FIR Filter Complexity Analysis Using Multi-Rate Signal Processing Approach

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https://doi.org/10.5109/7183433

出版情報：Evergreen. 11 (2), pp.1263-1267, 2024-06. 九州大学グリーンテクノロジー研究教育センター
バージョン：

FIR Filter Complexity Analysis Using Multi-Rate Signal Processing Approach

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(Received September 14, 2022; Revised April 8, 2024; Accepted June 14, 2024).

Abstract: Finite impulse response (FIR) filter design using multi-rate signal processing (MSP) is an imperative procedure for frequency converter. Designing of decimator and interpolator in multi-rate signal, the outcomes are reducing the complexity but no effect on filter performance. The designing of digital filter is key technique in decimation for multi-rate that is emphasized in this paper. It is analyzed that the performance of filter using direct design method and multi-rate design method. It is found out multi-rate design method is more efficient in all respective over direct design method. In this proposed method FIR filter design using Kaiser Window design technique. By using this technique for filter designing method is more efficient over previous work. It is also shown in multi-rate sampling rate conversion factor 12 in this methodology that is cascaded in form and every phase corresponding to 3, 2 and 2. At the structure design stage it is also shown the requirement of component using multi-rate will be less compared to direct method.

Keywords: Decimator, FIR low pass filter, Kaiser Window, Multi-rate signal processing, Sampling rate conversion.

1. Introduction

MSP is an important technique to understand the digital frequency conversion mechanism. The procedure of convert a signal on or after a specified coefficient to a diverse sample rate is called sampling rate translation [1, 2]. The system, that utilize numerous sample rates in the handing out of digital signals are called MSP. There is much relevance related to MSP: antenna system and radar, speech and audio processing system and communication system, system etc [3, 4]. In conventional communication system, the baseband signal transmission rate is lower in comparison to intermediate frequency (IF) signal, so matching the sampling rate of one and other side[5, 6]. There is no confusion that the sampling rate of previously increased that, one equal to increase or added up to the sampling points[7, 8]. Whereas the receiving process, the signal from the analog to digital converter (ADC) by means of upper sampling rate is complicated to important, that is directly processing in the processor. Due to the above reason sampling rate pulling out is necessary to reduce the sampling speed. By the different way, the discrete sampling pointer is used for re-sampled and in the end the frequency of sampling signal will be downgraded for suitable point for the receiving the data[9, 10]. So that, the decimation and interpolation is key technique in multi-rate signal processing for up and down sampling[11, 12].

In present situation maximum transceivers are included the processing of base band signal, analogy conversion and digital frequency conversion[13, 14]. The objectives of analog frequency are exchange at the affecting of high frequency analog signal changing in suitable IF. That is reduced the constraints of the input devices and the conversion of frequency is get lower transmission rate of baseband signal [15, 16]. The digital up and down converter is designed for a low pass filter for complexity reduction in this article. It is shown that if sampling rate conversion factor is large then it would require more filter order than multistage[17, 18]. So in multistage designing in two stages or three stages filter design in cascade. The advantages of multi-rate signal processing are following:

a). Less computational requirements
b). Small amount of storage for filter coefficients
c). Minimum computational complexity
d). Low filter order necessary in MSP

Basic operations in used for achieve the above requirement in multi-rate signal processing is 'Decimation' [19, 20]. For reduction of sampling rate used in 'Decimation'. To perform, this frequently imply low pass filter on a input signal. The throw out some samples from it [21, 22]. Downward samples are extra precise phrase that referred to immediately the procedure of throw away sample, exclusive of the low pass filtering procedure. Filtering operations remove the un-desired spectral value. At the same time a linear procedure, the digital signal processing sense of decimation is to some extent dissimilar by the mathematics logic of decimation [23, 24].

2. Design of digital filter in multi-rate approach

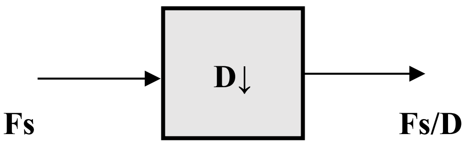


Fig.1: Decimator diagram with factor 'D'.

Design of digital filter using MSP approach by decimator design: in the decimator the factor that is used only the relation of the output sampling rate and the input sampling rate. This is represented by the symbol by 'D', so that the ratio of output sampling rate and input sample rate and its ratio is represented by decimator factor 'D' shown in Fig. 1.
The majority of this instantaneous motivation to decimation is only to decrease the sample rate by the side of the production of one structure consequently a structure working by a inferior sampling rate. It could be participation the signal. Other than a supplementary general enthusiasm is to reduce the cost of processing of signal using decimator. The computation and recollection mandatory to put into practice a digital signal processing structure normally is comparative with the sampling rate. So the employ of an inferior sampling rate frequently result inside a cheaper execution [24, 25].

On the other hand, if twice the sampling rate, an equivalent filter is required4th times as numerous operations to implement [26]. The reason is in cooperation quantity of numbers for every second. The filter lengths are increased by two, consequently complication and about by four[1, 2]. Accordingly, if divide in two the sampling rate, decrease the exertion consignment through a factor of four. It is supposed with the intention of, if reduction of the sample rate by 'D', the work intended for a filter go downward to (1/D) ^2. Constraint on factors of decimation involved throw left sample[3]. Thus simply decimation by means of integer factor and can't decimation by means of fractional factor. Example is shown in Fig. 2 signal is decimated by factor of 12.

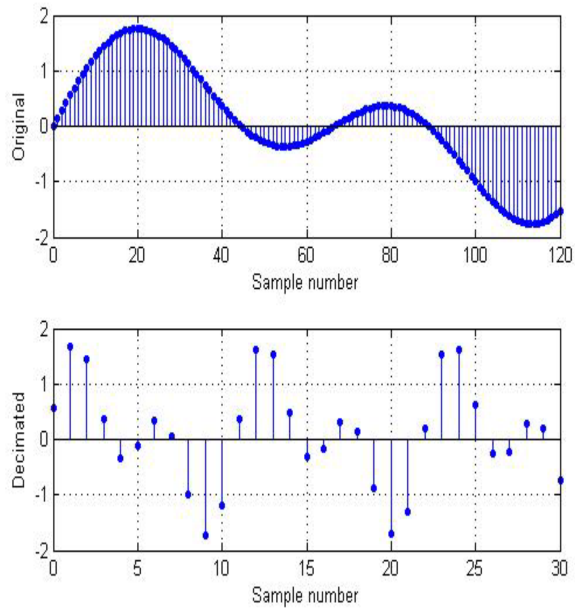


Fig.2: Decimation of original signal with factor '12'

2.1 Decimation in multiple stages

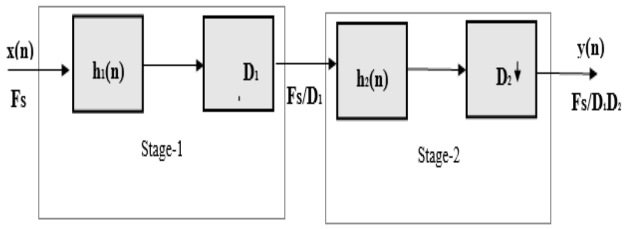


Fig. 3: Decimation in multistage

The sampling rate of input signal designed for the succession {x(n)} is 'F0=Fs'.

Decimation in many stages is improved than the single-phase decimation[4, 5]. For the reason that the filter coefficients are also reduced [6, 7]. Other than it ensures with the intention of rejection the anti-aliasing occur within the on the whole decimation process. This can be designed for every phase to keep away from anti-aliasing contained by the frequency band of attention exposed in Fig. 3.

3. Design step using Kaiser Window techniques

Consider the low-pass filter specification;stop band attenuation (α_s), pass band attenuation (α_p) and attenuation (δ) in decibels (dB) are given as:

$$\alpha_p = 20\log\frac{1+\delta}{1-\delta} \tag{1}$$

$$\alpha_s = -20 log\delta \quad (2)$$

With the help of stop band frequency (w_s), pass band frequency (w_p), calculating transition width (B) in rad/sec:

$$B = w_s - w_p \quad (3)$$

Calculating Fourier coefficients $h_d(n)$ by means of Fourier series technique; The idyllic frequency response $H(e^{jw})$:

$$H(e^{jw}) = 1, for\ |w| \leq w_c$$
$$= 0, for\ w_c \leq |w|\ w_c \quad (4)$$

Determine the parameter α from following equation:

$$\alpha = 0, for\ \alpha_s \leq 21$$
$$= 0.58(\alpha_s - 21)^{0.4} + 0.58(\alpha_s - 21), for\ 21 \leq \alpha_s \leq 50$$
$$= 0.11(\alpha_s - 8.7), for \alpha_s > 50 \quad (5)$$

Choosing the parameter 'D' using following method:

$$D = 0.922, for\ \alpha_s \leq 21$$
$$= \frac{\alpha_s - 7.95}{14.36}, for \alpha_s \geq 21 \quad (6)$$

Calculating the filter order for sampling frequency (w_{sf})

$$N \geq \frac{w_{sf} D}{B} + 1 \quad (7)$$

Table 1. Parameter for direct stage filter specification.

Filter Parameter	Direct Method
Sampling frequency (Fs)	180 MHz
Stopband frequency (fs)	2.5MHz
Passband frequency (fp)	2.1MHz
Stopband attenuation (δs)	±0.2dB
Passband attenuation (δp)	-42dB
Filter order (N)	**1069**

It is depending on the filter coefficient that is requisite to realize the preferred filter frequency response. The narrow-band filter might be implementing in a straight line otherwise by means of the multi-rate technique. As per the low pass filtering specification given in table 1, the filter pass band frequency 2.1MHz and their pass band attenuation -40dB. The sampling frequency, stop band frequency and pass band frequency for this specification given in Table 1. The filter order is too high, it means designing and implementation of large filter order required a greater number of component as well as degrading filter performances. By the above specification given in table 1proposed a low-pass filter that is shown their gain vs. frequency response of low-pass filter in Fig. 4.

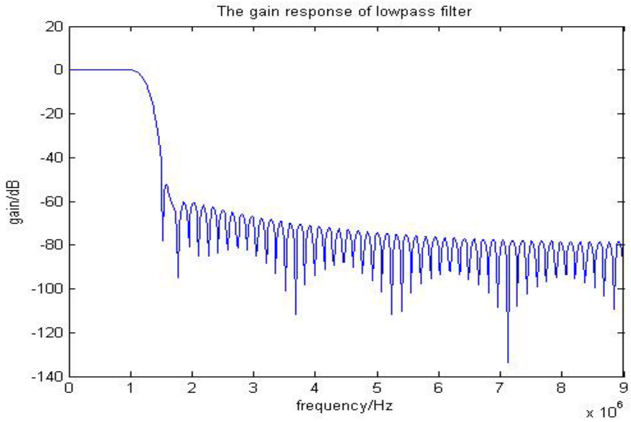


Fig. 4: Gain response of low pass filter.

Design of low pass filter using multi-rate approach, the specification is given in Table 2.

Table 2. Evaluation of filter coefficients using the narrow-band filter existing technique as well as proposed method

Filter Parameter's	1st stage	2nd stage	3rd stage
fp(MHz)	2.1	2.1	2.1
fs(MHz)	17.5	7.5	2.5
Fs(MHz)	60	20	10
Filter order (Existing)	**42**	**40**	**261**
Filter order (proposed)	**11**	**10**	**61**

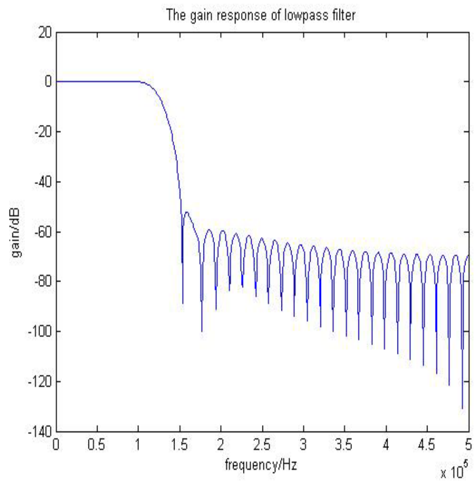


Fig. 5: Gain response of low pass filter after decimation

Find out the filter coefficients are mandatory in the direction of put into operation. The narrow-band filters are considerably a smaller amount during multi-rate techniques than the straight technique. Amongst the two case discussed, here one is previous method designed by Hamming window method. The proposed approach is using by Kaiser Window designing method. Here decimation factor used D=12, that is designed in cascaded manner 3, 2 and 2. Therefore, it is measured downward sampling factor D=12 to execute the specified narrow-band filter. In Fig. 5 it is shown that the frequency vs. gain response of low pass filter after decimation, by this response it is shown after decimation there is no effect on filter response.

4. Complexity analysis

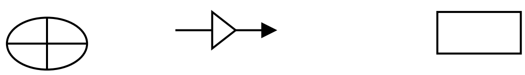
Here predictable the mandatory filter coefficient for mutually this method to come across the difficulty of the narrow-band filter. During the implementation of the narrow-band filter we calculate the filter coefficient of each block. The filter coefficient calculated the by the MAT LAB using Kaiser Window Function. The values obtain through MAT LAB.

4.1 Structure to implement the filters

The input and output of this filter is represented through the difference equation as given here

$$y(n) = \sum_{k=0}^{M-1} h(k)x(n-k) \tag{8}$$

Where y(n),x(n) and h(n) are the output signal, input signal, impulse response respectively. Symbol used in FIR filter structure



Adder Multiplier Delay element

Computations of equation (8) can be performed using the direct form-1 structure shown in Fig. 6.

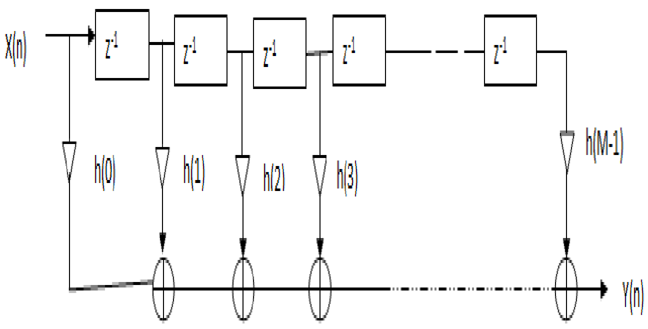


Fig.6: FIR direct -1 form structure.

Table 3. Comparison of component required of the narrow-band filter obtain by direct approaches well as multi-rate approach.

Approach	Multiplier	Adder	Register
Direct Method	1069	1068	1068
Multi-rate approach (Existing)	343	342	342
Multi-rate approach (Proposed)	82	81	81

From the table 3, the result is with the intention of number of component necessary to realize the narrow-band filter is considerably a smaller amount in accomplished by multi-rate method than the direct technique.

5. Conclusion

In this article, the FIR filter is designed using direct method and multi-rate down conversion method. Here compared the filter order by the existing work (Hamming window) method and Kaiser Window techniques. It is found that Kaiser Window designing method has the better response in direct designing and Multirate designing approach. The complexities are also analyzed using FIR filter Direct form-I method and shown that the component required in for structure for multi-rate design also less compared to direct design method. This design method improves the existent instance performances of signal processing application and mostly reduced the hardware requirement or FIR filter design.

References

1) M. Elimelech, and W.A. Phillip, "The future of seawater desalination: energy, technology, and the environment," *Science (80-.).*, **333** (*6043*) 712–717 (2011). doi:10.1126/science.1200488.
2) M.A. Lambert, and B.J. Jones, "Automotive adsorption air conditioner powered by exhaust heat. part 2: detailed design and analysis," *Proc. Inst. Mech. Eng. Part D J. Automob. Eng.*, **220** (*7*) 973–989 (2006). doi:10.1243/0954 4070 JAUTO222.
3) Singh, Mahesh K., A. K. Singh, and Narendra Singh. "Multimedia analysis for disguised voice and classification efficiency." *Multimedia Tools and Applications* 78.(20): 29395-29411(2019). doi: 10.1007/s11042-018-6718-6.
4) Syahrudin, Muhamad Fuad, Muhammad Arif Budiyanto, and Muhammad Aziz Murdianto. "Analysis of the Use of Stern Foil on the High Speed Patrol Boat on Full Draft Condition." *Evergreen* 7.(2): 262-267 (2020). doi:10.5109/4055230.

5) Amiruddin, Hilmi, et al. "CNGDI Engine Performance Using a Vaned Diffuser Turbocharger Compressor with Varying Injection Timings.". 414-420 (2021). doi: 10.5109/4480723
6) “Rathore, R. P. S. “Reconfigurable digital radar receiver implemented in FPGA using Under-sampling, Direct IQ generation, Multi-rate filter and pulse compression.” In 2014 IEEE International Microwave and RF Conference (IMaRC) 174-177, (2014). IEEE. https:// doi:10.1109/IMaRC.2014.7039001.”
7) Singh, Mahesh K., A. K. Singh, and Narendra Singh. "Disguised voice with fast and slow speech and its acoustic analysis." *Int J Pure Appl Math* 118.(14), 241-246 (2018).doi: doi: 10.1109/ICASSP.2018.8462169.
8) Balaji, V. Nithin, P. Bala Srinivas, and Mahesh K. Singh. "Neuromorphic advancements architecture design and its implementations technique." *Materials Today: Proceedings* (2021). doi: 10.1016/j.matpr. 2021.06.273.
9) Akbar, Lasta Azmillah, et al. "Method development of measuring depth of burn using laser ranging in laboratory scale." *Evergreen* 7.2: 268-274 (2020). doi: 10.5109/4055231.
10) Singh, Mahesh K., A. K. Singh, and Narendra Singh. "Multimedia utilization of non-computerized disguised voice and acoustic similarity measurement." *Multimedia Tools and Applications* 79.(47), 35537-35552 (2020). doi: 10.1007/s11042-019-08329-y.
11) Punyavathi, G., M. Neeladri, and Mahesh K. Singh. "Vehicle tracking and detection techniques using IoT." *Materials Today: Proceedings* (2021). doi: 10.1016/j.matpr. 2021.06.283.
12) Daud, Dalina, et al. "Radical Scavenging Activity Assay and Red Fluorescence Microscopy Studies: Antioxidant Properties of Selected Young and Mature Leaves for Application in Pharmaceutical Industry." *Evergreen* 7.2: 216-220 (2020). doi:10.5109/4055222
13) Singh, Mahesh, Durgesh Nandan, and Sanjeev Kumar. "Statistical Analysis of Lower and Raised Pitch Voice Signal and Its Efficiency Calculation." *Traitement du Signal* 36.(5): 455-461 (2019). doi: 10.18280/ts.360511.
14) Berawi, Mohammed Ali, et al. "Determining the Prioritized Victim of Earthquake Disaster Using Fuzzy Logic and Decision Tree Approach." *Evergreen* 7.2: 246-252 (2020). doi:10.5109/4055227.
15) Padma, Uppalapati, Samudrala Jagadish, and Mahesh K. Singh. "Recognition of plant’s leaf infection by image processing approach."*MaterialsToday: Proceedings* (2021). doi: 10.1016/j.matpr. 2021. 06.297.
16) Veerendra, G., et al. "Detecting plant Diseases, quantifying and classifying digital image processing techniques." *Materials Today: Proceedings* (2021). doi: 10.1016/j.matpr. 2021.06.271.
17) Singh, Mahesh K., Narendra Singh, and A. K. Singh. "Speaker's voice characteristics and similarity measurement using Euclidean distances." *2019 International Conference on Signal Processing and Communication (ICSC)*. IEEE, 2019.doi: 10.1109/ICSC 45622. 2019. 8938366.
18) Alhamid, M. I., et al. "Study on the Effectiveness of Ozonation Technique in Preventing Scale Precipitation on Closed System Cooling Towers." *Evergreen* 6.1: 65-70 (2019). doi:10.5109/2321013.
19) Wong, Keng Yinn, et al. "Effects of surgical staff turning motion on airflow distribution inside a hospital operating room." *Evergreen* 6.1 52-58 (2019). doi:10.5109/2321008.
20) Balaji, V. Nithin, P. Bala Srinivas, and Mahesh K. Singh. "Neuromorphic advancements architecture design and its implementations technique. "*Materials Today: Proceedings* (2021). doi: 10.1016/ j.matpr. 2021.06.273.
21) Anushka, R. L., et al. "Lens less Cameras for Face Detection and Verification." *2021 6th International Conference on Signal Processing, Computing and Control (ISPCC)*. IEEE, 2021. doi: **doi:** 10.1109/ISPCC53510.2021.9609392.
22) Wulan, P. P. D. K., Juli Ayu Ningtyas, and Miranda Hasanah. "The effect of nickel coating on stainless steel 316 on growth of carbon nanotube from polypropylene waste." *Evergreen* 6.1: 98-102 (2019) doi:10.5109/2328411.
23) Nandini, A., R. Anil Kumar, and Mahesh K. Singh. "Circuits Based on the Memristor for Fundamental Operations." *2021 6th International Conference on Signal Processing, Computing and Control (ISPCC)*. IEEE, 2021. doi: 10.1109/ISPCC53510.2021.9609439.
24) Syahrul, Azwan Shaedi, et al. "Entropy generation minimization of two-phase flow in a mini channel with genetic algorithm." *Evergreen, Joint Journal of Novel Carbon Resource Sciences & Green Asia Strategy* 6.01: 39-43 (2019). doi:10.5109/2321004
25) Satya, P. Mohana, et al. "Stripe Noise Removal from Remote Sensing Images." *2021 6th International Conference on Signal Processing, Computing and Control (ISPCC)*. IEEE, 2021. doi: 10.1109/ISPCC53510.2021.9609457.
26) Hamzah, N., et al. "Identification of CNT growth region and optimum time for catalyst oxidation: experimental and modelling studies of flame synthesis." *Evergreen Jt J Novel Carbon Resour Sci Green Asia Strategy* 6: 85 (2019). doi:10.5109/2328409.