

Design and Implementation of a Digital FIR LPF with Variable Pass-Band for ECG Signal using FPGA

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Abstract: In this paper, we discuss a practical way to filter an ECG signal in the background of interference signals. This will be achieved by designing the low pass filter (LPF) with Kaiser window of variable pass-band (four pass-bands) of type finite impulse response (FIR) using a digital programmable device (Cyclone II EP2C70F896C6 FPGA from ALTERA) placed on education and development board DE2-70, this filter is designed by VHDL with Quartus II 9.1 design environment and consists of two filters (LPF) connected on serial. The proposed method depends on keeping FIR filters coefficients constant for all required pass-bands and changing the cut off and sampling frequencies together so that the frequency response remains constant for all pass-bands, just these pass-bands sweep on the frequency domain and thereby the pass-band of the filter will be changed without changing its coefficients. Using FPGA to design digital FIR LPF enable as of realization parallel processing to achieve digital convolution algorithm in the real time, where 1500 operation of adding, shifting, multiplying and dividing can be done in 0.5 ms (sampling period) for digital samples of 10 bits length, this equivalent 3 million operation per second where the general processor cannot do that.

Key words: Variable PASS-BAND, ECG, FPGA, LPF, FIR

I. INTRODUCTION

In all ECG devices several filters are used to change the filter pass-band for signal selection, thereby the filter specifications get worse because of limited resources.

For processing ECG signal, the flowing types of filters are used :

Low pass filter (LPF) for removes the high frequency noises in ECG signal:

LPF 50, 100, 150, 200Hz.

Mbachu C.B in [1] presented method of designing LPF with Kaiser window for constant pass-band. Seema Verma in [2] presented FIR filter using window techniques for ECG signal Processing for constant pass-band.

The re-tuning of digital filters lie in re-calculate filter coefficients and recording them in a constant memory. Usually the coefficients computing program is large and complicated (MATLAB11) and not implemented in the devices which contain the filters. Besides, storing the coefficients needs a constant memory of a large capacity that associated with the number and order of filters.

The re-tuning of digital filter by replacing the set of coefficients with another one is associated with transient cases results a distortion of output signal by a degree proportional to the filter quality, and to parameters changing step.

II. RESEARCH IMPORTANCE AND ITS OBJECTIVES

In this research, one digital filter was designed of type FIR LPF with several cut off frequencies 50, 100, 150, 200 Hz as shown in Fig.1, this is achieved through changing the cut off and sampling frequencies together and keeping the frequency response remains constant for all pass-bands,

without designing several filters. Besides a design of the ECG signal simulator was performed to test the proposed filter algorithm.

- Changing the sampling frequency is done by using DDS with high resolution specifications, so the filter coefficients remain constant for all required pass-bands.
- This algorithm allows highly abbreviation of filters number by required pass-bands number (four pass-bands) and increasing the attenuation factor of the interference signals and reducing the transition band width.
- The filter is designed with required specifications on the maximum pass-band ($F_{c \max}$) which relates to the maximum sampling frequency ($F_{\text{sam max}}$), then these specifications are achieved for all other pass-bands.
- This algorithm can be used for all filter types LPF, HPF, BPF, BSF and in all devices that require using digital filters (communication systems, radar systems, navigation systems, medical electronics systems...ets).

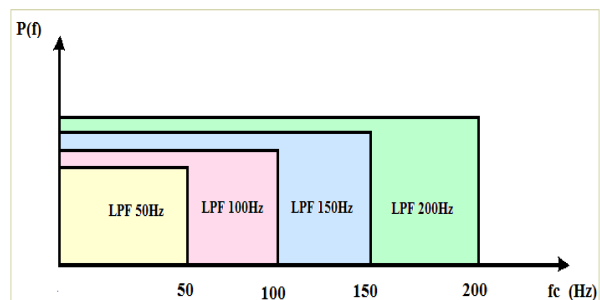


Fig.1 Digital filter pass-bands type FIR LPF

III. THE USED TOOLS AND SOFTWARE

- FPGA chips with highly accuracy, speed, and level specifications placed on education and development board DE2-70 [3].
- DDFS which is considered as highly resolution techniques in frequency synthesizing domain, and it was built in FPGA chips.
- FIR digital filters of Highly filtering specifications in accuracy and stability.
- VHDL programming language with Quartus II 9.1 design environment.
- MATLAB11 programming environment for digital filter design.
- GDS-1052-U digital oscilloscope with Free Wave program.
- PC computer for designing and injecting the design in the chip.

IV. THE BLOCK DIAGRAM OF THE PRACTICAL DESIGN

The block diagram of the practical design shown in Fig.2a and b shows the used practical platform using a digital programmable device (Cyclone II EP2C70F896C6 FPGA, Altera), which was placed on an education and development board (DE2-70, Terasic), and it consists of:

- Clock pulses generator $F_{CLK}=50\text{MHz}$ to give clock and sampling pulses for all blocks.
- Digital FIR LPF filter with pass-bands F_{c1} , F_{c2} , F_{c3} , F_{c4} with time convolution algorithm.
- Two DAC with 10bits to convert the signal from digital to analog form , before filtering (DAC1) and after filtering (DAC2).
- PC computer to link DE2-70 board via USB and inject the design in the Cyclone II EP2C70F896C6 FPGA chip [3].
- ECG simulator[4]with direct digital frequency synthesizer DDFS_ECG and DDFS_ Interference ,so that the synthetic ECG specifications would be closer to actual ECG signals.
- DDFS_ F_{sam} [4] to give sampling signals of frequencies F_{sam1} , F_{sam2} , F_{sam3} , F_{sam4} .
- Digital oscilloscope GDS-1052-U connected to PC via USB to show the input and output signals of the digital filters in time domain due to various interference signals.

This design is done for ECG signal and digital filter with the specifications below.

V. ECG SIGNAL SPECIFICATIONS

- The frequency range of the ECG signal is: first pass-band (0 to 50Hz), second pass-band (0 to 100Hz), third pass-band (0 to 150Hz), forth pass-band (0 to 200Hz).
- The processing on ECG signal frequency $F_{ECG}=0.5, \dots, 2\text{Hz}$ which represents 30, \dots, 120 BPM (Beats Per Minute) [5].
- Sampling frequencies for all components : $F_{sam}=500\text{ Hz}$, 1000 Hz ,1500 Hz ,2000 Hz ($T_{sam}=2, 1, 0.7, 0.5\text{ms}$).
- Number of samples: $M=1501$.
- The processing ward length for sampling ECG signal:

signed 10 bits.

- $\text{SNR}_{\text{inp}}=1/1$ for sinusoidal interference signals.

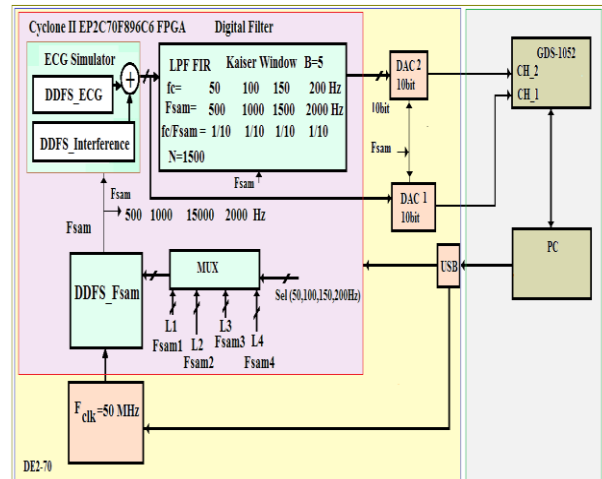


Fig. 2(a): The diagram of studying and searching procedure of ECG signal

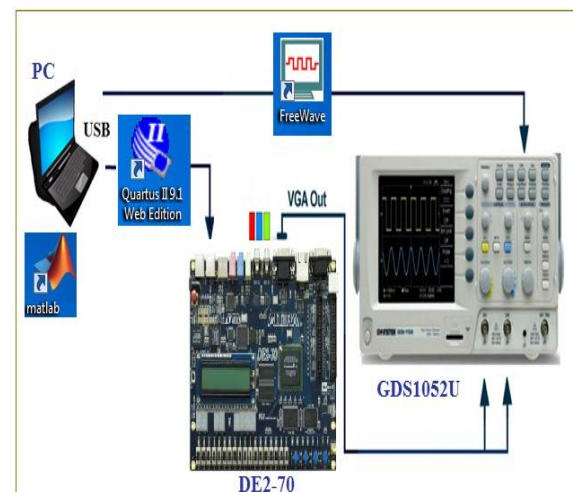


Fig. 2(b): The components of the practical platform (software and hardware) of studying and searching of ECG signal

VI. DIGITAL LPF SPECIFICATIONS

- Coefficients ward length: signed of 10bits.
- The number of used digital multipliers: 1501 multipliers of 10×10 bits.
- The number of the parallel shift registers with 10bits length is $1500SR (Z^{-1})$.
- One adder of 20bits with 1501 inputs and one output of 30bits.
- Different logical and mathematical operation elements (AND, NOT, XOR, \dots etc).
- The cut off frequencies of the filter are: $F_{c1}= 50\text{Hz}$, $F_{c2}= 100\text{Hz}$, $F_{c3}= 150\text{Hz}$, $F_{c4}=200\text{Hz}$.
- The sampling frequencies: $F_{sam1}= 500\text{Hz}$, $F_{sam2}= 1000\text{Hz}$, $F_{sam3}= 1500\text{Hz}$, $F_{sam4}= 2000\text{Hz}$.
- The ratio of cut off frequency to sampling one is: $F_{c1}/F_{sam1}=F_{c2}/F_{sam2}=F_{c3}/F_{sam3}=F_{c4}/F_{sam4}=1/10$.
- Window type: Kaiser with factor: $\beta = 5$.

- Filter order is $P=M-1=1500$.
- Stop-band attenuation factor is : (54dB).
- Maximum transition band width (Df_{max}) for F_{sam} max=2000 Hz [6] is:

$$P * \frac{Df_{max}}{F_{sam max}} = 3.2 \quad (1)$$

For one filter $\Rightarrow P * \frac{Df_{max}}{F_{sam max}} = 3.2 \Rightarrow$

$$Df_{max} = \frac{3.2 * F_{sam max}}{P} = \frac{3.2 * 2000}{1500} = 4.2Hz$$

For tow filters $\Rightarrow 2P * \frac{Df_{max}}{F_{sam max}} = 3.2 \Rightarrow$

$$Df_{max} = \frac{3.2 * F_{sam max}}{2P} = \frac{3.2 * 2000}{2 * 1500} = 2.1Hz$$

- Minimum transition band width (Df_{min}) for F_{sam} min=500Hz is:

For one filter $\Rightarrow P * \frac{Df_{min}}{F_{sam min}} = 3.2 \Rightarrow$

$$Df_{min} = \frac{3.2 * F_{sam min}}{P} = \frac{3.2 * 500}{1500} = 1.06Hz$$

For tow filters $\Rightarrow 2P * \frac{Df_{min}}{F_{sam min}} = 3.2 \Rightarrow$

$$Df_{min} = \frac{3.2 * F_{sam min}}{2P} = \frac{3.2 * 500}{2 * 1500} = 0.53Hz$$

- There is a possibility to develop this algorithm through serial connection for several filters to get high degree of attenuation and low transition band width.

VII. FILTERING ALGORITHM

The digital FIR filter output signal can be represented according to the following relationship [6]:

$$y(n) = h(n) * x(n) = \sum_{m=0}^{M-1} h(m).x(n - m) \quad (2)$$

Where: $x(n)$ input signal samples in digital form,
 m : the samples number of impulse response of the filter,
 n : the samples number of input and output signal ,
 $h(n)$:the impulse response for digital filter and it is given to LPF according the following relationship [6]:

$$h(0) = 2 \frac{F_c}{F_{sam}}$$

$$h(n) = \frac{1}{n\pi} \sin(2n\pi \frac{F_c}{F_{sam}}) \quad (3)$$

$$-\left(\frac{M-1}{2}\right) \leq n \leq \left(\frac{M+1}{2}\right) \quad (4)$$

Where:

F_c cut off frequency of the filter,

F_{sam} sampling frequency of the input signal.

For digital FIR LPF, its coefficients are computed in MATLAB programming environment, Fig.3 shows convolution algorithm of the FIR filter for input signal and impulse response samples of length M . Where Z^{-1} represents digital delay line of 10bits and a delay time equal to sampling pluses period T_{sam} .

To change pass-band of LPF, cut off frequency (F_c) of the filter must be changed, so the coefficients are changed, then designing new another filter is achieved.

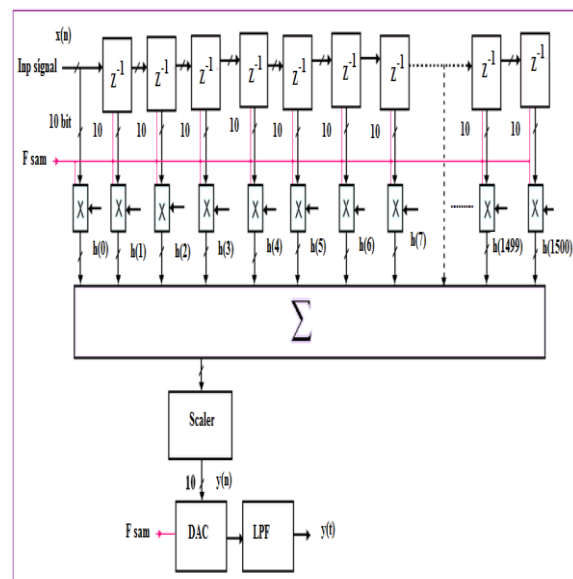


Fig. 3: Convolution algorithm $y(n)$ of the LPF for samples of input signal and impulse response of length $M=1501$

The digital FIR filter output signal with Kaiser window can be represented according to the following relationship [6]:

$$y_{Kaiser}(n) = y(n) * w(n) \quad (5)$$

Where:

$$w(n) = \begin{cases} I_0 \left\{ \beta \left[1 - \left(\frac{2n}{M-1} \right)^2 \right]^{1/2} \right\} / I_0(\beta), \\ \text{for } : -\left(\frac{M-1}{2}\right) \leq n \leq \left(\frac{M+1}{2}\right) \\ 0, \text{ elsewhere} \end{cases} \quad (6)$$

$$I_0(x) = 1 + \sum_{k=1}^L \left[\frac{(x/2)^k}{k!} \right]^2$$

Where typically $L < 25$ (7)

When $\beta = 0$, the Kaiser window rectangular window.
And when $\beta = 5.44$, the Kaiser Window Hamming window.
The value of β is determined by the stop-band attenuation requirements and may be estimated from one of the following empirical relationships:

$$\begin{aligned} \beta &= 0, \text{ if } A \leq 21 \text{ dB} \\ \beta &= 0.5842(A - 21)^{0.4} + \\ &0.07886(A - 21), \text{ if } 21 \text{ dB} \leq A < 50 \text{ dB} \\ \beta &= 0.1102(A - 8.7), \text{ if } A \geq 50 \text{ dB} \end{aligned} \quad (8)$$

Where $A = -20 \log_{10}(\delta)$ is the stop-band attenuation, $\delta = \min(\delta_p, \delta_s)$, since the pass-band and stop-band ripples are nearly equal.
The number of filter coefficients, P , is given by:

$$P \geq \frac{A - 7.95}{14.36 \Delta f} = \frac{A - 7.95}{(14.36 * Df) / F_{sam}} \quad (9)$$

Where $\Delta f = Df / F_{sam}$ is the normalized transition band width:

$$\begin{aligned} \text{For } F_{sam} &= 2000 \text{ Hz and } Df = 4.2 \text{ Hz} \Rightarrow \\ P &\geq \frac{A - 7.95}{14.36 \Delta f} = \frac{52 - 7.95}{14.36 * 4.2 / 2000} = 1461 \Rightarrow P = 1500 \\ \text{For } F_{sam} &= 500 \text{ Hz and } Df = 1.06 \text{ Hz} \Rightarrow \\ P &\geq \frac{A - 7.95}{14.36 \Delta f} = \frac{52 - 7.95}{14.36 * 1.06 / 500} = 1447 \Rightarrow P = 1500 \end{aligned}$$

$$\begin{aligned} \text{For } A &= 54 \text{ dB} \Rightarrow \\ \beta &= 0.1102(A - 8.7) = 0.1102 * (54 - 8.7) = 5 \end{aligned}$$

A typical Kaiser window function for $P=1500$, $\beta=5$, and $A=54$ dB is depicted as Fig.4 below.

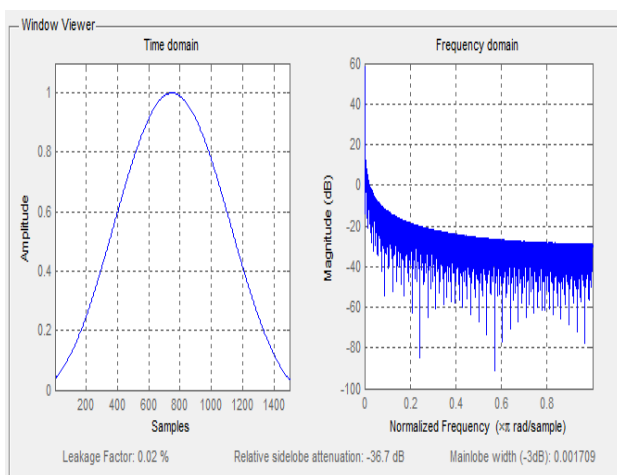


Fig. 4: Typical Kaiser window function for $P=1500$, $\beta = 5$, and $A=54$ dB

VIII. THE PROPOSED MATHEMATICAL ALGORITHM FOR FILTERING

The proposed algorithm depends on changing the cut off and sampling frequencies together where the ratio between them remains constant for the filter, so the filter coefficients remain constant, while the frequency response curve sweeps on the frequency domain according the required pass-band. We note from (3) when F_c changes, the coefficients $[h(0), h(n)]$ change, and then the filter pass-band changes and this requires a new memory to store the new coefficients and this applied on each new pass-band. In the proposed algorithm, the cut off frequency (pass-band) and sampling frequency are changed where the following ratio remains constant:

$$\begin{aligned} F_{ci} / F_{sam i} &= const \Rightarrow \\ [h_i(0) &= const, h_i(n) = const] \end{aligned} \quad (10)$$

Then the coefficients $[h(0), h(n)]$ remain constant for all pass-bands and this achieve changing the filter pass-band through changing the sampling frequency.
In this research, a LPF is designed with four pass-bands:

$$(F_{c1}= 50 \text{ Hz}, F_{c2}= 100 \text{ Hz}, F_{c3}= 150 \text{ Hz}, F_{c4}=200 \text{ Hz})$$

And four sampling frequencies:

$$(F_{sam1}= 500 \text{ Hz}, F_{sam2}= 1000 \text{ Hz}, F_{sam3}= 1500 \text{ Hz}, F_{sam4}=2000 \text{ Hz})$$

Where:

$$\begin{aligned} \frac{F_{ci}}{F_{sam i}} &= const = c, \quad i = 1 \dots 4 \\ \frac{F_{ci}}{F_{sam i}} &= \frac{F_{c1}}{F_{sam 1}} = \frac{F_{c2}}{F_{sam 2}} = \frac{F_{c3}}{F_{sam 3}} = \frac{F_{c4}}{F_{sam 4}} \\ &= \frac{1}{10} = const = c, \quad i = 1 \dots 4 \\ \frac{F_{ci}}{F_{sam i}} &= \frac{50}{500} = \frac{100}{1000} = \frac{150}{1500} = \frac{200}{2000} \\ &= \frac{1}{10} = const = c, \quad i = 1 \dots 4 \end{aligned}$$

To change the sampling frequency, a direct digital frequency synthesizer is used because of its high resolution specifications.

This filter is designed by VHDL [7] with Quartus II 9.1 design environment and consists of two filters (LPF) connected on serial according to the block diagram shown in Fig.5.

This algorithm is useful for all digital filters that have the responses: LPF,HPF,BPF,BSF and for types IIR, FIR, with a condition that the filter order is constant for all required pass-bands.

The filter specifications in the MATLAB11 programming environment are shown in Fig.6 for $F_{c1}=50 \text{ Hz}, F_{sam1}=500 \text{ Hz}$, and in Fig.7 for $F_{c4}=200 \text{ Hz}, F_{sam4}=2000 \text{ Hz}$, where we note that the frequency response curve is constant and coefficients are

constant, also this situation is achieved for other pass-bands which have $F_c/F_{sam}=1/10$.

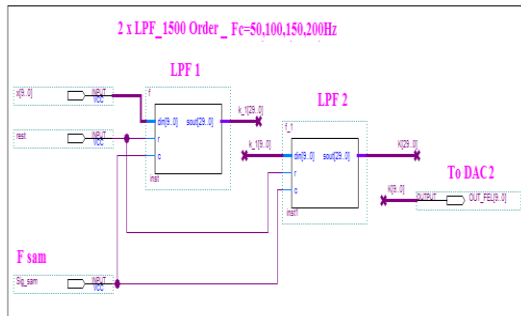


Fig.5: Digital LPF block diagram

A typical impulse response FIR LPF with Kaiser window for $P=1500$, $\beta = 5$, is depicted as Fig. 8 below.

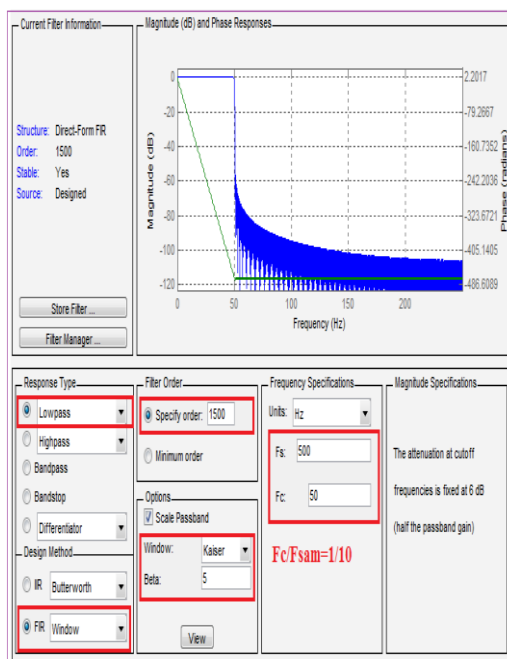


Fig. 6: Filter specifications for pass-band 50Hz , $F_{c1}=50\text{Hz}, F_{sam1}=500\text{Hz}$

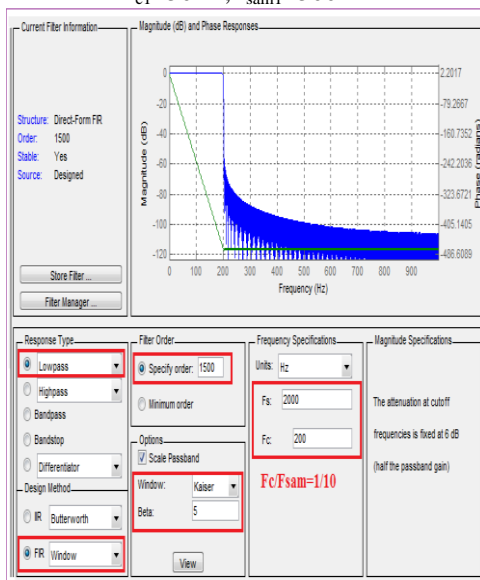


Fig.7: Filter specifications for pass-band 200Hz, $F_{c4}=200\text{Hz}, F_{sam4}=2000\text{Hz}$

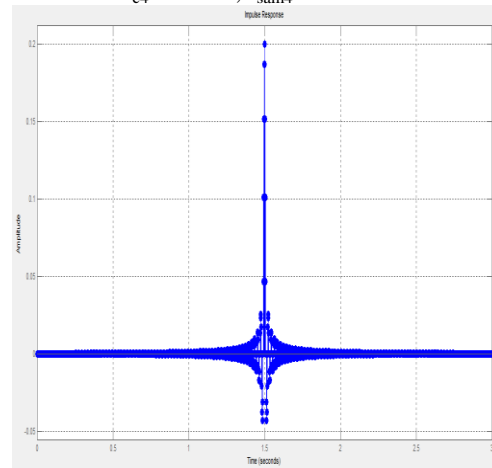


Fig.8 :Impulse response FIR LPF with Kaiser window for $P=1500$ and $\beta = 5$

IX. THE PRACTICAL RESULTS

The practical design results in time domain for input and output signals of the LPF digital filter were taken by digital oscilloscope of type GDS-1052U.

Fig.9 shows on channel 1 of the oscilloscope the ECG signal without the interference effect before filtering and on channel 2 the ECG signal after filtering by digital LPF .This ensures that the filter must not distort the signal after filtering because it has a linear phase response.

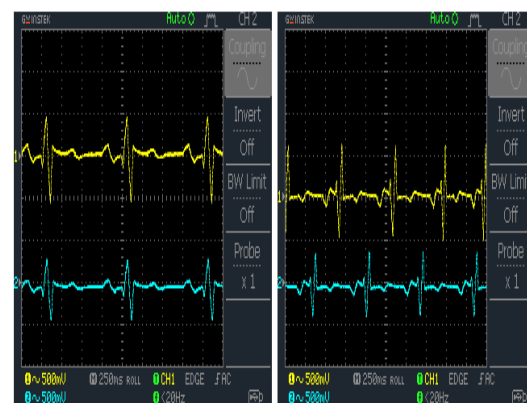


Fig. 9: Input and output signal of the digital LPF without interference signal



Fig.10: Input and output signal of the LPF of pass-band 50Hz in case of interference signal of frequency 52Hz

Fig.10 shows on channel 1 of the oscilloscope the ECG signal with interference signals of frequency 52 Hz ,to test the digital LPF of pass-band 50Hz, before filtering and on channel 2 the ECG signal after filtering by digital LPF ,we note the possibility of removing the interference signals completely.

Fig.11 shows on channel 1 of the oscilloscope the ECG signal with interference signals of frequency 102 Hz ,to test the digital LPF of pass-band 100Hz, before filtering and on channel 2 the ECG signal after filtering by digital LPF, we note the possibility of removing the interference signals completely.

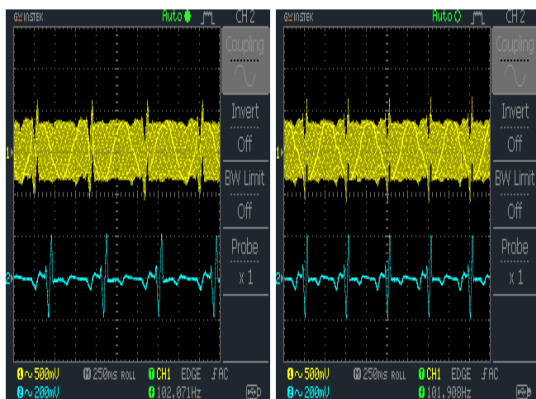


Fig. 11: Input and output signal of the LPF of pass-band 100Hz in case of interference signal of frequency 102Hz

Fig.12 shows on channel 1 of the oscilloscope the ECG signal with interference signals of frequency 152 Hz ,to test the digital LPF of pass-band 150Hz, before filtering and on channel 2 the ECG signal after filtering by digital LPF, we note the possibility of removing the interference signals completely.

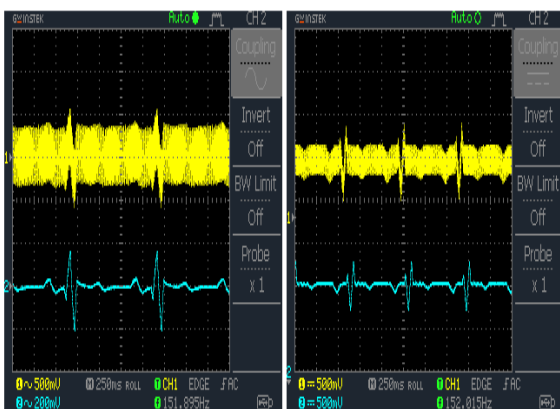


Fig. 12: Input and output signal of the LPF of pass-band 150Hz in case of interference signal of frequency 152Hz

Fig.13 shows on channel 1 of the oscilloscope the ECG signal with interference signals of frequency 202 Hz ,to test the digital LPF of pass-band 200Hz, before filtering and on channel 2 the ECG signal after filtering by digital LPF ,we note the possibility of removing the interference signals completely and all cases with factor $SNR_{inp}=1/1$.

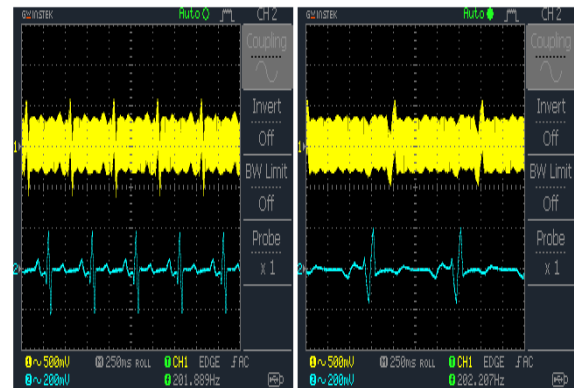


Fig. 13: Input and output signal of the LPF of pass-band 200Hz in case of interference signal of frequency 202Hz

X. CONCLUSION

From the practical results shown on Fig. (10 to 13) we note the possibility of removing the interference signals for multiple pass-bands within the frequency ranges: (0 to 50Hz), (0 to 100Hz), (0 to 150Hz), (0 to 200Hz), that for one filter of constant coefficients, but of variant sampling frequency according to the required pass-band, and of $SNR_{inp}=1/1$ for the sinusoidal interference signals. This means that the four filters of the intended pass-bands is replaced with one filter in the executed design. This algorithm can be applied for all filter kinds LPF, HPF, BPF, BSF and for filters FIR, IIR and for a large number of pass-bands and multiple ranges within the definite conditions and in all devices that require using digital filters (communication systems, radar systems, navigation systems, medical electronics systems...ets). To develop this research in the future , an ECG signal can be taken from any person for diagnosis and implement the filtering by using the proposed filter.

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