



e-ISSN: 2319-8753 | p-ISSN: 2347-6710

IJRSET

International Journal of Innovative Research in
SCIENCE | ENGINEERING | TECHNOLOGY

INTERNATIONAL JOURNAL OF INNOVATIVE RESEARCH

IN SCIENCE | ENGINEERING | TECHNOLOGY

Volume 12, Issue 4, April 2023

ISSN INTERNATIONAL
STANDARD
SERIAL
NUMBER
INDIA

Impact Factor: 8.423

9940 572 462

6381 907 438

ijirset@gmail.com

www.ijirset.com

Design and Implementation of Adaptive Filter using Multi-Rate Approach for Acoustic Noise Cancellation

Gaurav Samadhiya¹, Prof. Satyarth Tiwari²

M. Tech. Scholar, Department of Electronics and Communication, Bhabha Engineering Research Institute, Bhopal, India¹

Guide, Department of Electronics and Communication, Bhabha Engineering Research Institute, Bhopal, India²

ABSTRACT:- More and more people around the world suffer from digital signal processing research field. The increase hardware complexity and increase area are the main reasons for this field. The multi-rate approach used for narrow band filter is designed and implemented in Xilinx Vertex-E XCV50E device family. The multi-rate approach is design using the decimator and interpolator structure in VHDL. Each structure is simulated using Xilinx Vertex-E XCV50E device family and compared the existing structure. The resulting structure is hardware efficient and consumes less slices compared to existing structure.

KEYWORDS: Filter Coefficient, Finite Impulse Response, Pass-band Frequency, Narrow Band Filter.

I.INTRODUCTION

Generally, Digital signal processing (DSP) is extensively used to influence, adjust, improve or filter signals such as speech, audio, image and telecommunication signals. These types of signals can be processed in the analog domain, but at the same time, digital domain offers higher speed, better accurateness, higher suppleness, maximum storing capabilities, and manageable implementation [1-3]. DSP has grown to be an elemental part of research for various real-world applications, such as mobile phones, GPS, radar, and many other applications. In circuit theory, a filter can be considered as an electrical network in which, both the amplitude parameter and phase quality get changed depending on the frequency [4-5]. Normally, a filter will not append original frequencies to the input signal, thus the frequency components of the signals cannot be changed. This leads to the variation in the relative amplitudes and phase of the various frequency components. Thus, the filters can be frequently deployed in the electronic configurations to accentuate signals with particular frequency series and thus rejects the signals with other frequency series [6-7].

Most of the signals used in the real world applications get infected in several ways depending upon the unwanted signals. Normally, the filters can be used for performing several information tasks such as, filtering, smoothing and prediction [8]. For signal processing and analysis purposes, it is very important to relieve interferences and thus reduces its effects.

This can be attained through performing several signal filtering techniques [9]. Filtering is performed at the received signal, and thus they can be interconnected with the probe drive frequency. Here, it is most easily concerned while plotting a time with respect to the amplitude of the signal [10]. Hence, it is very easy to notice that the shape of the signal mainly depends on the time duration or the sensing duration of the probe coil. For instance, if a probe coil is placed on the surface of the conductor, it produces a wave signal. Here, the signal does not need an emergence of the wave to have a large number of frequencies. The filtering method mainly depends on the essential scheme in which, most of the imperative structures of the signals are often determined on the lower frequency components and thus diminishes the higher frequency components. Thus the signal to noise ratio (SNR) is considered to be very high at small frequencies based on the several signal classes corrupted by white noise. According to this scheme, the distribution of energy from the central structures of the signal with noise can be controlled with respect to the high-frequency component of the signal. Here, the particular frequency component allows us to reclaim the most imperative structures of the signal from the noisy environment. Similarly, the other frequency components are conquered by the signal, whereas the previous components are considered as the dominated high-frequency components.



II. MULTIRATE APPROACH

The procedure of changing over a flag from an offered rate to an alternate rate is called testing rate transformation. The frameworks which utilize various examining rates in the handling of advanced flag are called multi-rate flag preparing [5].

Annihilation is the procedures of bringing down the word rate of a carefully encoded flag, which is inspected at high frequencies much over the nyquist rate. It is typically done to build the goals of an oversampled flag and to expel the out-of-band clamor. In a sigma-delta ADC, oversampling the simple information motion by the modulator alone does not bring down the quantization commotion; the ADC should utilize an averaging channel, which fills in as a decimator to expel the clamor and to accomplish higher goals. An essential square diagrammatic portrayal of the decimator is appeared in Figure 1. The decimator is a blend of a low pass channel and a down sampler. In Figure 1 the exchange work, $H(z)$ is illustrative of performing both the activities. The yield word rate of the decimator is down inspected by the factor M , where M is the oversampling proportion [6]. The capacity of low pass separating and down testing can be done utilizing an averaging circuit. The exchange capacity of the averaging circuit is given by condition (1.1). It sets up a connection between the information and yield capacities (1.1)

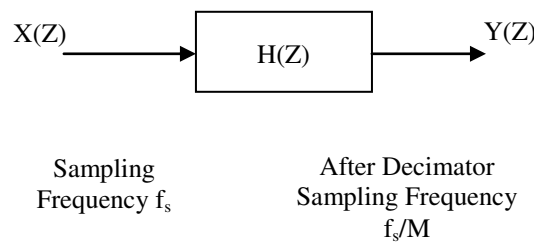


Fig. 1: Block Diagram of Decimator

$$H(Z) = \frac{X(Z)}{Y(Z)} = \frac{1}{M} \sum_{x=0}^{M-1} Z^{-x} \tag{1.1}$$

"Up sampling" is the way toward embeddings zero-esteemed examples between unique examples to expand the examining rate. (This is classified "zero-stuffing".) Up testing adds to the first flag undesired ghastly pictures which are fixated on products [7] of the first inspecting rate.

"Introduction", in the DSP sense, is the procedure of up-examining pursued by sifting. (The separating evacuates the undesired phantom pictures.) As a straight procedure, the DSP feeling of insertion is to some degree not the same as the "math" feeling of addition, yet the outcome is reasonably comparable: to make "in the middle of" tests from the first examples.

The outcome is as though you had quite recently initially tested your flag at the higher rate. Expanding the inspecting recurrence use interpolator

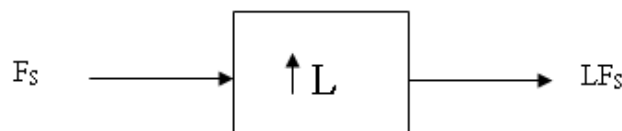


Fig. 2: Interpolation with factor L

Since addition depends on zero-stuffing you can just interject by number components; you can't insert by fragmentary variables. (Be that as it may, you can join insertion and annihilation to accomplish a general sane factor, for instance, 4/5 Up-examining adds undesired phantom pictures to the flag at products of the first inspecting rate, so except if you expel those by separating, the up-tested flag [7] isn't equivalent to the first: it's contorted.

A few applications might have the capacity to endure that; for instance, if the pictures get expelled later by a simple channel, yet in many applications you should evacuate the undesired pictures through advanced sifting. Along these lines, introduction is undeniably increasingly normal [8] that up-inspecting alone.

III. PROPOSED STRUCTURE

In this work the design of a decimation filter is presented for integrating with an existing designed modulator to form a complete sigma-delta ADC. We use multi-organize destruction channel which implies the single pulverization channel is supplanted by fell channels. In this part, we will discuss the channel engineering utilized in this work, including their structures, qualities and downsides. the initial phase in structuring an obliteration channel is to choose which sorts of channels will be utilized and where demolition will happen. This section investigates the issues associated with picking channel design for a listening device application. The general intensity of a few designs is analyzed, bringing about the three-arrange engineering that is picked to actualize this channel.

A typical need in gadgets and DSP is to confine a restricted band of frequencies from a more extensive data transmission flag. Narrowband channels rather catch just a little piece of the range. They are said to have a thin band-pass. The band-pass is basically the amount of the range the channel permits to pass. This is typically estimated in nanometers. Limited band channel comprises of decimator, restricted band and interpolator.

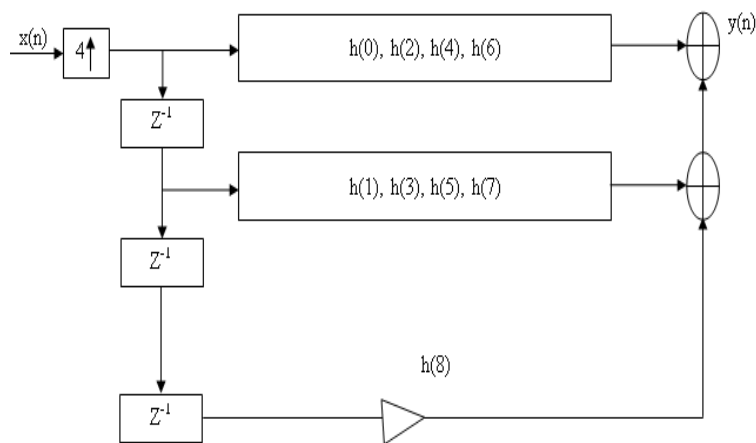


Fig. 3: Interpolator structure with filter order $N_3=9$ and $L=4$

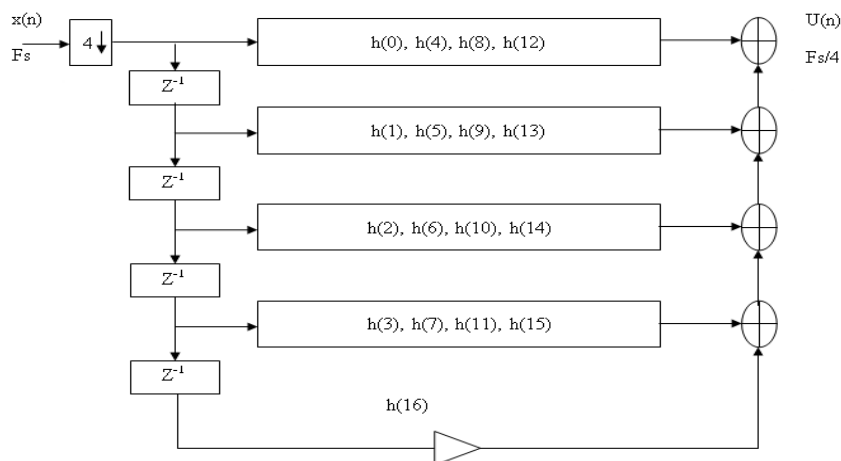


Fig. 4: Decimator structure with filter order $N_1=17$ and $M=4$



Table 1: Theoretical Analysis of Direct Method and Multi-rate Approach for Different Filter Coefficient and Sampling Frequency

Author	Sampling		Filter Coefficient	Multiplier	Adder	Register	Cycle Period
Direct Method	1		08	08	07	07	08T _s
	1		09	09	08	08	09 T _s
	1		17	17	16	16	17 T _s
	1		38	38	37	37	38 T _s
	F _s =150Hz		150	150	149	149	150 T _s
Multi-rate Approach	Decimator	2	09	09	08	08	09 T _s
		4	17	17	16	16	17 T _s
	NBF	2	75	75	74	74	75 T _s
		4	38	38	37	37	38 T _s
	Interpolator	2	05	05	04	04	05 T _s
		4	09	09	08	08	09 T _s

IV.SIMULATION RESULT

The proposed architecture has very low hardware complexity compared to direct approach based structures, because direct method requires more multiplier compare to proposed architecture. In the proposed architecture, calculate the decimator and interpolator structure for design a narrow band filter.

Table 1 shows the theoretical analysis of direct method & multi-rate approach for different filter coefficient and sampling frequency. Table 2 shows cell usage for the Comparison of Performance of the Proposed Implementation and the Existing Implementation of Narrow Band Filter.

Table 2: Comparison Result for Previous and Proposed Approach for N=5

	Previous Approach	Proposed Approach
Number of Slice	45	34
Number of Flip Flops	39	32
No. of Slice LUTs	63	57
MCPD (ns)	17.564	14.352



Fig. 5: Output Test Bench Waveform of Narrow Band Filter (Filter Coefficient N=5)



The design as were discussed in figure 3 and figure 4 were implemented using VHDL and then were tested on model sim to determine the number of slice and maximum high frequency. In figure 5, figure 6 and figure 7 have shown the output waveform of narrow band filter and chat between filter order and slices. In figure 7, compare the result between numbers of slice and filter order. Increase the filter order also increase the number of slice shown if figure 7.

Table 3: Comparison Result for Previous and Proposed Approach for N=9

	Previous Approach	Proposed Approach
Number of Slice	74	66
Number of Flip Flops	71	64
No. of Slice LUTs	141	113
MCPD (ns)	22.432	17.085

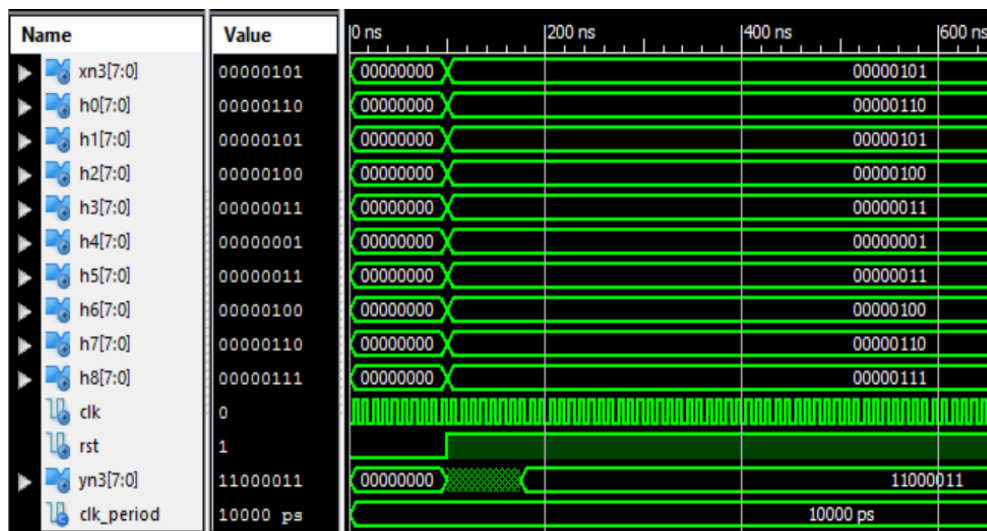


Fig. 6: Output Waveform of 9_tap Narrow Band Filter

V.CONCLUSION

The narrowband filter is realized in FIR filter. Based on the direct approach, the filter requires 150 filter coefficients to meet the desired frequency response. To implement such a large order FIR filter in hardware involves large resources and sometime difficult to implement in resource constrained application. Keeping this in view, we have used Multirate approach to design the narrowband filter. We have used down sampling factor 2 and 4 for this purpose and found that, down sampling factor 4 requires significantly less filter constants than 2. To implement the narrowband filter, we therefore chosen down sampling factor 4 and designed the decimator, interpolator and narrowband filter. The total number of filter coefficients required to realize the decimator, interpolator and the narrowband filter 64 which is almost 58% less than the direct method.

REFERENCES

- [1] Dandan Wei and Qing Xu, "Selective Partial Update Adaptive Filtering Algorithms for Block-Sparse System Identification", Security and Communication Networks, Hindawi, 2022.
- [2] H. A. Mohamed-Kazim and I. Abdel-Qader, "Coefficientgradient-based individualized s adaptation mechanism for robust system identification," Circuits, Systems, and Signal Processing, vol. 40, no. 6, pp. 3033–3074, 2021.



- [3] Q. Ling, M. A. Iqbal, and P. Kumar, "Optimized LMS algorithm for system identification and noise cancellation," *Journal of Intelligent Systems*, vol. 30, no. 1, pp. 487–498, 2021.
- [4] M. Salah, M. Dessouky, and B. Abdelhamid, "Design and Implementation of an Improved Variable Step-Size NLMS Based Algorithm for Acoustic Noise Cancellation," *Circuits, Systems, and Signal Processing*, vol. 41, pp. 551–578, 2021.
- [5] M. W. Munir and W. H. Abdulla, "On FxLMS scheme for active noise control at remote location," *IEEE Access*, vol. 8, pp. 214071–214086, 2020.
- [6] G. Li, H. Zhang, and J. Zhao, "Modified Combined-Step-Size Affine Projection Sign Algorithms for Robust Adaptive Filtering in Impulsive Interference Environments," *Symmetry*, vol. 12, 2020.
- [7] V. Tejaswi, A. Surendar, and N. Srikanta, "Simulink implementation of RLS algorithm for resilient artefacts removal in ECG signal," *International Journal of Advanced Intelligence Paradigms*, vol. 16, no. 3–4, pp. 324–337, 2020.
- [8] K. Fan, H. Qiu, C. Pei, and Y. Chen, "Robust non-negative least mean square algorithm based on step-size scaler against impulsive noise," *Advances in Difference Equations*, vol. 2020, no. 1, pp. 1–10, 2020.
- [9] J. Kapoor G. Mishra and M. Rai "A Comparative Study on Characteristics and Properties of Adaptive Algorithms applied to Noise Cancellation Techniques" 2018 International Conference on Computational and Characterization Techniques in Engineering & Sciences (CCTES) pp. 258-261 2018.
- [10] H. Ullah, M. Amir, I. Ul Haq, S. U. Khan, M. K. A. Rahim, and K. B. Khan, "Wavelet based de-noising using logarithmic shrinkage function," *Wireless Personal Communications*, vol. 98, no. 1, pp. 1473–1488, 2018.
- [11] K. B. Khan, M. Shahid, H. Ullah, E. Rehman, and M. M. Khan, "Adaptive trimmed mean autoregressive model for reduction of Poisson noise in scintigraphic images," *IJUM Engineering Journal*, vol. 19, no. 2, pp. 68–79, 2018.
- [12] Basant Kumar Mohanty, Pramod Kumar Meher and Sujit K. Patel, "LUT Optimization for Distributed Arithmetic-Based Block Least Mean Square Adaptive Filter", *IEEE Transactions on Very Large Scale Integration (VLSI) Systems*, Vol. 78, No.06, April 2016.
- [13] Azadeh Safari and Yinan Kong "Four tap Daubechies filter banks based on RNS", 978-1-4673-1157-1/12/\$31.00 © 2012 IEEE.
- [14] Yajun Zhou, Sch. of Autom., HangZhou Dianzi Univ., Hangzhou China "Distributed Arithmetic for FIR Filter implementation on FPGA" 1053-587X/\$25.00 © 2011 IEEE.
- [15] V.Sudhakar, N.S.Murthy, L.Anjaneyulu, "Fully Parallel and Fully Serial architecture for realization of high speed FIR Filters with FPGA's", *IEEE June 2010 (IJSETR)*, Volume 3, Issue 6, ISSN: 2278 – 7798.
- [16] Animesh Panda, Satish Kumar Baghmar and Shailesh Kumar Agrawal, "FIR Filter Implementation on A FPGA Allowing Signed and Fraction Coefficients with Coefficients Obtained Using Remez Exchange Algorithm" Vol 1, No 2 (October 2010).



INNO SPACE
SJIF Scientific Journal Impact Factor
Impact Factor: 8.423



ISSN INTERNATIONAL
STANDARD
SERIAL
NUMBER
INDIA



INTERNATIONAL JOURNAL OF INNOVATIVE RESEARCH

IN SCIENCE | ENGINEERING | TECHNOLOGY

9940 572 462 **6381 907 438** **ijirset@gmail.com**



www.ijirset.com

Scan to save the contact details