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FPGA Based System to Remove Noise from ECG Signal

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ABSTRACT

Now a day's ECG Signal assumes a large task in the finding of different heart illnesses. An ECG signal is significant for medicine exercise on the heart. It is required to be clean. This paper is set as a technique of denoising the ECG signal. Where a High Pass Filter, Weighted Window Filter, and Savitzky-Golay Filter are applied to denoise the ECG signal. Distinctive ECG signals are taken from the MIT-BIH Database to verify our proposed method utilizing MATLAB programming. Which is realized as corresponding hardware in VIVADO making ARTIX-7 FPGA as the target device.

Keywords: *ECG signal filtering, Savitzky-Golay filter, Gaussian window filter, Baseline-wandering, EMG, SNR, Fixed point arithmetic, Field Programmable Gate Array (FPGA).*

I. INTRODUCTION

Electrocardiography is the manner closer to delivering an electrocardiogram ECG or EKG, a chronicle, a diagram of voltage versus time of the electrical motion of the heart making use of cathodes set at the skin. These cathodes discover the little electrical adjustments which are the final results of cardiovascular muscle depolarization pursued by using repolarization during every heart cycle (heartbeat). The ECG signal represents the electrical interest of the coronary heart. ECGs may be recorded as short irregular tracings or steady ECG observing. The figure of the ECG signal is shown below.

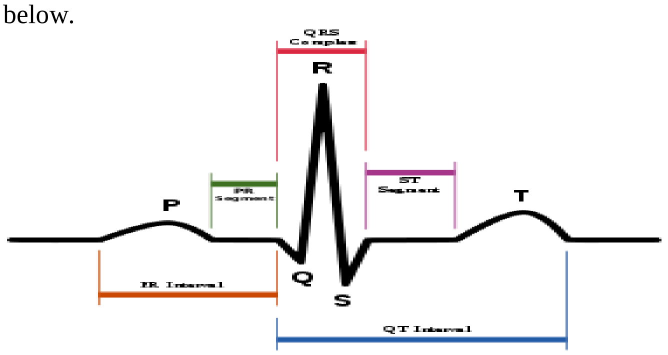

Fig.1. The Ideal and Theoretically shaped ECG signal

The above shown ECG signal has P wave which lies within the frequency variety of 0.67 Hz to 5Hz then a QRS complicated that's the overwhelming piece of the sign lies within the frequency variety of 10 Hz to 50 Hz and afterward T wave which lies in a frequency range of 1 Hz to 7 Hz. The Signal appeared above is the appropriate ECG signal, yet through and with the aid of they are not unreasonably Ideal. Typical musicality produces four elements – a P wave, a QRS, a T wave and a U wave – that each has a sensibly novel example.

- The P wave speaks to atrial depolarization.
- The QRS complex speaks to ventricular depolarization.
- The T wave speaks to ventricular repolarization.
- The U wave speaks to cardiovascular muscle repolarization.

2. LITERATURE REVIEW

From the paper of Henrik herlufsen "Windows to FFT analysis (part 1)" in Technical Review is introduced which led to creating a different filter for ECG signal Filtering [11]. In 1964 Savitzky and Golay published a paper "smoothing and differentiation of data by simplified least square procedure" [15]. In 2005 S.Hargittai [4], proposed an idea in the paper of "Savitzky-Golay least-squares polynomial filters in ECG signal processing" in Computers in Cardiology. In 2017, Nilotpal Das and Monisha Chakraborty published a paper "Performance of different filters in denoising an ECG signal [6]. Lecture note of the IEEE signal processing magazine issued by Ronald.W.Schafer in July 2011 got a lot of properties and information about the Savitzky-Golay filter [14]. In 2015 Shivangi Agarwal published a paper giving an idea about how can we implement the Savitzky Golay filter [13]. From the book of "Digital Signal Processing with Field Programmable gate array" by Dr.Uwe Meyer-Baese of third edition 2007[19] we got an idea on how to make DSP filters in Verilog. The Book "Fixed point representation and fractional maths" by Erick L. Oberstar gave an Idea of Fixed-point number representation of the 2007 edition [18]. Source of sample ECG signals are taken from MIT-BIH Arrhythmia Database [5] as 100.mat, 118.mat, etc. There are many published papers which are studied but the above-mentioned papers played a major role in my project so far.

3. DENOISING OF ECG SIGNAL

Denoising of ECG Signal is wanted because a Noisy ECG signal can cause false alarm and can result in incorrect diagnosis as a result it can result in the wrong remedy of the patient or over the remedy of the patient. The Noises in ECG are Baseline wander, Muscle artifact, Electromyogram noise The noise Baseline wander originates because of the Respiration. The muscle artifact noise ordinarily brought about through muscle action close to the head, for instance, gulping or head traits and Electromyogram noise which a High-frequency noise. The maximum power of the ECG signal lies in the variety of 0.5 Hz to a hundred Hz [20]. The objective is to denoise the ECG sign without losing a single

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statistic. QRS detector or any other ECG processing or thresholding blocks are not so robust they can leave out any beat count number so this pre-processing denoising gadget is needed to bypass a noiseless ECG sign to the next stages[21]. This paper first has a look at the performances of various filters as in [1], [22], [6] then it is going to be decided what will be the correct system to easy the ECG signal.

3.1 Removing of Low-Frequency Noise

In this section we're going to mention the low-frequency noise that is present inside the signal. Low-frequency noise refers back to the baseline-wander noise that shifts the baseline of the ECG signal. Abrupt adjustments within the ST phase are the most essential ECG marker when coping with acute coronary syndrome due to ischemia or myocardial infarction. To suppress this noise we're designing a high pass filter whose cut-off frequency should be greater than 0.5 Hz however we need to take care that it must not put off the P-wave whose frequency is from 1Hz to 7Hz. So taking the situations into the account the reduce-off frequency of the high bypass clear out must be among 0.5Hz to 1Hz. Butterworth IIR Filter architecture is used right here for the building of excessive bypass clear out [22]. Because Butterworth filter haven't any ripples within the stopband frequency which will now not allow an excessive amount of noise and greater over the frequency reaction of the sort of filter is flat. The frequency response of this filter of n^{th} orders is shown below.

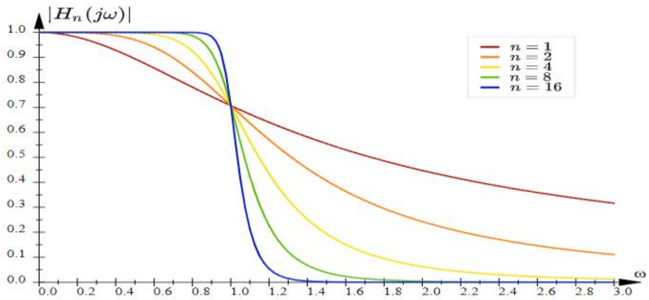

Fig.2. The example Frequency response of Butterworth filter

As seen in the figure shown as an example, as the order of the filter n increases the response gets stricter and clearer and gives better performance. The transfer function of the butter worth filter [12] is given by

$$H(\omega) = \frac{\omega^n}{\sqrt{1+\omega^{2n}}} \qquad (1)$$

3.1.1 Removing of High-frequency noise component from ECG Signal.

Suppression of high-frequency noise things from the ECG sign the usage of a traditional filter isn't always an efficient manner. Because in such cases we need to layout one filter which has a very sharp cut-off frequency and now have flat bandpass as well, this could be very much hard to the layout. So instead of the usage of everyday filtering technique it's miles wanted to perform the test on the filters select the right candidate for the ECG denoising as cited in [1]-[16] and some of them are my own. The list of filters that are to be examined are as follows

- Savitzky-Golay filter
- Smooth Filter
- Moving average filter
- Moving weighted window filter
- Gaussian filter
- Median filter
- Rectangular window fir filter
- Butterworth filter

3.1.2 Parameters set for testing

1. Savitzky-Golay filter: Order of the polynomial of the filter out = zero. Frame length = 17. A Savitzky-Golay Filter needs Pre-Determined estimations of order and frame length for its manufacture. This method of filtering has been proposed by Abraham Savitzky and Marcel J.E.Golay inside in the year of 1964.

2. Smooth Filter: Window size = 30. Smooth Filter is used for creating an approximated output wave by removing noises.

3. Moving average filter: Window length = 15. This filter runs by using calculating average in a selected frame given via us and calculating a moving average, a moving average is an estimation to dissect data points via making a progression of averages of various subsets of the full records collection.

4. Gaussian Window filter: Window size = 7. Window type: Gaussian window. A weighted common is an average that has growing variables to present various weights to records with the aid of an aspect at diverse points in a window. It is discussed in element within the later segment of the paper.

5. Gaussian filter: Window length =7. Gaussian filter out is clear out whose force reaction is a Gaussian feature or an estimation to it because a true Gaussian reaction isn't realized physically.

6. Median filter: Window length = 15 The Median Filter is a non-linear digital filtering approach, often used to suppress noise from a photograph or signal.

7. Rectangular window FIR filter: Window Size = 15. It is an FIR clear out designed via the use of square window feature with body duration of N, it's miles given through [12].

$$w(n) = 1,\ n = 0,1,2\ldots\ldots,N \qquad (2)$$
$$= 0,\ \text{otherwise}$$

8. Butterworth filter: Cut-off frequency = 50 Hz Order =10 Response = low pass. The Butterworth filter is a kind of filter out supposed to have a frequency as flat as conceivable inside the passband. It is also alluded to as a flat band clear out. So this is chosen for noise elimination.

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3.1.3 Parameters that are to be Evaluated

SNR: It is called Signal to Noise Ratio which means the ratio of the electricity of signal to the energy of the undesirable noise signal [8], it is given by.

$$\text{SNR value} = \frac{\text{Power of Signal}}{\text{Power of Noise}} \quad (3)$$

When the SNR is to be found in dB i.e. Decibel scale then there are two formulas, when we are finding them based on power [8] then,

$$\text{SNR} = 10\log_{10}\left(\frac{P}{N}\right) \quad (4)$$

Where P is termed as Signal power and N is termed as Noise power. Meanwhile on the other hand, when it is to be determined based on RMS values of signals and noise [9], [10] then,

$$\text{SNR} = 20\log_{10}\left(\frac{V_s}{V_N}\right) \quad (5)$$

A signal is dominating than the noise which means that the filter to evaluate some other parameter that's called correlation co-efficient which is denoted by COR [22]. The correlation Co-efficient is the measure of the relation among signals. If the COR is measured 1 then it manner the reconstructed signal precisely matches the Clean signal and it is the right fee of COR. The value or mathematical expression of COR is given [13] by,

$$\text{COR} = \frac{\sum_{i=1}^{n}(C_i - \bar{C})(Di - \bar{D})}{\sqrt{\sum_{i=1}^{n}(C_i - \bar{C})(Di - \bar{D})}} \quad (6)$$

Where C_i is the set values of n samples of a clean signal which has a mean value of $\bar{C}$. Whereas Di is the set values of the denoised signal which has mean value $\bar{D}$.

3.1.4 Results got through the Experiments

The experiment is carried out in the MATLAB software using the databases from [5] and got the performance evaluation of the filters as shown in the table below.

Table I. Performance results of different filters.

Filter	**SNR(dB)**	**COR**
Savitzky-Golay	22.69	0.801557
Smooth Filter	17.01	0.177373
Weighted window (Gaussian)	16.86	0.957624
Gaussian Filter	17.01	0.100202
Median Filter	22.82	0.896177
Rectangular Window FIR Filter	22.73	0.250770
Butterworth	22.69	0.801557
Moving Average	22.68	0.38644

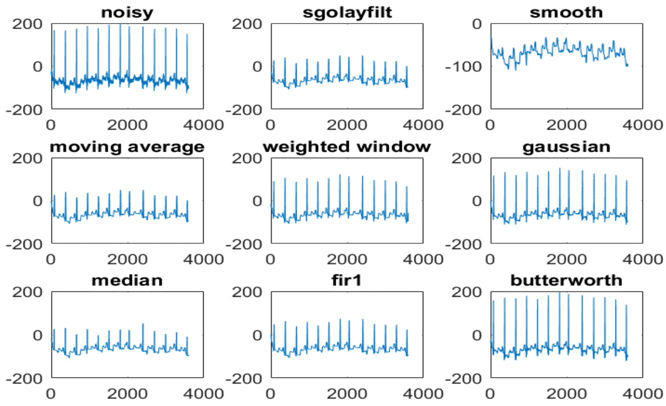

Fig.3. Time-domain response on ECG signal by different filters from top left corner raw signal, Output of Savitzky-Golay filter, smooth filter, moving average filter, weighted window filter, Gaussian filter, Median filter, Rectangular window filter, Butterworth filter.

Based on the table and after the evaluation of the filter out for denoising is chosen by considering each the parameter SNR and COR It is visible that those filter like Weighted window filter out has a sufficient SNR which is probably much less as compared to other filters but while it comes in case of the correlation coefficient it is the satisfactory filter. Another suitable competitor is the median clear out its SNR is much extra than the Weighted Window filter but the correlation co-efficient is much less than that of the weighted window filter. And for the smoothing operation, the Savitzky-Golay filter is the quality option because of its exact SNR and additionally exact COR as well. So, the answer to the disposing of high-frequency noise casting off is the cascading of any two filters. So, we have once more achieved the test but this time there's cascading of filters. Our first candidate is weighted window Savitzky-Golay clear out i.e the cascaded system of the weighted window filter and Savitzky-Golay clear out, and 2nd candidate is median Savitzky-Golay filter i.e the cascaded system of median filter out and Savitzky-Golay clear out. The assessment test is accomplished by way of adding excessive pass Butterworth filter also. Now let's talk about that match the result of the suit is as shown in Table 2.

Table 1: Performance comparison table

System	SNR(dB)	COR
HPF+WWF+SG	10.345474	0.998486
HPF+MF+SG	8.220329	0.995429

In the above-proven Table 2 HPF approach High pass filter out, SG means Savitzky-Golay Filter and MF signifies Median filter out and WWF method Weighted window filter (Gaussian

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Window filter). As shown in the table above there isn't always an excessive amount of difference between the two structures in case of correlation coefficient but while involves SNR the system which includes Median filter out offers poor performance compared to the device which includes Weighted window clear out. So According to the take a look at in overall performance parameter the cascaded system of Butterworth filter, Weighted window filter, and Savitzky-Golay filter are selected.

4. PROPOSED SYSTEM

In this section the whole system is realized. First of all the signal will first go to high pass filter for baseline correction then going to Gaussian window filter to remove some high frequency noise and finally go to Savitzky-Golay filter for smoothing. The figure given below is the diagram of the proposed framework.

Fig. 4. The block diagram for denoising an ECG signal.

Now we're going to explain every and every block of the block diagram.

4.1 High Pass Filter

High pass filter is the first block of this design it is used to get rid of the low-frequency noise of baseline wander which originates from the respiration through lungs which shifts baseline of the ECG sign due to that could make it tough to locate the R-top of the signal. To put off this noise, we need an excessive pass filter the cut-off frequency of the excessive skip clear out the need to be such that it does not do away with the p-wave of the sign. So ideally 0.5Hz cut off frequency is enough for suppressing noise but in exercise, we need to do plenty of trial and blunders first we want to know the essential frequency of the noise and additionally that of P wave then decide how much cut-off frequency should be utilized in the layout of clear out. Butterworth filter is sufficient for this cause with the cut-off frequency of 0.99Hz with a sampling fee of 500 Hz and order 7. The response of the input to the filter is the noise affected ECG sign we are getting a baseline wonder eliminated noise signal. The input and output response of the filter out in the Time domain is as shown in Fig. 5.

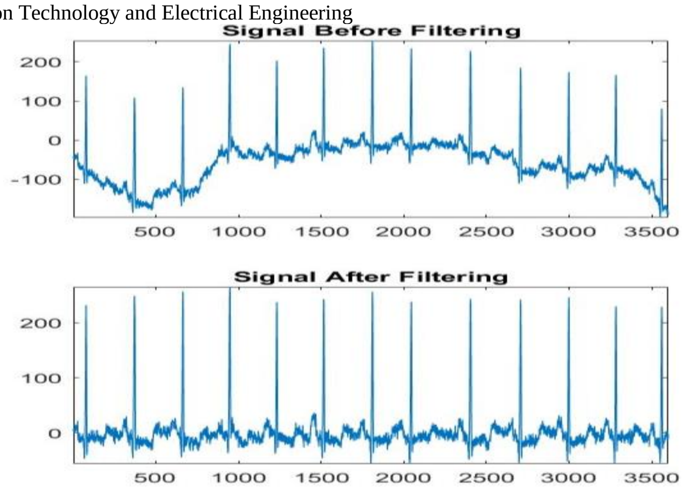

Fig.4. Removal of Baseline Wander by the High Pass filter with Input (above) and Output (beneath)

4.2 Gaussian Window Filter

What is the weight feature? A weight feature is a mathematical gadget utilized when acting an operation to give a few additives more "weight" or impact at the result than exceptional components in a similar set [11] after effect of this use of a weight characteristic is a weighted sum or weighted common. Weighted Window is referred to as in the sense that the coefficients of the window characteristic are giving a few samples of the sign a greater weight than the others that is best possible when the window feature will be non- planar, non-square so in this case Gaussian window is used to create this filter. Since the genuine Gaussian capacity has never-ending span, practically speaking we have to window it with a few preferred limited window, or shorten it. The coefficients of a Gaussian Window are processed from the accompanying condition:

$$w(n) = e^{-\frac{1}{2}\left(\alpha\frac{n}{(L-1)/2}\right)^2} = e^{-n^2/2\sigma^2} \quad (7)$$

Where $-(L-1)/2 \leq n \leq (L-1)/2$, and α is conversely corresponding to the standard deviation, σ, of a Gaussian arbitrary variable. The careful correspondence with the standard deviation of a Gaussian likelihood thickness work is $\sigma = (L-1)/(2\alpha)$.

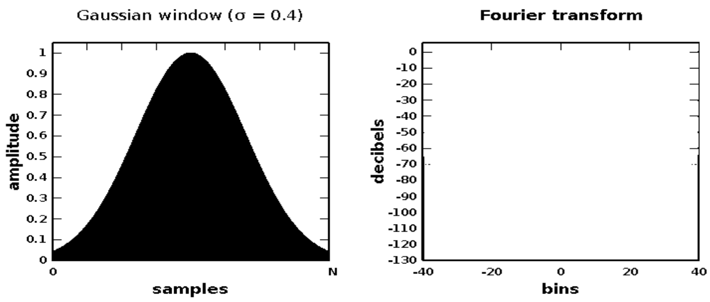

Fig. 5. Gaussian Expansion

The window size of 7 is sufficient. Effective will be using the Gaussian window. Giving a 15 coefficient FIR filter. The Time-domain Response on the ECG signal by the Gaussian window filter is given in the figure below.

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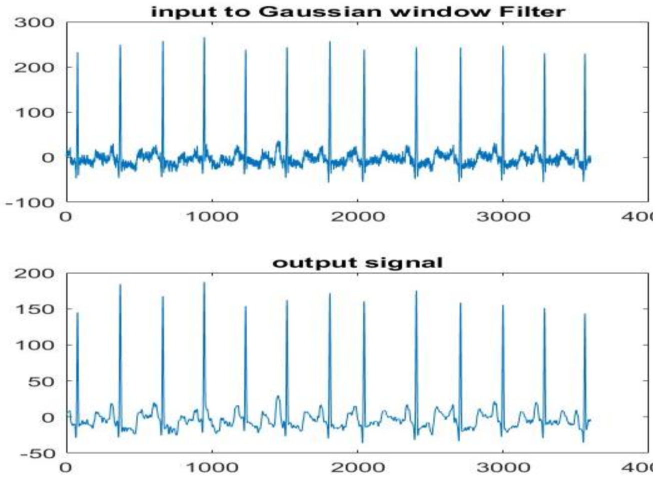

Fig.6. The time-domain response of Gaussian Window filter with Input (above) and Output(beneath)

4.3 Savitzky-Golay Filter

A paper named "smoothing and differentiation of data with the aid of a simplified least rectangular procedure" through Abraham Savitzky and M. J.E.Golay within the year of 1964 within the magazine Analytical Chemistry[15]. Later on S.Hargittai[4] and S.S.Mostafa[3] gave a concept that this procedure may be used to make a filter out which smooth the records of an ECG signal. Shivangi Agarwal used a lot of mixtures of this clear out[13] and additionally carried out on the EEG signal[16]. From the lecture note through Ronald.W.Schafer[14] the use of that expertise the Savitzky-Golay clear out is created. A polynomial filter out is regularly considered as piece-by-piece becoming of a polynomial function to the signal. The becoming is finished by a method of least squares (LS) estimate between the X matrix and therefore they vector:

$$y = Xa \qquad (8)$$

Where X is the design matrix for the polynomial approximation problem. The popular Least Square solution is given through:

$$a = (X^TX)^{-1}X^Ty \qquad (9)$$

Or,

$$a = Z.y \qquad (10)$$

Where,

$$Z = (X^TX)^{-1}X^T \qquad (11)$$

Z is referred to as the convolution coefficient. When we need to match a polynomial function of order p, we get a chain of equations within the following form:

$$Y_i = a_p.x_i^p + a_{p-1}.x_i^{p-1} + \dots\ a_1.x_i^1 + a_0.x_i^0. \qquad (12)$$

for i = 1...2n+1

Or, in general,

$$(C*y) = Y_i = \sum_{i=\frac{m-1}{2}}^{\frac{m-1}{2}} C_i y_{j+i}\ , \qquad (13$$

$$\frac{m+1}{2} \le j \le n - \frac{m-1}{2}$$

Using the above convolution equation in 13 or the use of the polynomial feature equation in 12 the ones equations are used in making of Savitzky-Golay filter.In Fig. 8 shown below is the frequency response of our Savitzky-Golay filter.

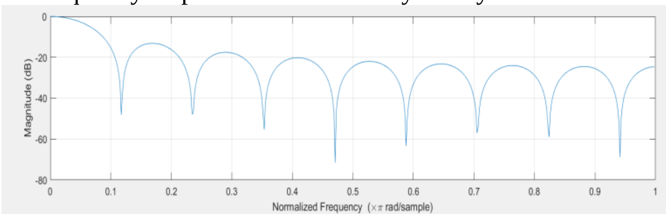

Fig.8. Frequency Response of Savitzky-Golay Filter

According to Schafer [14] and Hargittai [4] for filter to work properly it is found that the polynomial order should be in even number and frame length order should be in odd number. If the frame length is high or order is low or both the possibility the filter will work as a smoothing filter and when the frame length is low or order is high or both occurs then the filter will focus more on detail capturing. So in this paper the ECG signal is needed to be smooth so it is required to have low order and big frame length by ensuring that is does not change the signal. So order of polynomial of 0 and frame length of 17 is eligible. The time domain response of the Savitzky-Golay filter is shown in the figure below.

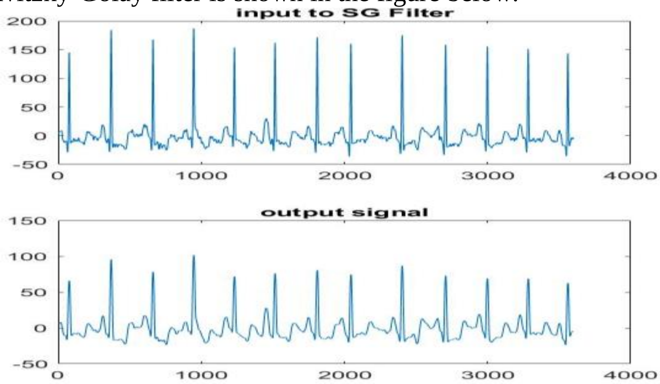

Fig. 9. Time domain response of Savitzky-Golay filter on ECG signal.

The desired result is achieved what is needed so the result is shown below in a summarised manner as shown below.

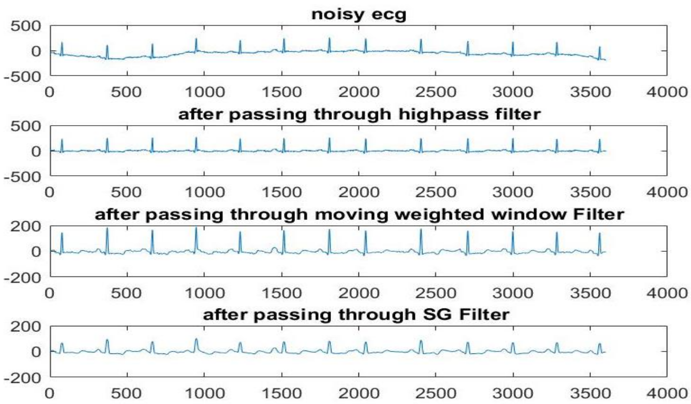

Fig.10. The summarized result input(top) output from High-pass filter(2nd from top) output from Gaussian Window filter(3rd from top) and output from Savitzky-Golay filter(last)

5. FIXED POINT ARITHMETIC

Some times more precision is needed than the precision provided by the integers but handling a floating point number is very hard, In Verilog writing a floating point number is very tedious so we need a fixed point arithmetic conversion to get the job done. The fractional numbers which cannot be expressed in Verilog we can realize in the Verilog program just by scaling them. In designing of filters if the coefficient values are rounded off the coefficient changes and the filter property and behavior changes. So, Q number format is used for scaling the coefficients in designing of different filters. Q format numbers are generally fixed point numbers, which are stored and operated upon as regular binary signed integers, thus allowing standard integer hardware/ALU to perform rational number calculations. The notation of Q format is given as $Q_{m.n}$ where Q indicates that the number is in the Q format notation [23], m is the number of bits set aside to designate the two's complement integer portion of the number, exclusive or inclusive of the sign bit therefore if m is not specified it is taken as zero or one, n is the number of bits used to designate the size of the fractional portion of the number, i.e. the number of bits to the right of the binary point. If n = 0, the Q numbers are integers – the degenerate case. The Q format number can be written in the form of $Q_{m.n}$ or Q_n format. If the m is ignored and if there are any bits empty before the fraction bit those empty bits are chosen for non-fractional part and sign part. For example in a 32-bit device if Q20 format is used this signifies that 20 bits are taken for representation of fractional part and remaining 12 bits in which 11 are for non-fractional part and 1 bit is sign bit. These are the following way to convert a number into Qn number:

- Multiply the floating point number by 2n
- Round to the nearest integer.

In this project Q_{20} format is used to represent the coefficients of the filters that are designed in this project. I.e. 20 bits are given for the fractional part remaining 11 bits are for non-fractional part and 1 bit is sign bit[18]. The coefficients that are generated from the MATLAB software are converted into Q20 format by multiplying them by 220 and round them to the nearest integer. Similarly if a Q20 number is to be converted to a normal float number again just divide the number by 220. to make sure satisfactory fixed-point operation of the IIR filter, we'll examine the subsequent intimately below[17].

- Coefficient Quantization
- Internal Quantization
- Overflow
- Stability

Coefficient Quantization: Coefficient quantization affects the frequency response. to seem at the effect of quantization, it's useful to seem at the filter within the z-domain and use pole/zero plot. This shows how the zeros (depths within the frequency response plot) and poles (peaks within the frequency response plot) are positioned. In fact, the difficulty with IIR stability has got to do with the denominator coefficients and their positions, as poles, on the pole/zero plot.

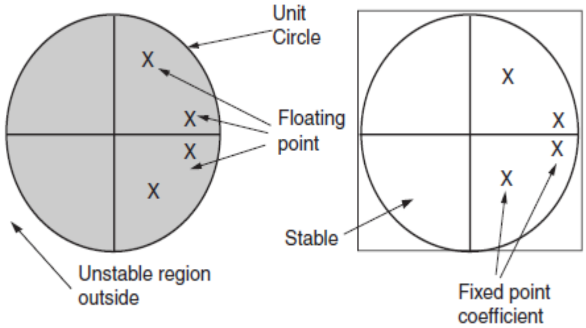

Fig.11. PZ plot in theory and in fixed point arithmetic[17]

The poles for the floating-point version of the plot are shown on the left, and it can be seen that they are within the unit circle, that is, the values of the coefficients are less than 1 and thus in the stable region. After the coefficients are quantized, these poles move, having an impact on the frequency response, and if they move onto the unit circle, that is, the poles equal 1, then potentially it's an oscillator, and if the poles become greater than 1, then the filter definitely becomes unstable. the pliability of the FPGA fabric allows larger multipliers to be constructed to scale back the consequences of coefficient quantization.

Internal Quantization: With a DSP function, there are multiplication and addition/subtraction operations. However, there's bit growth thanks to these operations, and at some point, the bit widths need to be reduced. Operations like wrapping/saturation for the foremost significant bits and rounding/truncation for the smallest amount significant bits need to be used. More detail are often found in DSP reference books. The rounding process reduces the bit-width, but it's a source of noise and contributes to output round-off noise and, hence, affects the signal-to-noise. The pliability of FPGA architecture allows for increasing the word length and reducing the round-off noise.

Overflow: For a fixed-point implementation, there is a fixed bit width and, hence, a range. As a result of calculations, the filter may exceed its maximum/minimum ranges. For example, a two's complement value of 01111000(+120) + 00001001(+9) = 10000001 = (-127). So, the large positive number becomes a large negative number. This is referred to as wraparound, which may cause huge errors. Using saturation logic can affect this example. Within the example, the results would be 01111111 (+127).To minimize the consequences of overflows, scaling are often used. Therefore, values can never overflow. There are different sorts of scaling and these tend to be employed by DSP processors to suit within their fixed structure. However, this has an impact on the signal to noise ratios.

Stability: Looking at the feedback path of the IIR filter, it is basically a higher-order polynomial [17]:

$$1 + a_1.Z^{-1} + a_2.Z^{-2} + a_3.Z^{-3} + \ldots + a_n.Z^{-n} \qquad (14)$$

The MATLAB provide mathematical functions like roots to assist solve these equations and supply the roots of the polynomial. Any of the roots are greater than 1 indicates instability. It also indicates the amount of poles that are on the brink of the unit circle, and then can highlight which

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coefficients may have more attention and thus more precision bits.

6. MATLAB FDA TOOL

The Filter Design and Analysis Tool (FDATool) may be a powerful interface for designing and analyzing filters. FDATool enables you to quickly design digital FIR or IIR filters by setting filter performance specifications, by importing filters from your MATLAB workspace, or by directly specifying filter coefficients. FDATool also provides tools for analyzing filters, like magnitude and phase response plots and pole-zero plots. FVTool, which may be launched from FDATool, provides a separate window for analyzing filters. you'll use FDATool as a convenient alternative to the instruction filter design functions. This chapter contains the following sections:

- Making of IIR filters
- Making of FIR filters

This tool is generally used for computing the coefficients of the filters.

A. Making of IIR filters

The IIR filter can be made by two ways one is by making nth order filter or decomposing them into bi-quad filter, The pole zero plot of the filter is shown below.

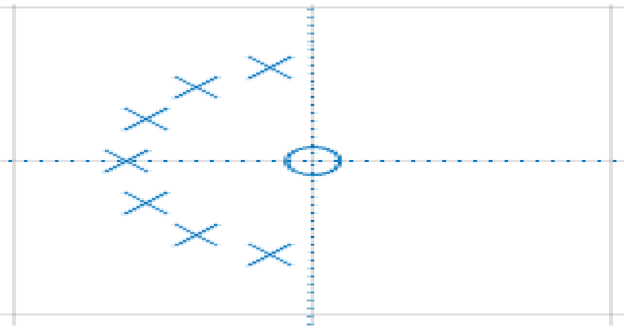

Fig.12. Pole-Zero plot of IIR filter

The issue with Nth Order structures [17] is that as the order of the filter increases, so does the complexity of the implementation. The results in worsening range and precision of numbers, and for larger number of taps, it becomes a challenge. The round-off errors that come from all the multiplications and additions contribute to rounding errors. The structure used to minimize quantization errors is the 2nd order filter, also known as a bi-quad. This section will show the making of IIR filter with pictures.

Step1: Put response type, cut off frequency, and other specification

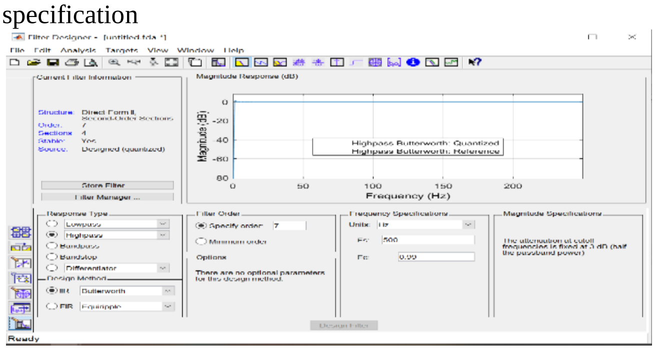

Fig.13. Step1 to make IIR filter

Step2: choose the length of fraction and input and output

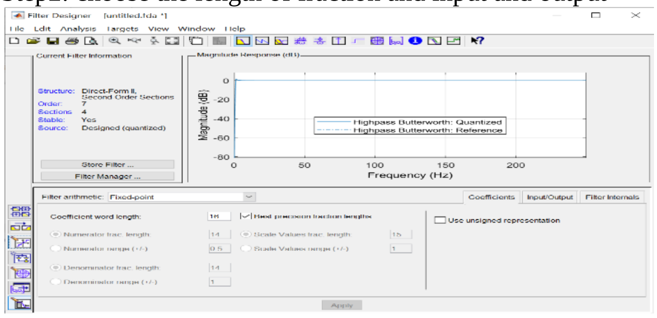

Fig. 14. Step 2 to make IIR filter

Step 3:Click of Generate HDL

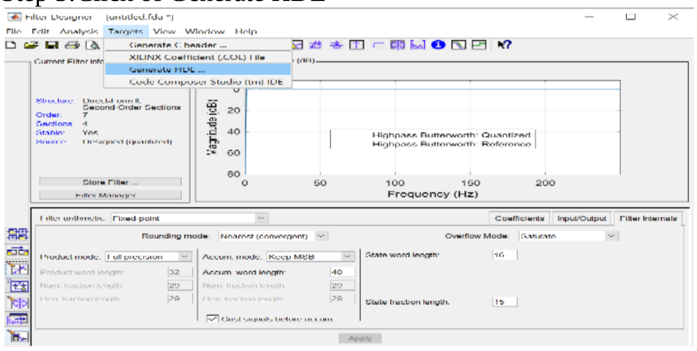

Fig. 15. Step 2 to make IIR filter

B. making of FIR filters

In this section making of FIR filter is shown in FDA using Gaussian filter as an example.

Step1: Give the function of the filter or the list of coefficient

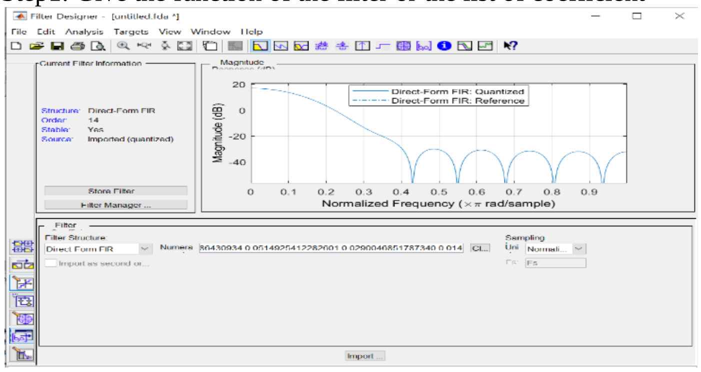

Fig. 16. Step 1 to make FIR filter

Step2: click on Generate HDL after choosing sufficient word length

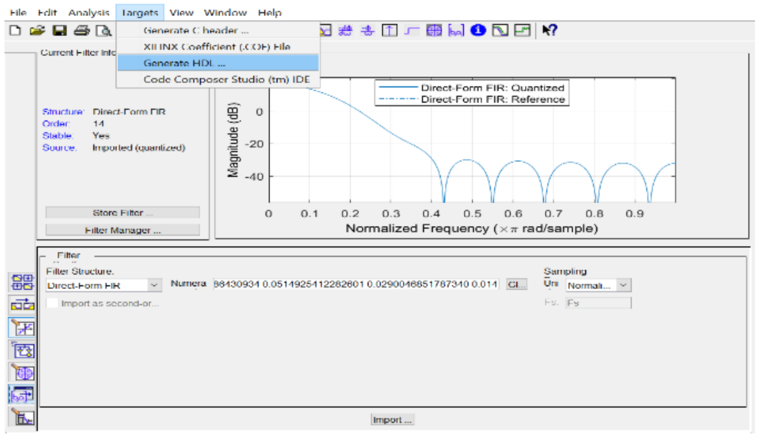

Fig. 17: Step 2 to make FIR filter

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7. RTL IMPLEMENTATION

Any set of rules or mathematical formula which are modeled in MATLAB that are transformed to its equal RTL implementation using Xilinx Vivado layout suite 2017. The Xilinx Artix-7 FPGA[22] having a specification of 7a100tcsg324-1 is selected for FPGA verification because it has XADC and XDAC interface[23]. RTL stands for the Register transfer stage. It is a layout abstraction that models a synchronous digital circuit in terms of the drift of virtual alerts (facts) among hardware registers, and consequently the logical operations performed on those signals. Register-transfer-level abstraction is employed in hardware description languages (HDLs) like Verilog and VHDL to make excessive-degree representations of a circuit, from which decrease-stage representations and ultimately actual wiring is frequently derived. Design at the RTL level is a typical exercise in the current virtual layout. Systems which can be carried out in Verilog are shown under:

1. Butterworth filter (high pass, f_C = 0.99Hz, order = 7)
2. Gaussian window FIR filter (number of coefficients = 15)
3. Savitzky-Golay filter (poly.order = 0, frame length = 17)

The whole gadget is made for 32-bit information transmission and a sampling frequency of 500 Hz is selected. The improvement of most of the filters is done via using dataflow modeling of the sign-in line with the algorithm and mathematical modeling of the signal.

7.1 Butterworth Filter

The Butterworth IIR filter is created by using direct form 2 structure, at the side of cutoff frequency of 0.99 Hz and order of seven, The quantized coefficients obtained from MATLAB are shown inside the desk beneath. These coefficients are then transformed to a 16-bit constant-factor format using the matlab FDA tool.

Table 3. SOS coefficients of Butterworth filter

SOS	Numerator Coefficients			Denominator Coefficients		
SOS1	1	-2	1	1	-1.99432	0.99447
SOS2	1	-2	1	1	-1.98445	0.98460
SOS3	1	-2	1	1	-1.97767	0.97783
SOS4	1	-1	0	1	-0.98763	0

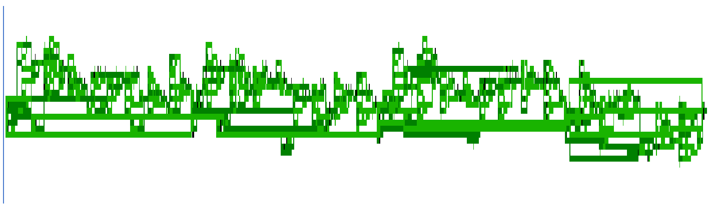

Fig. 18. RTL implementation of Butterworth filter

The utilization summary table is shown below.

Table 4.Utilization summary of Butterworth filter.

Site type	used	available	Utilization%
Slice LUTs	4083	63400	6.44
Slice Registers	377	126800	0.30
DSP48E1	57	240	23.75
Bonded IOB	67	210	31.90
Clock(BUFGCTRL)	1	32	3.13

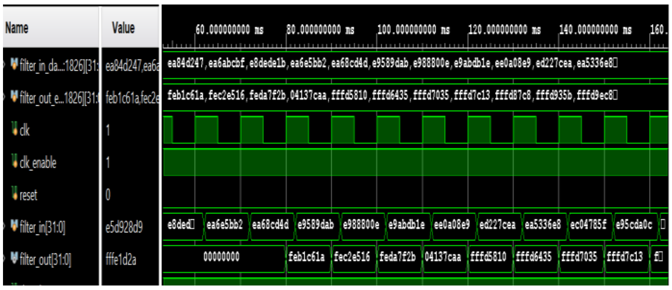

Figure 19: Simulation waveform of the Butterworth filter

7.2 Gaussian window filter

The coefficients got from the MATLAB follows the Gaussian expansion giving 15 coefficients. The obtained quantized coefficients are. 0.0063, 0.0144, 0.0290, 0.0515, 0.0805, 0.1107, 0.1340, 0.1428, 0.1340, 0.1107, 0.0805, 0.0515, 0.0290, 0.0144, 0.0063.The RTL implementation of the Gaussian window FIR filter is as shown below.

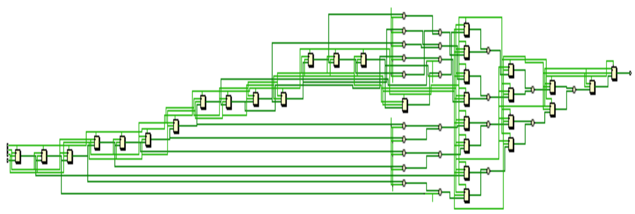

Fig. 20: RTL implementation of Gaussian window filter

The utilization summary of the Gaussian Window FIR filter is as shown below.

Table 5.Utilization summary of gaussian window filter.

Site type	used	available	Utilization%
Slice LUTs	921	63400	1.45
Slice Registers	1044	126800	0.82
DSP48E1	6	240	2.50
Bonded IOB	74	210	35.24
Clock(BUFGCTRL)	1	32	3.13

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7.3 Savitzky Golay Filter

The savitzky-Golay Filter performs operation on the basis of least square polynomial technique, so a mathematical polynomial generated on the basis of frame length and order is convolved with a signal therefore resulting in smoothing. In Savitzky-Golay filter the frame length is always in 2M+1 form, in this case as the frame length is 17 value of M will be 8, and the order N is 0. When the order N of the Savitzky-Golay filter is 0 it acts like a moving average filter by taking 2M+1 samples and it is also learned that lower order Savitzky-Golay filter and higher frame length or any one of them results in better smoothing and higher order and lower frame length or any one of them results in detail capturing i.e it will not smooth the signal it will capture even small distortions if present. So in the case of this project a moving average filter of 17 samples will be RTL implemented. The RTL implementation of the Savitzky-Golay Filter is as shown below.

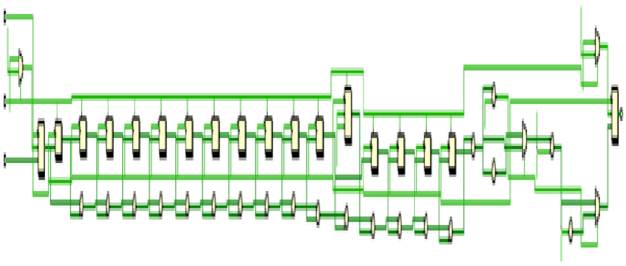

Fig. 22: RTL implementation of Savitzky Golay filter

The utilization summary of Savitzky-Golay Filter is given below.

Table 6.Utilization summary of Savitzky Golay filter.

Site type	used	Available	Utilization%
Slice LUT's	2145	63400	3.38
Slice Register	1527	126800	1.20
DSP48E1	5	240	2.08
Bonded IOB	89	210	42.38
Clock(BUFGCTRL)	1	32	3.13

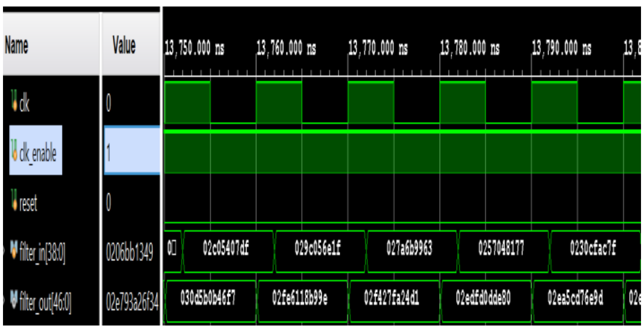

Fig. 23. Simulation waveform of Savitzky-Golay filter

7.4 The Final System

After the implementation of all of the subsystems they may be assembled and fashioned a final module. And here is the RTL Implementation.

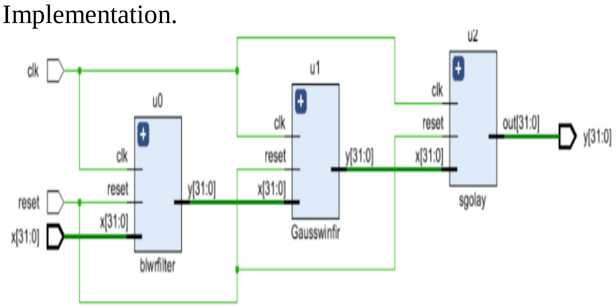

Fig. 24. RTL implementation of the final system

The utilization summary is shown below

Table 7. Utilization summary of the final system.

Site type	used	available	Utilization%
Slice LUT	7166	63400	11.30
Slice Register	2948	126800	2.32
DSP48E1	68	240	26.25
Bonded IOB	82	210	39.05
Clock(BUFGCTRL)	1	32	3.13

Simulation result

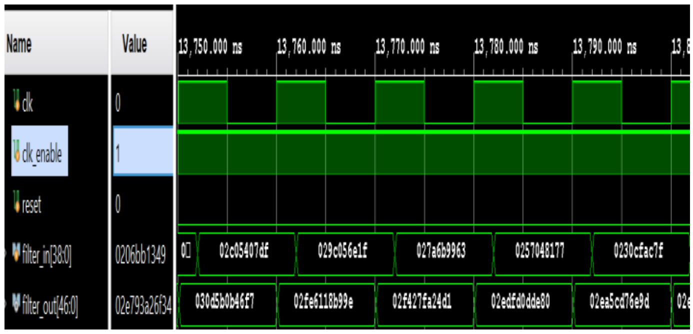

Fig. 25: Simulation waveform of the final system

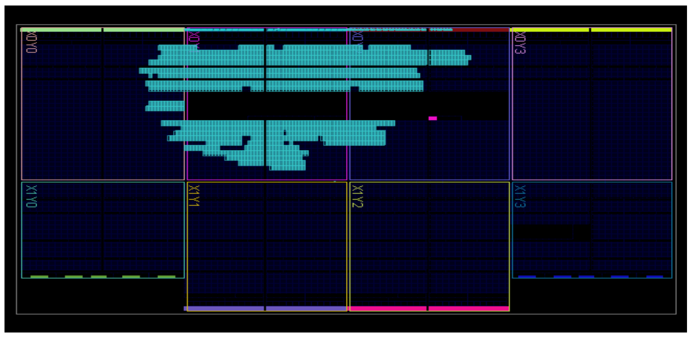

Fig. 26: Floor plan of the design

8. CONCLUSION

The noise corrupted ECG signal is handed through the Butterworth clear out for baseline wander removal due to it′s the flat-band reaction after the removal of baseline wander noise the excessive frequency noise corrupted signal then handed to Gaussian window filter due to its high amount of correlation value (COR = 0.957624) then to smooth the sign Savitzky-Golay filter do that task is chosen because of its high SNR cost (SNR =22.69 dB). The method used for denoising the ECG signal is the device obtained with the aid of cascading Butterworth clear out, Gaussian window filter, and Savitzky-Golay filte. It is giving a clean ECG sign with SNR value of 10.345474 dB and correlation value of 0.998486. The Implementation is done in the Artix-7 FPGA device having XADC and XDAC interface to convert the analog signal to a digital equivalent.

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