FIR Filter Designing using MATLAB Simulink and Xilinx system Generator

KUMUDINI SAHU[1], RAHUL SINHA[2]

[1] ME Scholar, ET & T, CSIT Durg, C.G., India
[2] Assistant Professor, ET & T, CSIT Durg, C.G., India

---***---

Abstract - *Xilinx System generator is used to design efficient DSP algorithm on FPGA. FIR Filter is very important type of Digital Filters which is a vital element in Digital Signal Processing. FIR filter is a type of digital system that filters discrete-time signal and the main signal, main objective performing frequency domain filtering by processing sample data. It is use in various applications like Speech recognition, Speech synthesis, digital audio, Telecommunication, seismic signal processing (noise elimination), and several other areas of signal processing. This paper presents the Finite Impulse Response (FIR) filter designing using Xilinx System generator blockset in MATLAB Simulink. The design of 19-Tap FIR filter using MATLAB Simulink and Xilinx system generator for removal of noise from input signal. The implemented structure of 19 tap FIR filter uses the input signal with including noise. We designed structure uses sampling frequency of 100 Hz, pass band frequency of 0.58 Hz while the stop band frequency is of 2 KHz. This is a single rate FIR filter structure for 19 tap. The main objective of this designed filter is to reduce the noise from given signal. This Filter is designed using the FDA tool with 1 dB pass band attenuation & 60 dB stop band Attenuation. This FIR Filter is designed with the help of MATLAB SIMULINK (Win_2013) and XILINK System Generator (ISE_Win_14.4).*

Key Words: FIR filters, MATLAB SIMULINK, XILINX System Generator, DSP, Digital Filter and FDA Tool.

1. INTRODUCTION

Digital signal processing techniques are used extensively in a number of applications such as communication and multimedia. DSP functions such as FIR filters and transforms have numerous advantages over their analog counter parts. Digital circuits are not dependent on precise values of digital signals for their operation. Digital circuits are less sensitive to changes in component values. They are also less sensitive to variations in temperature, ageing and other external parameters. Digital processing of a signal facilitates the sharing of a single processor among a number of signals by time-sharing. This reduces the processing cost. In addition multi-rate processing is possible only in digital domain. Storage of digital data is very easy.

Digital filters are useful structures for digital signal processing applications and in signal analysis and estimation [1]. Digital filters are widely used in the world of communication and computation. An operation of digital filter design is calculation of filter transfer function coefficients that decide the response of the filter. Typical filter applications include signal preconditioning, band selection, and low/high pass filtering. Digital filters are categorized as finite impulse response (FIR) and infinite impulse response (IIR) filters. Although FIR filters are more complex, they have certain advantages over IIR filters due to which they are more widely used in filtering applications. IIR filters do not provide stability at higher orders whereas the FIR counterparts are always stable and are particularly useful for applications where exact linear phase response is required.

2. FINITE IMPULSE RESPONSE (FIR) FILTER

In signal processing, a finite impulse response (FIR) filter is a filter whose impulse response (or response to any finite length input) is of *finite* duration, because it settles to zero in finite time. The impulse response of an Nth-order discrete-time FIR filter lasts for $N + 1$ samples, and then settles to zero[1]. A Finite impulse response (FIR) filter is a filter structure that can be used to implement almost any sort of frequency response digitally. An FIR filter is usually implemented by using a series of delays, multipliers and adders to create the filter output. Figure 1 shows the basic block diagram for an FIR filter of length N. The delays in operating on prior input samples. The h_K values are the coefficients used for multiplication, so that the output at time n is the summation of all the delayed samples multiplied by the appropriate coefficients.

The processer selecting the filters length and coefficients is called as filter design.

3. DESIGN IN SIMULINK

Simulink is a block diagram environment for multi-domain Simulation and Model-Based Design. In Simulink, we can design the filter in the Simulink with the automatic generation of codes, simulation and continuous test. In MATLAB, there are the customizable block libraries and solvers for modelling and simulating dynamic systems. It can make the blocks and exports simulation results to MATLAB for the further analysis. The blocks which are required are discussed below.

1. The *'Xilinx system generator'* [23] is a high level tool for designing high-performance DSP systems using FPGAs. The system generator tool enables us to integrate Xilinx with Simulink; it creates a .ISE file which is used in Xilinx using the model file of Simulink.

2. Xilinx block sets works only in the gateway blocks, i.e. gateway-in and gateway-out. Any sample based input is given to the gateway-in block; the output can be seen on the scope by passing through the gateway out block.

3. FDA Tool is the important tool of MATLAB which is used to design the filter of required specification. There are different responses (i.e. High pass, Low pass, Band pass, Band stop, Differentiator, Integrator, etc) and Design method (i.e. IIR, FIR) for implementing the filter. By placing the filter order, frequency specifications and magnitude specifications, windows can be customized. Tools create coefficients in the MATLAB workspace in matrix form through the specifications provided.

4. IMPLIMENTATION OF FIR FILTER
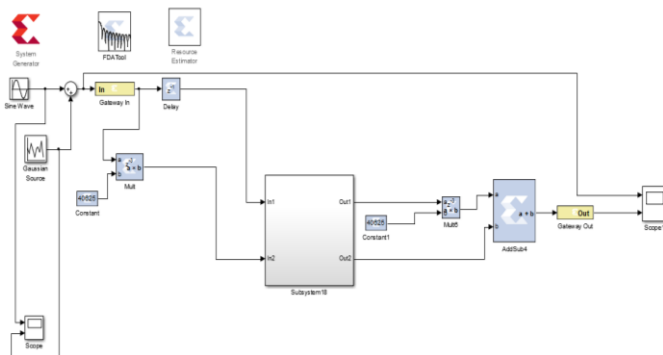
The 19-tap FIR filter can be designed as below,

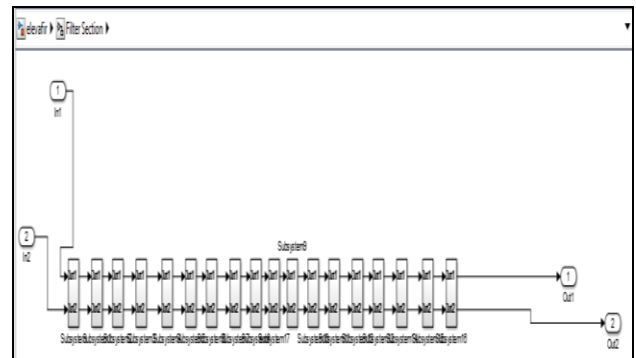**Figure 1:** 19-tap FIR Low pass Filter

Filter Subsystem

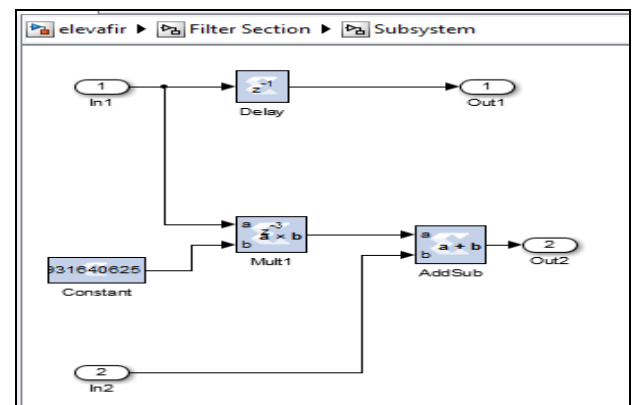

Figure 2: FIR Filter subsystem

Subsystem

Figure 3: FIR Filter, filter section subsystem

Filter Response

The magnitude response of 19 order FIR filter in which pass band frequency is 0.58 Hz and stop band frequency is 2 Hz can be observed in figure 4.
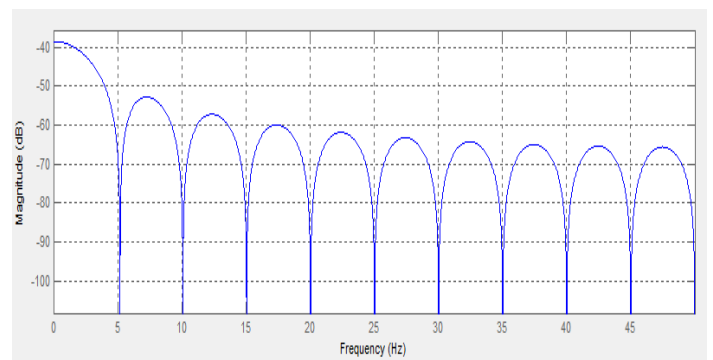

Figure 4: Magnitude response of FIR filter (N=19)

5. SIMULATION AND RESULTS

The 19-tap FIR filter is designed. We have got all the specifications. The sampled input signal and the noise displayed on the scope is shown below
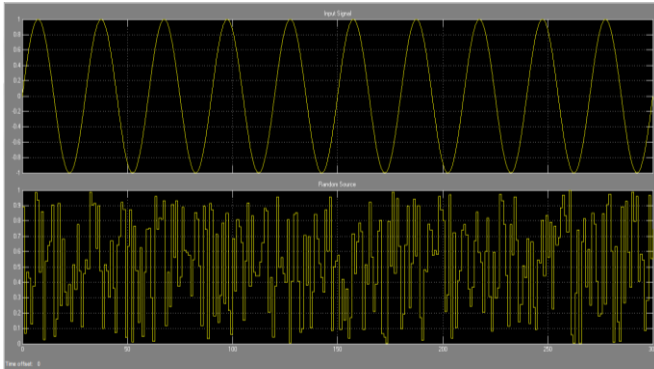

Figure 4: 19-tap FIR Low-pass filter input signal

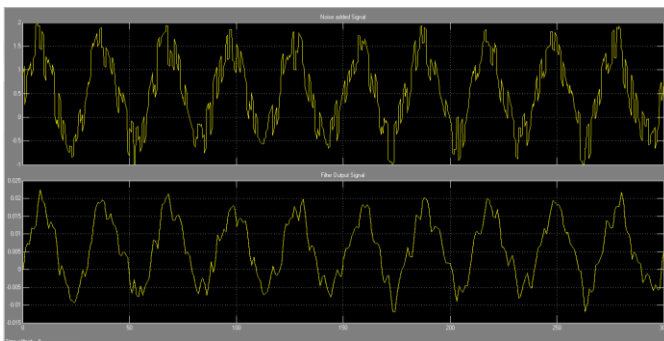The noisy input signal and the output filtered signal obtained is shown below,

Figure5: Noisy Input signal, Filtered Output Signal

6. CONCLUSIONS

Adaptive Noise Cancellation is an alternative way of cancelling noise present in a corrupted signal. The principal advantage of the method is in its adaptive capability, its low output noise, and its low signal distortion .It has been proved that the proposed NLMS algorithm gives better error performance. The implementation and simulation of Adaptive LMS filter using NLMS algorithm have been done using MATLAB Simulink environment and their response have been studied in waveform in the given simulation results.

FIR Filter based on Xilinx Block-set model has been proposed with the coefficient generated from the MATLAB Simulink model. Through regressive simulation of MATLAB Simulink model it has been found that filter length 19 provides good noise cancellation for NLMS algorithm, of FIR Filter. The system works good in the MATLAB as well as in Xilinx environment. The resource utilized for the device were 1077-slices, 1584-FFs, 32-IOBs Blocks and of 20-Mulipliers/DSP blocks.

REFERENCES

[1] Chandak, P.R., Giradkar, V.P & Wadmalwar, A.T., (February 2015) "Design of FIR filter using Matlab Simulink and Xilinx system Generator" *International Journal Of Engineering And Computer Science* ISSN:2319-7242 Volume 4 Issue 2 Page No. 10384-10387.

[2] Kamble, P., Deote, N., Wanjari, N., & Prof. Gaikwad, S., (January 2015) "Implementation of FIR Filter Structure for Audio Application Using Xilinx System Generator" *International Journal of Advanced Research in Computer Science and Software Engineering* Volume 5, Issue 1, ISSN: 2277 128X.

[3] Dewasthale, M.M. & Dr. Kharadkar ,R.D., (2014) "Improved NLMS Algorithm with Fixed Step Size and Filter Length using Adaptive Weight Updating for Acoustic Noise Cancellation" *Annual IEEE India Conference (INDICON)978-1-4799-5364-6/14/*.

[4] Maurya, L., Srivastava, V. K., & Mehra, R., (Feb-2014) "Simulink based design simulations of band pass FIR filter" *IJRET: International Journal of Research in Engineering and Technology* eISSN: 2319-1163 | pISSN: 2321-7308 Volume: 03 Issue: 02 |.

[5] Gaikwad, S.R. & Dekate, K.N.,(Oct.-Nov.2014)"Design of FIR Filter for Audio Application" *International Journal of Emerging Trends in Engineering and Development* Issue 4, Vol.6 ISSN 2249-6149.

[6] Gaikwad, S.R., & Gawande, G. S., (2014) "Design and Implementation of Efficient FIR Filter Structures using Xilinx System Generator" *International Journal of scientific research and management (IJSRM)* Volume-2,Issue-3,Pages-599-604.

[7] Rao, C.M., Dr. Charles, B. S. & Dr. Prasad, M.N.G., (July 2013) "A Variation of LMS Algorithm for Noise Cancellation" International *Journal of Advanced Research in Computer and Communication Engineering*, Vol. 2, Issue 7, ISSN: 2319-5940.

[8] Aggarwal, A.,Satija A., & Nagpal T., (April 2013) "FIR Filter Designing using Xilinx System Generator", *International Journal of Computer Applications* (0975 – 8887), Volume 68– No.11,.

[9] Patel, S., (October 2013) "Design and Implementation of 31-order FIR Low-pass Filter using Modified Distributed Arithmetic based on FPGA" *International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering* Vol. 2, Issue 10, ISSN: 2320 – 3765.

[10] Kumar, T. & A.,SMIEEE,(2013) "Low Power VLSI Implementation of Adaptive Noise Canceller Based on Least Mean Square Algorithm" *International*

*Conference on Intelligent Systems, Modeling and Simulation, pp.*276-279, *IEEE.*

[11] Yadav, J., Kumar, M., Saxena, R. & Jaiswal, A. K., (June 2014) "Performance analysis of LMS adaptive FIR filter and RLS adaptive FIR filter For noise cancellation" *Signal & image processing : an international journal (SIPIJ)* vol.4, no.3, 4304

[12] Balraj, K., Raman, A. & Gupta, D. C., (2012) "A DA Serial Multiplier Technique based on 32- Tap FIR Filter for Audio Application" Proc. of the Intl. *Conf. on Advances in Electronics, Electrical and Computer Science Engineering*— EEC 2012 Edited by Dr. R. K. Singh. ISBN: 978-981-07-2950-9 doi:10.3850/ 978-981-07-2950-9 778

[13] Borisagar ,K.R., Sedani, B.S., and Kulkarni G. R., "Simulation and Performance Analysis of LMS and NLMS Adaptive Filters in Non Stationary Noisy Environment" *International Conference on Computational Intelligence and Communication Systems* 978-0-7695-4587-5, 2011 *IEEE.*

[14] Ferdouse, L., Akhter N., Nipa T.H. & Jaigirdar, F.T., (January 2011)"Simulation and Performance Analysis of Adaptive Filtering Algorithms in Noise Cancellation" *IJCSI International Journal of Computer Science* Issues, Vol. 8, Issue 1, 185 ISSN (Online): 1694-0814.

[15] G., M., G., Kumar, V. & Shasidhar, M., (2011)"Performance Analysis of Adaptive Algorithms for Noise Cancellation", CICN, 2011, *Computational Intelligence and Communication Networks, International Conference on* 2011, pp. 586-590, doi:10.1109/CICN.2011IEEE.127

[16] Yaghoobian, H., Khazaei, A. A., Zarchi, M.S. & Hosayni, S.J.S, (2011) "Modeling of a Robust and Fast Noise Cancellation System" *Computer Modeling and Simulation, UKSIM European Symposium* on IEEE, pp. 210-213.

[17] Thenua, R. K. & Agarwal, S.K., (2010) "Simulation and performance Analysis of adaptive filter in Noise cancellation" *International Journal of Engineering Science and Technology* Vol. 2(9), 4373-4378

[18] Mousa, A., (2010) "Adaptive Noise Cancellation Algorithms Sensitivity to Parameters" 978-1-61284-732-0*IEEE.*

[19] Salivahanan, S., Vallavaraj, A. & Gnanapriya, C., (2010)*Digital Signal Processing*, 2nd ed., Published by Tata McGraw Hill Education Private Limited, 2010.

[20] Bellanger, M. G., Adaptive Digital Filters and Signal Analysis, 2nd ed. *New York: Marcel Dekker*, 2001.

[21] J. Breckling, Ed., *The Analysis of Directional Time Series: Applications to Wind Speed and Direction*, ser. Lecture Notes in Statistics. Berlin, Germany: Springer, 1989, vol. 61.

[22] S. Zhang, C. Zhu, J. K. O. Sin, and P. K. T. Mok, "A novel ultrathin elevated channel low-temperature poly-Si TFT," *IEEE Electron Device Lett.*, vol. 20, pp. 569–571, Nov. 1999.

[23] "Xilinx ISE 9.2i Software Manuals: Constraints Guide, and Development System Reference Guide". San Jose, California, USA, pp. 1-844

BIOGRAPHIES

Kumudini Sahu is studying as ME (Communication) Scholar in Department of Electronics & Telecommunication Engineering in from the Chhattisgarh Swami Vivekanand Technical University, India and BE degree in Electronics & Telecommunication from the Chhattisgarh Swami Vivekanand Technical University, INDIA in 2013. Her research is focused on Digital signal processing.

Rahul Sinha, received the ME (Communication) from the Swami Vivekanand Technical University, INDIA in 2013 and BE degree in Electronics & Telecommunication from the Pt. Ravishankar Shukla University, INDIA in 2005. He has done Advanced PG Diploma in VLSI Design in 2006. He has served as an ASIC engineer in a private organization for more than two years. Presently working as Assistant Professor in Department of Electronics & Telecommunication, Chhatrapati Shivaji Institute of Technology, DURG (CG). His research interest includes signal processing & VLSI.