Recommendations and Suggestions for Future Work

The depth knowledge achieved is in a number of aspects by using digital signal processing techniques with MATLAB package for medical monitoring equipments such as ECG. It provides the real concepts along with the theoretical backgrounds of reduction of power line interference, frequency tones and high frequency noise from original ECG signal. This enhances the understanding and self confidence in the field of electronics and biomedical engineering. In the thesis, the adaptive signal processing filtering technique based on LMS algorithm could be implemented for more signals and also improvement of the thesis can be further implemented with different algorithms such as NLMS and RLS to achieve more accurate results if possible. It could also be investigated and implemented for the reduction of multiple of harmonics from ECG signal. In further research it would be of interest to make a broader study and look at some companies for related project for the implementation. A hardware design implementation of the thesis will also be considered a major breakthrough.

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