iJRASET
International Journal For Research in Applied Science and Engineering Technology



INTERNATIONAL JOURNAL FOR RESEARCH

IN APPLIED SCIENCE & ENGINEERING TECHNOLOGY

Volume: 7 **Issue:** IV **Month of publication:** April 2019

DOI: https://doi.org/10.22214/ijraset.2019.4267

www.ijraset.com

Call: 08813907089 | E-mail ID: ijraset@gmail.com

International Journal for Research in Applied Science & Engineering Technology (IJRASET)
ISSN: 2321-9653; IC Value: 45.98; SJ Impact Factor: 6.887
Volume 7 Issue IV, Apr 2019- Available at www.ijraset.com

Designing a Moving Average FIR Filter for Acquiring Heart Murmur

S.Sumathi[1], Dr. R. Dhaya[2], Shelina Jonapeter[3], Aishwarya. AJ[4]
[1] *Assistant professor, Velammal engineering college*
2 Associate Professor, Dept. of CSE, Rajalakshmi Engg College, Chennai
[2] *UG student, Velammal engineering college*

Abstract: Digital filters are widely used in many digital signal processing applications. Therefore, digital filtering is one of the basic needs of digital signal processing. This paper introduces the definition and basic principles of FIR digital filters, and the design based on MATLAB. After the description of the process of design, the MATLAB program is used to implement FIR filter using modified coefficient of hamming window function and also calculate the equivalent noise bandwidth.
Keywords: FIR filters, hamming window, hanning window, modified hamming window, ENBW (Equivalent noise bandwidth).

I. INTRODUCTION

Recent emerging medical applications such as electrocardiography (ECG), electro encephalography (EEG) and many biosignal measuring instruments require DSP processing performance at very low power. The digital signal processor (DSP) is ideally suited for such applications.
A typical medical application includes:

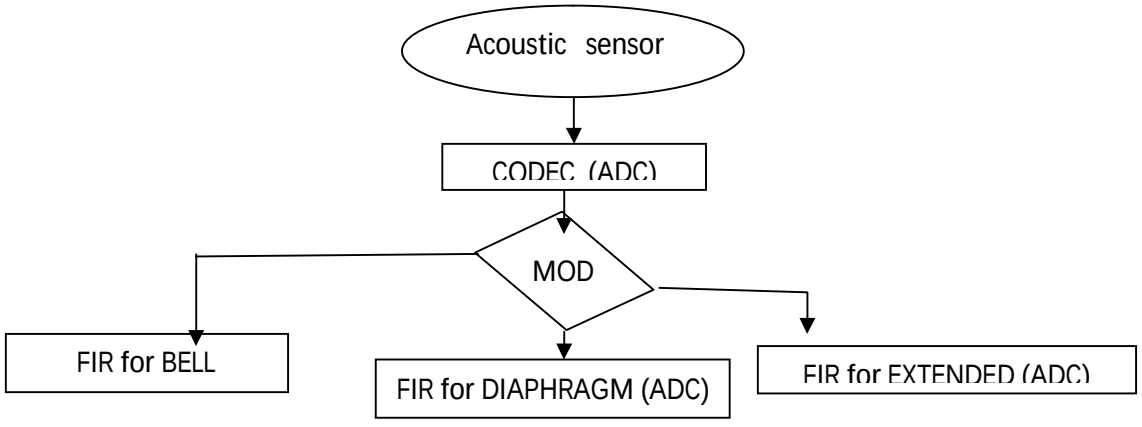
A. Bio-sensors which are analog in nature used as front end to pick up signals from the human body
B. To identify and study the health condition of the patient, we used special signal processing algorithms for analyzing the acquired measurements from the sensors and conditioned the signals for further processing
C. By the use of signal processing and IoT we can able to monitor patient from remote areas

II. DIGITAL STETHOSCOPE

Heart and Lung sounds are measured as audio output in three modes as Bell mode (20 Hz to 220 Hz), Diaphragm mode (50 Hz to 600 Hz) ,Extended mode (20 Hz to 2000 Hz). Heart rate in bell and diaphragm mode is displayed on liquid crystal display panel. In hospitals, acquired heart sound is displayed as waveform on the screen. We can able to store the heart sound waveform and zoom and playback of the same on the PC for future days.

III. DATA ACQUISITION

The analog front end mainly has three parts Diaphragm, microphone , audio plug. The acquired sound waves from the diaphragm (acoustic amplifier) are fed to the condenser microphone. The condenser microphone works under the principle of change its capacitance by changing its impedance, which produces a voltage swing proportional to the amplitude of the input sound waves. The voltage swing of the signal also depends on the bias voltage given for the microphone. The coupling of the acoustic sensor to the microphone is critical to pick up noise free sound signals from the human body.



©IJRASET: All Rights are Reserved

ISSN: 2321-9653; IC Value: 45.98; SJ Impact Factor: 6.887
Volume 7 Issue IV, Apr 2019- Available at www.ijraset.com

Based on the mode selected, digital signals from the front-end board are fed to the corresponding FIR filter. FIR BPF (Hamming window) filter with order 161 is used to remove noise and unwanted signals . The filtered output is fed back to the codec on the front-end board. Filtered output is transmitted over UART interface for display and storage. Heart beat detection algorithm is applied for bell and diaphragm modes. Heart rate, operating mode, and volume level are displayed on the LCD screen.

IV. DIGITAL FILTER

A digital filter takes a digital input, gives a digital output, and consists of digital components. In a typical digital filtering application, software running on a digital signal processor (DSP) reads input samples from an A/D converter, performs the mathematical manipulations dictated by theory for the required filter type, and outputs the result via a D/A converter. The digital stethoscope has three different finite impulse response (FIR) filters implemented for the three operating modes. FIR filters are preferred over IIR filters, because FIR digital filters have a linear phase, highly stable, non-recursive structure and arbitrary amplitude frequency characteristic etc. FIR filter designed using modified coefficient of the Hamming window function provides smaller main lobe width and sharp transition band compare to Hamming window. The filter being used is the FIR hamming window band pass with order of 161, which provides a sharp cutoff with attenuation of about 50dB. The sampling frequency is 12000 samples / second.

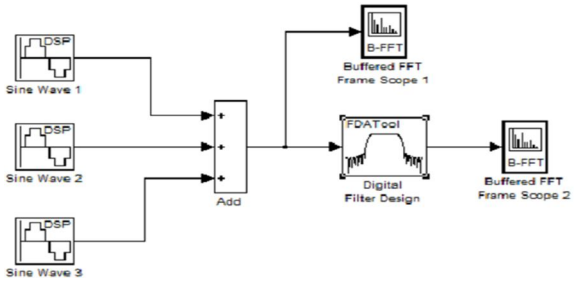


Fig.1 simulation of FIR filter with hamming window

Buffer shifting convolution algorithm is used for the realization of the filter. This type of filter is very useful in spectral analysis of different types of signals. The windowing of a signal in time also affects the resolution bandwidth that can be achieved. When calculating the spectrum of a signal segment the resolution bandwidth achieved with a window is always lower than the resolution bandwidth achieved without a window The simulation results show that the filter designed using modified window function is more efficient

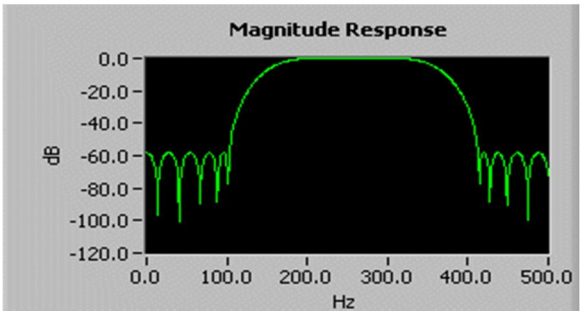


Fig.2 Magnitude response of FIR filter with hamming window

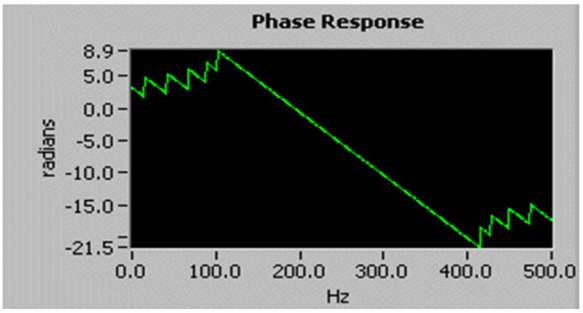


Fig.3 Phase response of FIR filter with hamming window

©IJRASET: All Rights are Reserved

Transition bandwidth describes how quickly a filter transitions from a passband to a stopband, or vice versa. The more rapid this transition, the higher the transition bandwidth; and the more difficult the filter is to achieve. Though an almost instantaneous transition to full attenuation is typically desired, real-world filters don't often have such ideal frequency response curves.

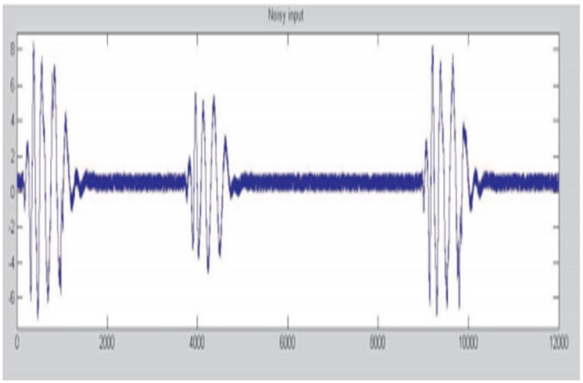


Fig.1 Bell mode Signal before filtering

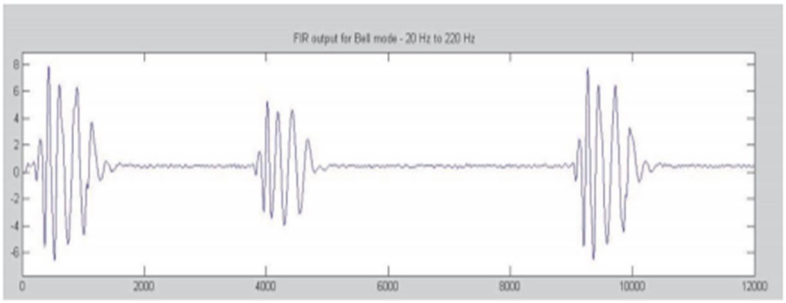


Fig.2 Bell mode Signal after filtering

Heart Beat Detection Heart rate detection algorithm has the following steps: 1. Smoothen the FIR filter output by using the 5 tap moving average filter. 2. Detect the S1 (first heart sound) by calculating the maximum slope of the smoothened waveform. 3. Disable the S1 detection logic for 100 msec. 4. Measure the number of samples between two consecutive S1's. 5. Calculate the heart rate using the following formula: (a) HR = ((Sampling Rate * 60)/Number of samples between two consecutive S1's) (b) Where, sampling rate = 12000 Hz 6. Apply eight data point moving average logic to compute the average heart rate.

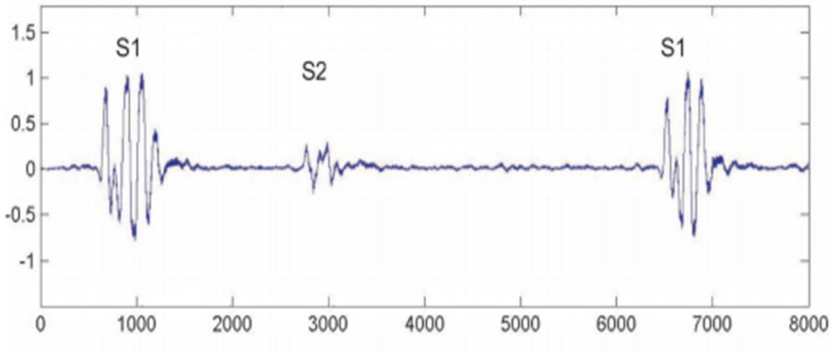


Fig.3 moving average FIR filter output

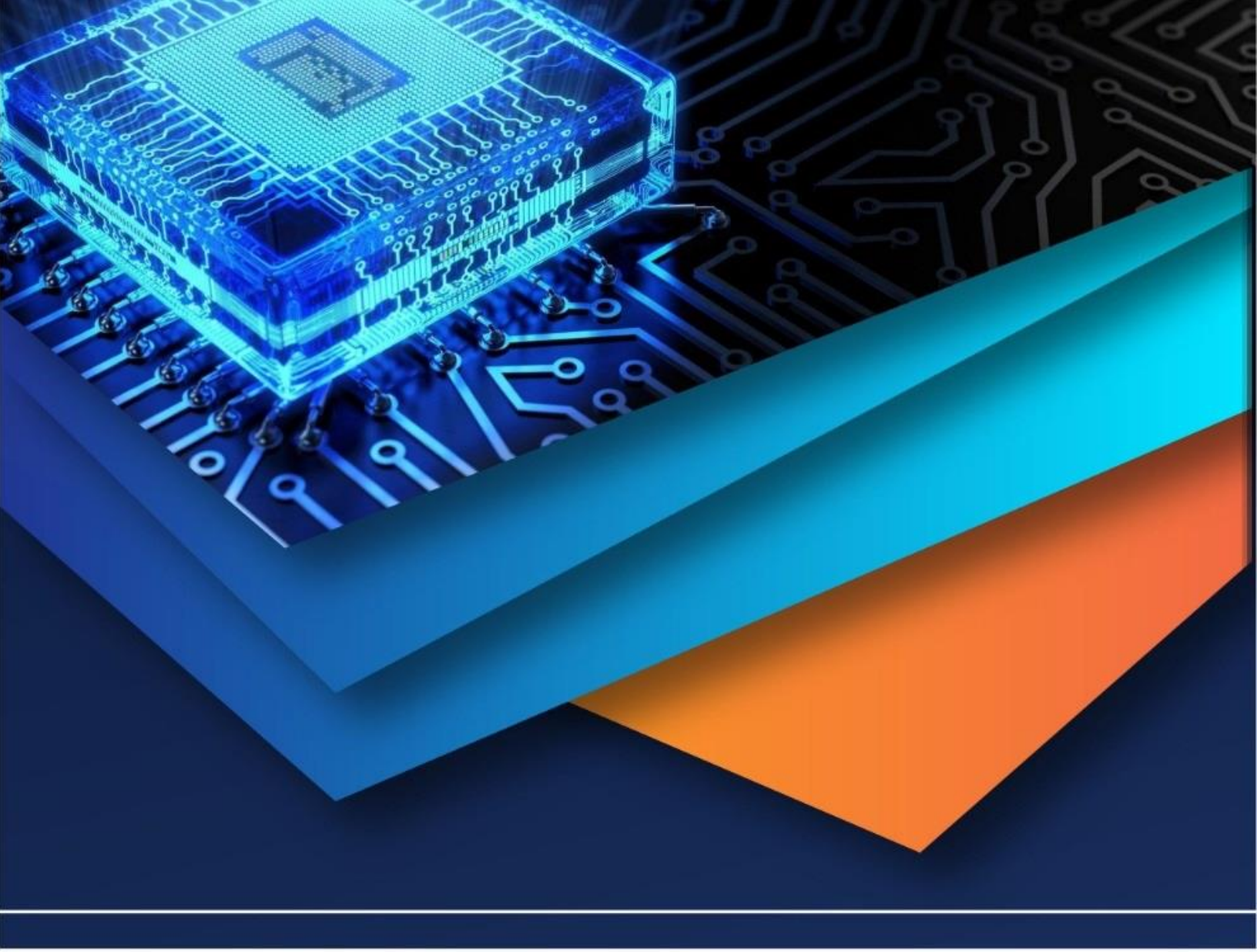
The moving average filter is implemented using a recursive algorithm. The order of the moving average filter used is 5. The following equation shows the moving average window: Where M – order of filter x[i] – i th input sample y[i] – i th filtered output UART Transmission The filtered output (at data rate 12000 sps) is decimated to 3000 sps and sent to UART for transmission. Decimation is implemented by averaging four samples. The following parameters are sent over UART to the PC application every one sec. • Heart rate • Operating mode • Volume The UART configuration that is used is 115200 bps, 8 data bits, 1 stop bit and no parity

©IJRASET: All Rights are Reserved

REFERENCES

[1] Oppenheim, R. Schafer, and J. Buck, "DiscreteTime Signal Processing "second edition, PrenticeHall,1999.
[2] J.G.Proakis and D.G.Manolakis, "Digital Signal Processing Principles, Algorithms and Applications" third edition Prentice-Hall, 2002.
[3] Sanjit K. Mitra, "Digital Signal Processing: A computer-base approach", Tata McGraw-Hill, 2nd Ed, 2001.
[4] T. Saramaki, "Finite impulse response filter design," in Handbook for Digital Signal Processing, Edited by S. K. Mitra and J. F. Kaiser, IBSN 0-471-61995-7 John Wiley &Sons Inc, 1993.
[5] Sonika Gupta, Aman Panghal Performance "Performance Analysis of FIR Filter Design by Using Rectangular, Hanning and Hamming Windows Methods" International Journal of Advanced Research in Computer Science and Software Engineering Volume 2, Issue 6, June 2012.
[6] Prof.Gopal S.Gawande, Dr.K.B.Khanchandani, T.P.Marode "performance analysis of fir digital filters design techniques" International Journal of computing & corporate research volume 2 issue 1 January 2012.
[7] Najat Sh.Jasim Mohammed "The Performance Enhancement Study of FIR Filters Based on Adjustable Window Function" Engg & Tech Journal vol.30, no.5, 2012.
[8] GAO Jinding, HOU Yubao, SU Long. "Design and FPGA Implementation of Linear FIR Low-pass Filter Based on Kaiser Window Function" Fourth International Conference on Intelligent Computation Technology and Automation in proceeding IEEE pp.496-498, 2011.
[9] Yunlong,W., Shihu,W., Rendong,J., "An Extreme Simple Method for Digital FIR Filter Design" in proc. Third International Conference on Measuring Technology and Mechatronics Automation (ICMTMA), IEEE, vol. 1, pp. 410-413, 2011.
[10] Mahrokh G. Shayesteh and Mahdi MottaghiKashtiban "FIR filter design using a new window function" in proceeding IEEE 978-1-4244-3298- 1/09,2009.
[11] S.Salivahanan, A.Vallavaraj, C. Gnanapriya, "Digital Signal Processing", Tata McGraw-Hill, 2000.
[12] Andreas Antoniou, "Digital filters analysis, Design and applications", Tata Mc Graw-Hill, 2nd Ed. 1999.

©IJRASET: All Rights are Reserved



10.22214/IJRASET

45.98

IMPACT FACTOR: 7.129

IMPACT FACTOR: 7.429

INTERNATIONAL JOURNAL FOR RESEARCH

IN APPLIED SCIENCE & ENGINEERING TECHNOLOGY

Call : 08813907089 (24*7 Support on Whatsapp)