# NEW APPROACH TO ELIMINATE NOISE ATTENDANTS ECG SIGNAL CORRUPTED

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**ABSTRACT**

The electrocardiogram signal for humans is very weak physiologically and in low frequency, and it is usually performed to detect the presence of an irregular heartbeat, as well as to diagnose the causes of chest pain, especially to ensure that there is no heart attack. This signal is captured by electrodes attached to the surface of human skin. As a result of the phenomenon of polarization between the electrode and the skin tissue, this will lead to dangerous interference in the ECG signal such as external electromagnetic field interference from power line intrusion, noise from random body movements, breathing movements, EMG noise, which must be removed before diagnosis. This study proposes to design and implemented three types of band-stop filters (Butterworth, Notch, and LMS adaptive stop-band filters) based on field-programmable analog array technology to process the damaged ECG signal. The properties of these filters have been improved and their parameters are calculated to minimize the noise from corrupted ECG signals. The obtained results clearly show that the proposed filters are effective in eliminating noise due to the fact that all parameters are significantly improved in comparison with the damaged signal is an indication that the proposed digital filters are working satisfactorily. The results which are given by the adaptive LMS filter provides the best performance compared with the other band-stop filters through improving the signal-to-noise ratio (20.423 dB). The Notch filter offers obvious advantages and achieves a signal strength of 0.342 dB after filtration and retaining detailed information on the QRS complex of the ECG, which is significant for the feature extraction of ECG signals and for pathological diagnosis.

**Keywords:** *ECG Signa,l Notch Filter, Adaptive Filter, Butterworth Filter, FPAA.*

## 1. INTRODUCTION

The filters are a fundamental portion of every communication system either to strip noise or to elicit the information [1]. A most demanding task for the system engineers is the realistic design and implementation of the compact filters [2]. The ECG is a greatly utilized non-surgical intervention mechanism used to gauge heart cadence via electrical activities [3]. Electrocardiogram (ECG) signal is a form of the biometric signal created by the heart through generating electrical activities. The electrical competence of the heart can be set in the sinoatrial node due to the persistent blood inflation activity in the human body. The ECG is a non-periodical signal, but extremely recurrent since it has three essential parts. They are P, QRS, and T waves. ECG is unparalleled, and it can exclusively be found in a spirited person. Moreover, the signal cannot be simulated by another spirited person. Therefore, ECG is possible to be utilized as a private identifier or as a biometric factor [4]. The ECG signal in most cases includes various types of noises like baseline deviation, muscle deformation, contact and quantization noise, power line intervention, and electrosurgical noise. This fuss decays the goodness of ECG signals throughout their treatment, causing false alarms. Therefore, it is necessary to strip the various noises and increase the SNR ratio of every heartbeat [5]. When searching in previous studies on noise removal from the ECG signal, it can be found that there are many researchers who have used different methods to remove this noise such as:

P. Mukherjee and A. Bakshi in (2020) propose a processing method of noise reduction ECG signals. The noise distorted ECG signal is thread through the Butterworth filter for baseline wander elimination. The high-frequency noise of distorted signal then passed through gaussian window filter because of its rise of correlation value then to fine the signal Savitzky-Golay filter is used. Distinctive ECG signals are taken from the MIT-BIH database to confirm our proposed Method utilizing MATLAB programming [6].

C. T.C. Arsene, et al.(2019) suggested two DL patterns, jointly with a norm wavelet-based method for decrease noise from ECG signals. First, applied CNN as a paradigm eligible for refusing extreme levels of noise in the ECG signals, a case, it has not been previously touched upon. The second DL pattern is an LSTM model, be composed of two LSTM layers. A wavelet technically depends on an empirical Bayesian procedure with a Cauchy prior is also applied for a comparison with the DL models, which are trained and examined on two synthetic datasets including genuine ECG signals. The outcomes display that the CNN paradigm is pre-eminent to the LSTM paradigm in the current accuracy both in the quality of outcomes and computational time [7].

Syed et al.(2018) present perception about how can PLI impact the biomedical signals like ECG and various filtered techniques utilized to give a fair thought of their features and restriction in decreasing the noise in an effective method.Various methods have been proposed to remove the noise signal from the damaged ECG using the conventional Butterworth Stop Band filter for different orders, the Notch filter, and the LMSAF, although the filtering process becomes difficult due to the presence of different types of noise [3].

R.R.Thirrunavukkarasu et al. (2020) developed an effective model for reduced ECG noise of the mother and fetal ECG signals which are affected by various external noises and internal noises using the Adaptive Threshold Filter(ATF), Extended Kalman Filter(EKF), and Dynamic time wrapping(DTW) algorithm.By merging the features of this method the noises can be effectively eliminated. MIT BIH arrhythmia database is used. This proposed method includes four strides.The second stride includes the pulling out fetal signal from mother signal and the next step is the highest peak rectification of mother and fetal ECG and the final stride is the removed noise from mother and fetal ECG using Dynamic Time Wrapping algorithm [9].

U. Biswas and Md. Maniruzzaman (2014) contrasted the characteristics of two adaptive filters, like normalized LMSAF and repetition-least-square adaptive filter (RLSAF with a conventional notch filter in frequency and time fields to strip the power line intervention from the ECG signal. Spectrogram analysis addition to power spectral intensity (PSI) is executed. The results of the analysis of the time and frequency domain clear the PSD worth gained for adaptive LMS filter is preferable than that of adaptive RLS and conventional notch filters [9].

M. Wasimuddin and N. Gupta (2020) used LMSAF adaptively to delete the mother's heartbeat signal from the fetal electrocardiogram signal. The reference signal is used to delete the intervention signal from the estimated signal. This was done by the LMSAF. A good correlation must be between the reference and the interference signal. It is a reminder that there is a rapport amidst the order of the filter and the impact of the output signal. [10].

Chang-Hsi Wu1,et al.(2019) proposed Moving Avera Filter (MAF) and the zero-crossing segmentation method to eliminate ECG signal baseline deviation. For that reason, the reconstructed ECG signal does not have an unusual wave shape [11]. Ngoc Thang Bui et al.( 2020) implemented IIR filters to eliminate environmental noise from the ECG signal. The high-pass IIR filter with five poles is designed and implemented by the master core, while the low-pass IIR filter implements by the slave core with seven poles and sampling frequency of ECG signals up to 400 Hz [12].

Lu al. (2019) suggested an adaptive two-stage filtre. In the first stage, the mentioned ECG signals are revealed under motionless events, and the noise signals also are acquired. In the second stage, the acquired noise signals turn into the reference signals, and eventually, the required ECG signal acquired. Based on what has been reached it can be said that the use of a two-stage adaptive filter can efficiently filter the motion artifacts, the filtered ECG signal is fully pure [13].

Arief et al. (2020) proposed a QVAT algorithm that automatically discovers QRS congregations is proposed and discovered R summits of an ECG signal. The algorithm is composites of several steps, which is a band-pass filter, exploration of variance, adaptive sill, and local maxima. Band-pass filters are utilized to minimize noise that can produce errors in the discovery of QRS waves. The probable noise caused by intervention caused by electromagnetic waves and the noise of muscle activity [14].

H. D. Hesar and M. Mohebbi (2020) presents a new Bayesian structure depending on a Kalman filter, which does not require the pre-defined paradigm and ability to adjust itself to various ECG morphologies. The suggested model, in contrast to previous Bayesian techniques, requires minimal processing and is only required to cognition the position of the R peaks to start ECG management. This model employs a filter bank including two adaptive Kalman filters, one for reducing QRS noise (partition of high-frequency) and the other for reducing T and P wave noise (partition of low-frequency). ECG signal features extracted from measurements are required for medical needs to detect heart distortions and various types of ailments. One of the criterion technologies that provides noise reduction and gets the features of the signal in the one-step forecast[15].

Carlos et al.(2018) develop UFIR-based fast algorithms in discrete-time state-space with an adaptive optimal averaging horizon, which is desired to reduce the mean square error. To make certain that the features of the ECG signal are adeptly conserved, it begins to be vitally important to utilize the related noise invalidation techniques and the keeping of typical waves like QRS complexes while decreasing noise. it is intractable to decrease the noise of the devices with traditional filter coefficients because the changing conduct over time of this noise is not precisely recognized[16].

To outdo the Limited possibilities of static filters, three types of band-stop filtering based on FPAA technology are proposed to reduce the noise attendant the ECG signal.The residuum of this paper is marshalled as the following: the related work section reviews the use of band rejection (notch, Butterworth, and LMS) filters to remove noise from an ECG signal. The background basics provide a theoretical basis for the filters used in the research using methodology that discusses the filter circuit realization mapped on the FPAA to remove noise from the ECG signal. Finally, simulation results and conclusions are reported.

## 2. RESEARCH METHOD

The methodology proposed in this paper includes the design and implementation of (Notch, Butterworth, and Adaptive Filters based on LMS) band stop filter modules to eliminate various noise from the ECG corrupted signal, using Field programmable analog array (FPAA) and the following is a description of the architecture design of each module.

### 2.1. Field Programmable Analog Array Kit

AnadigmDesigner2 is software for the Electronic Design Automation project used in conjunction with the Anadigm's 3rd generation of dynamically programmable Analog Signal Processor (dpASP) AN231E04 kit see Figure 1. The AN231E04 device is composed of a 2x2 matrix of fully Configurable Analog Blocks (CABs), encompassed by programmable interconnect resources and analog input/output cells with energetic components [17].

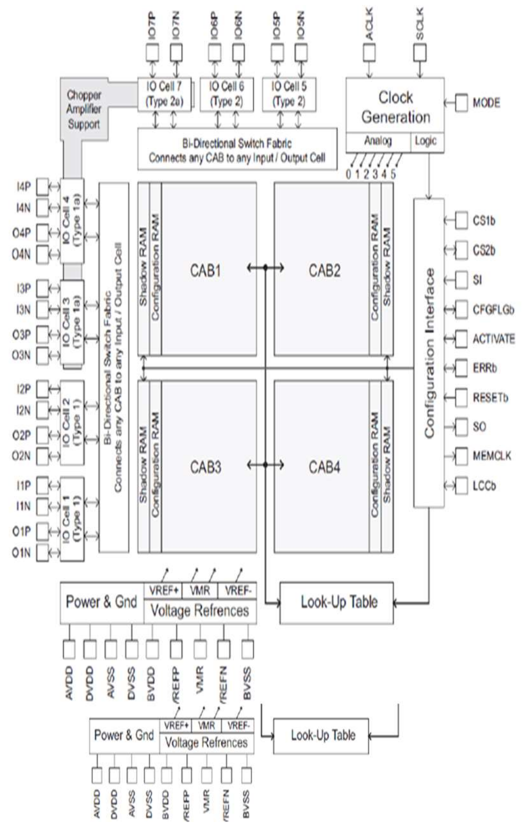


*Figure 1: Architectural overviews of Anadigm FPAA AN231E04[17]*

The Arbitrary Periodic Waveform Generator was used to build the ECG signal, according to the characteristics shown in Figure 2. Any signal can be represented by this CAM with no more than 256 samples or any number of samples determined according to the requirements of the user. Two periodical signals of similar frequency can also be generated by this CAM with up to 128 samples for everyone.

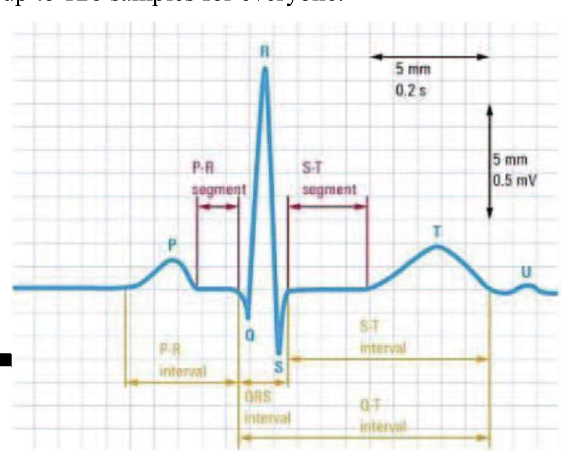

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*Figure 2: ECG signal characteristic [18].*

The corrupted ECG signal is constructed by adding noise to the original ECG signals generated see Figure 3.

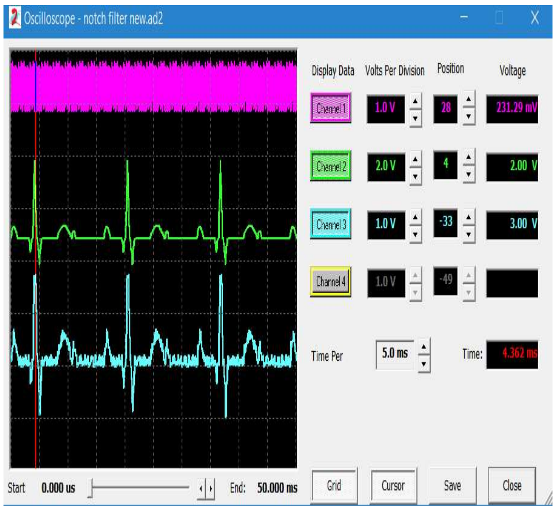


*Figure 3: The corrupted ECG signals*

The simulator pink probe indicates the noise signal, the simulator green probe represents original ECG signal while the simulator blue probe indicates a corrupted ECG signal.

## 2.1. Notch Filter design

A band-reject filter is defined as an electrical circuit that rejects a specific range of frequencies determined by the magnitude of the components and allows cross all the frequencies upper and lower that ambit. It can be split into a tight band and broadband, where the quite narrow band reject filter is also mentioned to be a Notch filter [19].The basic block diagrams of notch filtering are processed in Figure 4, where the d (n) is the input corrupts signal, X(n) is the clean signal, and N(n) is the noise [3].

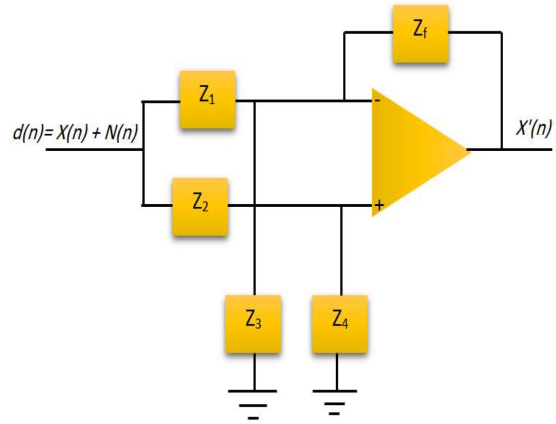


*Figure 4: Notch filter block diagram [28]*

The unique band rejection filter which has a high-quality parameter, and a very narrow stopping range is called a slit filter, see the frequency reaction of a Notch filter in Figure 5.which has a high-quality parameter, and a very narrow stopping range is called a slit filter, see the frequency reaction of a Notch filter in Figure 5,where r is the notch width [20].

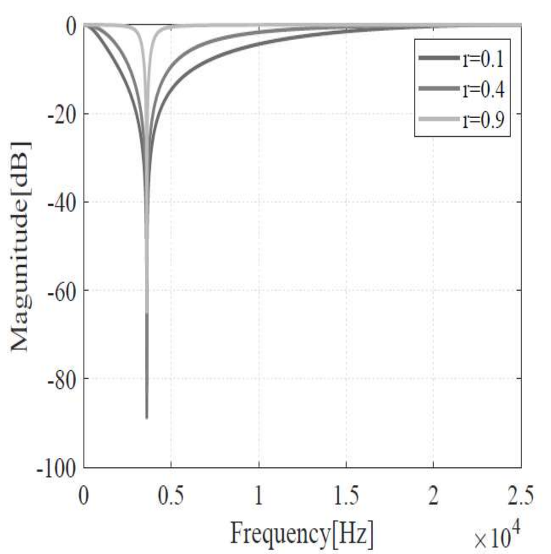


*Figure 5: Frequency response of the Notch filter[20]*

The simplest 2nd order notch filter is represented by two zeros placed *at* $\pm j\omega c$, where $fc$ ($\omega c = 2\pi fc$) performs the centre frequency. Thus, the transfer function can be represented as [21].

$$G(s) = \frac{s^2+w_c^2}{s^2+2\xi w_c s+w_c^2} = \frac{s^2+w_c^2}{s^2+2\,w_c\,cos\theta_s+w_c^2} \qquad (1)$$

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$$Q = \frac{1}{2\xi} = \frac{1}{2\cos\theta} = \frac{w_c}{2ʎ} \qquad (2)$$

Let $f_c$ be the cutoff frequencies and $f_1$ and $f_2$ be the lower and upper across the center frequency, so the 3dB bandwidth can be readily displayed as a function of the quality factor Q as (3).

$$BW(3dB) = f_c/Q = (f_2 - f_1) \qquad (3)$$

$$Q = f_c / (f_2 - f_1) \qquad (4)$$

Notch filter is created by combining the high pass and low pass filters in parallel connection during an amplifier circuit. It has two passbands and one stopband.$F_L$ is the cutoff frequency of low pass filter and $F_H$ is high pass filter cutoff frequency. So, the center frequency (*fc*) of the band-stop filter will be [20]:

$$f_c = \sqrt{F_L * F_H} \qquad (5)$$

Figure 6 display the performance of the Notch filter practical design in Anadigm Designer2 software. Full-cycle style CAMs (two biquadratic lowpass filter and two biquadratic high pass filter), and half-cycle style CAMs like the sum/difference stage are used. Sample/hold CAM is inserted to make sure the proper working of the circuit is done. The system is a switch with the two-phase clock of the frequency of 4 MHz.The characteristic of the designed notch filter is disply in table 1 with the power consumption of 163±49 mw. To satisfythe filter response considerations and to achieve close approximations of the ideal filter, this requires designing higher-order filters.

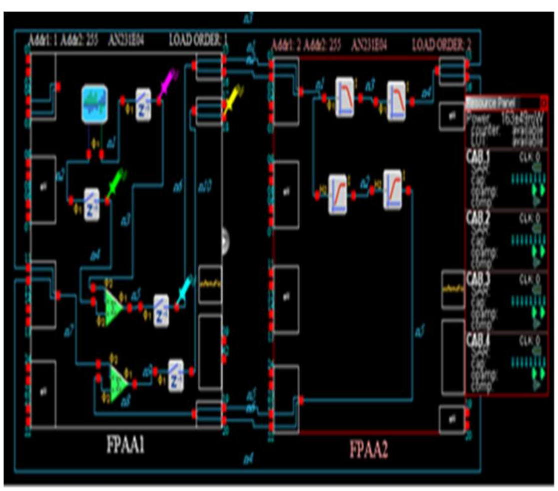


*Figure 6: Notch filter implementation circuit*

Table 1. *Notch filter characteristic*

| Type of filter | Corner Frequency | Gain | Quality Factor |
|---|---|---|---|
| High Pass filter1 | 80 Hz | 1 | 2 |
| High Pass filter2 | 80 Hz | 2 | 2 |
| Low pass filter1 | 30 Hz | 0.5 | 0.2 |
| Low pass filter2 | 30 Hz | 0.5 | 0.2 |

## 2.2. Butterworth Filter Design

British engineer Stephen Butterworth was the first to design this filter. It offered a perfect response with zero gain in stopband and unity in the pass and it is also recognized as an ideal filter. Hence the primary feature of Butterworth over other filters is a complete response at all frequencies [22]. The magnitude squared frequency response of a filter can be represented by the following relationship:

$$|H(jw)|^2 = \frac{1}{1+\varepsilon^2 Pn^2(w)} \qquad (6)$$

Where the filter order is *n*, *Pn(w)* is a polynomial of grade n, and w is a filter factor. The Butterworth filter has an extremely flat-exceed response to any amount of n [23]. Compared with other kinds of filters, this kind of filter is the preferable solution between the attenuation and the phase response.

It is occasionally called an utmost flat band filter because there is no undulation in the pass bandwidth or the stop bandwidth [24]. The basic block diagrams of the Butterworth BSF are processed in Figure 7 [3].

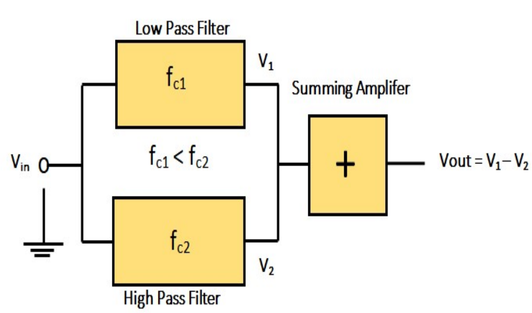


*Figure 7: Basic block diagrams of Butterworth Band stop filter [3]*

The Butterworth filter is an important filter and is often used to minimize the noise related to the ECG signal. The presence of noise will corrupt the signal and make the feature extraction and classification less accurate. To satisfy the filter response considerations and to achieve close approximations of the ideal filter, this requires designing higher-order filters. Figure 8 shows the circuit programmed on FPAA that implements the fourth-order Butterworth band-stop filter. Anadigm Filter® enables the design and implementation of high-performance filters in a fraction of the time required, different from any other automated device. The instrument automatically exam the realized CAM parameters. This allows for changing the design to get a better fit. In this paper, a Butterworth filter with specific parameters in table 2 is needed.

It is selected and adjusts the parameters in an automatically generated bode plot in Figure 9. The design required is sent to the Anadigm Designer®2 software for emulation and testing then prepared to transmit to the FPAA hardware for implementation.

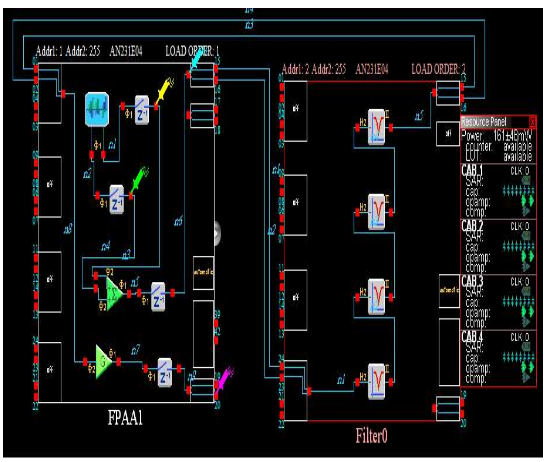


*Figure 8:Fourth order Butterworth band stop filter*

*Table 2. Butterworth filter characteristic*

| | |
|---|---|
| **Pass band ripple (dB)** | **3.01** |
| **Pass band gain (dB)** | **11** |
| **Stop band Atten. (dB)** | **20** |
| **Centre frequency (Hz)** | **35** |
| **Stop band widt (Hz)** | **80** |
| **Pass band width (Hz)** | **163.737** |

*Figure 9:. Frequency characteristics of the fourth order Butterworth filter*

### 2.3. Adaptive Filtering Based on LMS Algorithm

Figure 10 displays the basal idea of an adaptive filter. The adaptive filter is designed with the aim of obtaining a signal corresponding to the required *d(n)* signal after extracting it from the *x(n)* signal. The error signal is obtained by subtracting the required signal *d(n)*, from the filtered signal, *y(n)*.

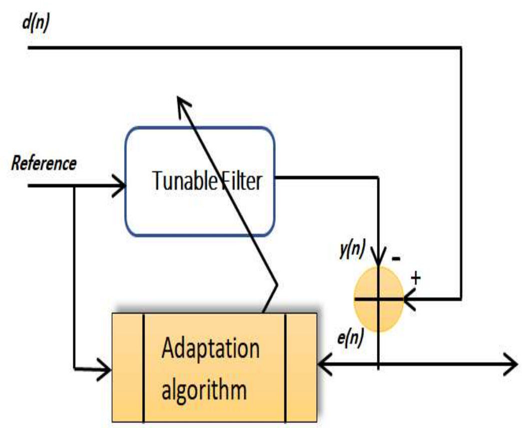


*Figure 10: Basic and typical block diagram of Adaptive filter [8]*

The error signal drives an adaptive logarithm that implements the filter parameters in a way that reduces the error signal. LMS and RLS algorithms are two of the most common adaptive filters that are quite used in communications to implement such missions as adjustment, echo abolition, noise revocation, and speech squeeze [26].

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Figure 11 displays the implementation of the current adaptive filter using a transversal FIR filter.The Unit-delay element, a multiplier, and an adder are three fundamental elements used to build the transversal filter. The filter's rank is determined by the numeral of delay units used in designing this type of filter. The adder unit task is, to sum up, all multiplier outputs, and make a total filter output according to the equation:

$$y(n) = \sum_{i=0}^{N-1} b(i).x(n-i) \quad (7)$$

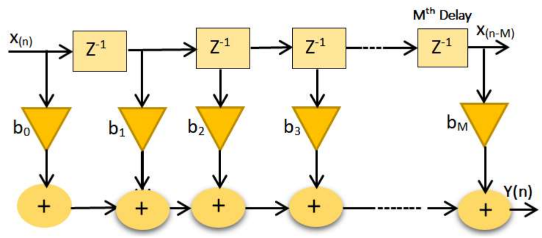


*Figure 11:. Transversal filter [26]*

The LMS algorithm is an extremely mightily utilized algorithm in adaptive filters. Minimum calculational intricacy, concourse in the steady milieu, equitable convergence in the average to the solution of Wiener, and stabilized attitude when performing with limited-arithmetical accuracy are the major characteristics that attract the utilize the LMS algorithm [27]. In this paper, parameters of the adaptive filter have been chosen so that it is always updated to reduce the total squared error from the time when the filter started the process to reduce the cost function. Several repetitions are used upon the transient portion until the LMS algorithm reaches the steady state. Through this duration, the adaptive-filter parameters, and the error of output are modified from their incipient rates to a near value to that of the optimal resolution. Figure 12 displays the construction of the band-stop adaptive filter. 4-output stage transversal was proposed as the fundamental for this adaptive filter.

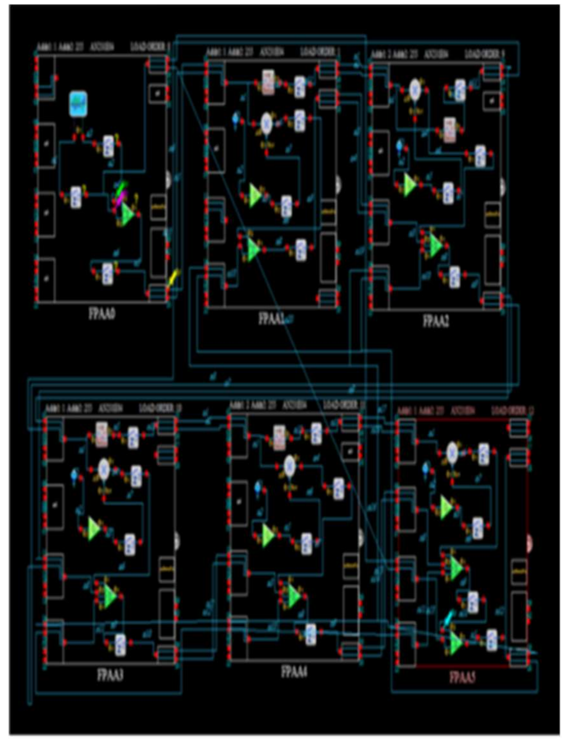


*Figure 12:. The structure of the band stop adaptive filter*

This type of filter has reprogrammable weighted taps built wherein the weight tap stays constant in the multiplier's unit reprogrammed, without the requirement for revolving the weight taps between the gathering of multipliers. A multiplier component is required for each tap summing unit is used to sum the results of every multiplication operations. Sample and hold CAM is inserted to make sure the proper working of the circuit is done. The reference signal is taken as the pure ECG signal generated by Arbitrary Periodic Waveform Generator CAM in $FPAA_0$.The prediction of e(n) must be as small as possible to get a great correlation between neighboring specimens of the input signal.If the forecast is perfect, the dynamic extent of *e(n)* should be junior to the dynamic extent of *x(n)*, bring about in a junior quantization noise for the identical amount of bits or the identical quantization noise with a junior amount of bits see summing amplifier CAM in Figure (12- FPAA1-5).

## 3. RESULTS AND ANALYSIS

In this section, the analysis results of filtering the ECG signal by Notch, Butterworth, and adaptive filters are compared, and the efficiency of each one is shown. The following

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subsections discuss the measurement results separately.

### 3.1. Notch filter results

The implementation of the circuit in Figure 6 was done and the output was plotted and clarified in Figure13. The output signals in Figure 13 represented by a yellow probe to simulate the anadigm designer of the notch filter are close to the original signal with 16.864 dB SNR attendant.

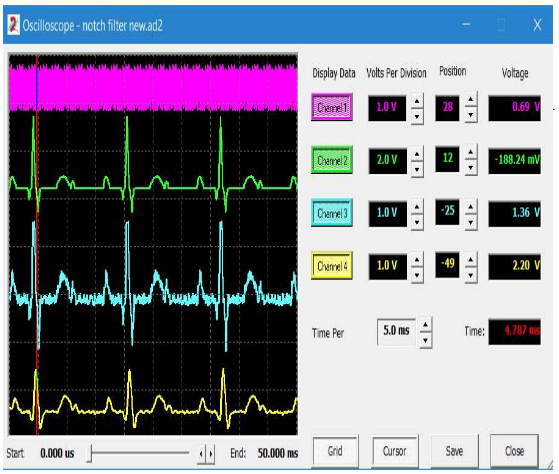

*Figure 13: Notch filter input and output ECG signals*

The feature provided by FPAA technologies allows the Notch filter to change its specifications towards application in the various areas including the medical field (hassle-free operations), where the challenge is to obtain accurate records of biomedical signals such as ECG that deal withbiomedical signals such as ECG that deal withbiomedical signals such as ECG that deal with interference caused by the power line.

### 3.2. Butterworth Filter Results

In this section, the simulated results of the proposed a 4-order filter designed in Figure 8 are presented and discussed which are obtained using the Andiagim simulator software is implemented in FPAA0 and the measured input-output ECG characteristic is shown in Figure 14.

*Figure 14:. Butterworth filter input and output ECG*

Where the time representation of the corrupted ECG signal with noise in the pink probe of the simulator and the output signal of the proposed Butterworth filter in the yellow probe of the simulator. It is seen that the distortion has been minimized. Despite the easily designed Butterworth filter using the Anadigm Filter® tool and the features provided by FPAA technologies, the performance of this filter in eliminating noise representedby SNR 16.6 dB is not ideal.

### 3.3 Adaptive Filter Results

The outcomes are calculated using the Anadigm designer2 and then transferred to the FPAA0-5 kit as shown in Figure15. The corrupted ECG signal which is taken as input is filtered using algorithms of the LMSAF. The performance of the authentic ECG signal is improved by passing it through LMSAF and then improving the SNR value of the ECG signal to reach 20.423 dB.

*Figure 15: Adaptive filter input and output ECG signals*

Table 3 gives a comparison of signal power between these three filters. It is clear that the response/settling time of the Notch filter is significantly preferable when comparing to other filters, which breed an impact on the parameters of the ECG wave.

*Table 3: Comparison between three type of filters*

| Performance Parameter | Filter | | |
|---|---|---|---|
| | Notch | Butterworth | LMS Adaptive |
| Signal strength ahead of filtration in dB | 0.133 | 0.201 | 0.394 |
| Signal strength after filtration in dB | 0.342 | 0.201 - 0.361 | - 0.364 |
| Impact on P, QRS, T | modified | modified | modified |
| Power consumption in mW | 163 ± 49 | 109 ± 33 | 819 ± 63 |

Another appropriate factor is the SNR of the LMSAF filter it is more than the Notch and Butterworth filters as shown in Table 4.

Table 4: *SNR for three type of filters*

| Filter type | SNR (dB) |
|---|---|
| Notch Filter | 16.864 |
| Butterworth Filter | 16.6 |
| Adaptive filter | 20.423 |

## 4. CONCLUSION

The main focus of the proposed task is to reduce noise and improve the efficiency of ECG devices. In this article, three different bandstop filters (Notch, Butterworth, and Adaptive filters) structures are designed and implemented based on FPAA technology. These Filters can execute ECG data nominate in real-time. The precise outcomes from the calculation of noise elimination, power consumption, and ECG signal power show that the Notch filter technique outperforms the other techniques applied for our evaluation and achieves a signal strength of 0.342 dB after filtration and it generates a better effect on the P, QRS, and t waves.It was found that the adaptive LMS filter denoised ECG signal with SNR= 20.423 dB associated with the ECG signal compared to the other filters under study.From all the comparison tables and graphs we conclude that LMS adaptive filter is the best filter for noise cancellation.The outcomes have signalized that characteristics obtained using the LMS filter are more robust and less susceptible to large perversions from normal values. This is absolutely a substantial utility for medical requirements.

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