Available Online at www.ijcsmc.com

International Journal of Computer Science and Mobile Computing

A Monthly Journal of Computer Science and Information Technology

ISSN 2320–088X

IMPACT FACTOR: 6.017

IJCSMC, Vol. 7, Issue. 12, December 2018, pg.213 – 221

Review of Polyphase Filtering Technique in Signal Processing

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ABSTRACT

The signals that are encountered in practice are mostly analog signals. These signals, which appear consistently in time and amplitude, are processed using active and passive circuit elements using a power network. This approach is known as the analog signal processing (ASP) - For example Radio and television communication. They may be processed using digital hardware, including adders, multipliers, and logic elements or any other special-purpose microprocessor. However, we needs to change these analog signals to the appropriate digital format such signal forms are called digital signals. The signals have noise which affects the quality of the signal. In this paper, poly-phase filter which can remove empty bands from the signal is reviewed.

Keywords: Poly-phase, FIR, Multi-rate, DFT, FFT, Active and passive filter.

INTRODUCTION

The modern world is surrounded by all kinds of signals in various forms. Some of the signals are by natural; however most of the signals are created in various processes. Some signals are very important (Eg speech), some are pleasant in nature (music), while many other are unwanted or unnecessary .In engineering, signals may be carriers of knowledge, both useful and unwanted. Therefore extracting or enhancing the helpful data or information from a mixture of conflicting data (mixed signals) is that the simplest variety of signal process. More generally, signal processing is an operation designed for extracting, enhancing, storing, and transmitting useful information [1]. The distinction between helpful and unwanted data is usually subjective yet as objective. Hence signal processing is an application dependent process.

It takes one in all the finite numbers of values at specific instances in time, and hence it can be represented by binary numbers, or bits. The “natural” signals to be appropriate for microprocessors they have to be reborn into a digital kind through a method known as Digital Signal process (DSP). In order to accurately represent these signals they need to go through certain types of predesigned sampling and quantization techniques based on different mathematical techniques [2]. Using DSP strategies has varied benefits over the standard analog strategies or processes, together with larger accuracy, performance, flexibility, speed and price.

1.1.1 Digital Filters

In many applications of the signal process, we want to change the relative amplitude and frequency divisions of a signal. This process is generally referred to as filtering. Since the Fourier transform of the output is product of input Fourier transform and frequency response of the system, the appropriate frequency response is to be used. An analog filter uses electronic logic such as parts of resistors, capacitors and op-amps to create the necessary filtering effect. Digital filter use digital processors to perform numerical calculations on sample values. A processor can be a general purpose computer such as a PC or a special Digital Signal Processor chip.

The following list gives the main advantages of digital filters

i A digital filter is programmable, i.e. It is operated by a program stored in the processor's memory. This means that digital filters can be easily changed without affecting the hardware.

ii Digital filters are easily designed, tested and implemented on a general-purpose computer.

iii. The characteristics of analog filter circuits (particularly those containing active components) are subject to drift and are dependent on temperature [3]. Digital filters do not suffer from these problems, and hence show stability with both time and temperature.

iv. Unlike their analog counterparts, digital filters can handle low frequency signals accurately. Because the speed of DSP technology is increasing, digital filters are being used for high-frequency signals in the RF (radio frequency) domain, which were last time the exclusive preserve of analog technology.

v. Digital filters are very much more versatile in their ability to process signals in a variety of ways; this includes the ability of some types of digital filter to Optimized for changes in signal properties

1.2 Signal Noise

Noise is outlined as Associate in any unwanted signal that interferes with the communication or calculus of another signal. A noise itself could be a signal that conveys data relating to the supply of the noise. Signal distortion is that the term typically wont to describe a scientific undesirable amendment in a very signal and refers to changes in a very signal because of the non–ideal characteristics of the transmission channel, reverberations, echo and missing samples [4]. Depending on its supply, a noise is classified into variety of classes, indicating the broad physical nature of the noise, as follows:

a. **Acoustic Noise**: emanates from moving, vibrating, or colliding sources and is that the most acquainted style of noise gift in varied degrees in everyday environments. Acoustic noise is generated by such sources as moving cars, air-conditioners, computer fans, traffic, people talking in the background, wind, rain, etc.

b. **Electromagnetic Noise**: gives the least bit frequencies and particularly at the radio frequencies. All electrical devices like Radio & T.V transmitters and receivers generate electromagnetic noise.

c. **Electrostatic Noise:** it is generated by the presence of a voltage or current flow. Fluorescent lighting is also a source of electro static noise.

d. **Channel distortions, echo, and fading:** due to non-ideal characteristics of communication channels. Radio channels, such as those at microwave frequencies used by cellular mobile phone operators, are particularly sensitive to the propagation characteristics of the channel environment [5].

e. **Processing Noise:** such noise results from the digital/analog conversion of signals, e.g. quantization noise in digital conversion of speech or image signals, or lost data packets in digital data communication systems.

1.3 Multirate DSP

Multirate digital signal processing (DSP) has attracted much attention over the past two decades due to the applications in sub-band coding of speech, audio and video, multiple carrier data transmission, etc. A key characteristic of multi-rate algorithms is their high processing potency. A multi-rate system can increase or decrease the sampling rate of individual signals before or while processing them. These signals then with different sampling rate can be simultaneously processed in various parts of the multi-rate system. Digital filter banks are the most important applications of multi-rate DSP [6].

A various number of different filter bank have been developed over last fifteen years. Among those filter banks, Cosine Modulated filter banks are very popular because they are easy to implement and can provide perfect reconstruction (PR). The Discrete Fourier Transform (DFT) polyphase filter bank is another popular filter bank that provides high computational efficiency, but suffers from the fact that it is not able to cancel alias components caused by sub-sampling the sub band signals. By introducing a certain modification to the DFT filter bank, we can overcome its disadvantage. The modified DFT (MDFT) filter banks can also provide PR.

1.4 Polyphase Filter

Polyphase filters have become a awfully vital part within the style of assorted filter structures thanks to the actual fact that it reduces the price and complexness of the filter by doing the method of devastation before filtering that reduces the multiplications per input sample. Teaching DSP essentially needs significant use of arithmetic, the nature of the material requires mathematics to precisely specify the methods and firmly establish their characteristics and performance. In digital signal process, many application areas require sampling rate alteration. The processes concerned within the alteration of the rate ar interpolation and devastation. One of the foremost economical structures to implement interpolation and devastation operations is that the point in time structure. Decimation is that the method of reducing the sample rate Fs in an exceedingly signal process system, and interpolation is the opposite, increasing the sample rate Fs in a signal processing system. These processes are quite common in signal process systems associate degreed are nearly perpetually performed exploitation an FIR filter.

Passive point in time filter implementation is thought to need smart quality passives, which is no longer self-evident with deep-submicron CMOS processes. Additionally, the tuning of the passive polyphase filters is difficult [7]. Therefore, with the high-speed transistors out there, active polyphase filters (APPF) have been proposed to overcome these limitations. The active point in time filters give vital blessings over their passive counterparts. With APPFs signal amplification, filter response activity and calibration ar doable. Contrary to PPFs, the most critical design parameters of APPFs are determined by the active elements. Compared to passive point in time filter implementation, edges of the APPF embody the likelihood for filter activity, calibration and signal amplification. Although practical APPF implementations have been presented, this is the first time APFFs have been analyzed in terms of gain and stability.

Literature Review

Tong Liu, et.al (2016) presented that Digital Down Conversion (DDC) is one of the key technologies for software radio, which converts high frequency data streams into low frequency data streams for easy subsequent real-time processing by DSP devices. This paper presents a under the digital frequency conversion method based on polyphase filtering [8]. Using Field Programmable Gate Array (FPGA) under the high speed digital frequency conversion was implemented, and introduced including Numerical Controlled Oscillator(NCO), frequency mixing module, CIC filter, HB filters, FIR filter, are discussed in detail. Meanwhile, use the software radio platform of my laboratory and the chipscope to collect data, and use the MATLAB analyze the data, to verify the correctness and feasibility of the theoretical simulation.

Gopal S. Gawande, et.al (2016) presented that Multi-rate filtering technique is widely used for meeting the sampling rates of different systems and it is a powerful technique in DSP which results in low-cost implementations of digital filters. Polyphase decomposition technique reduces the computational complexity by adopting parallelism in multirate digital filters [9]. Power consumption is usually a major style constraint whereas implementing multirate digital filters on reconfigurable hardware. Dynamic power consumption in digital implementations will be reduced by minimizing the switch activity at the output of CMOS gates. Most of the work is rotated around minimizing range of power of 2 or nonzero terms within the filter coefficients as every non-zero bit corresponds to an extra adder in hardware implementation. This paper proposes a completely unique rule to scale back the switch activity at the output of CMOS gate by reducing the quantity of nonzero terms within the filter coefficients. The proposed algorithm is applied to the polyphase structures and its impact on dynamic power consumption is analyzed. The structures are synthesized for Spartan6 FPGA board using Xilinx System Generator. The proposed algorithm reduces the power consumption up to 31 milli Watts for the polyphase structures.

Fred harris, et.al (2015) presented a review of efficient architectures to implement wide bandwidth filters based on use of PR non-maximally decimated polyphase filter banks. An earlier option used a two tier polyphase filter to implement variable bandwidth filters while the newer option uses interleaved polyphase filter banks to implement variable bandwidth filters with reduced delay through the filter [10]. Mixes of PR non maximally decimated filter banks without fine bandwidth adjustment and with fine bandwidth adjustment using inner tier processing or using interleaved processing offer flexible architectures to build efficient multi-resolution wideband software defined radios. The design parameters of these designs include IFFT size, polyphase filter with length related to selectable transition bandwidth and ripple performance, inner tier filter or polyphase inner tier filter option, and now interleave variable bandwidth channelizers. All these options prove to be green, requiring significantly less processing power to perform a specified filtering task.

Rahul M. Deshmukh, et.al (2015) presented that Multirate filter are widely used in many DSP application. Much efficient architectures are design to reduce the complexity of DSP system. Adder and Multiplier are the main fundamental blocks of filter which contributes in reduction of area, power and delay parameter of filter [11]. This paper presents polyphase FIR filter using bypass feed direct multiplier and polyphase FIR filter using shift and add multiplier. The proposed polyphase filter is design for filter of length nine .The proposed and conventional designs are simulated using Xilinx ISE 13.1 tool. On comparison, proposed design is efficient in terms of area and delay than conventional design.

David B. H. Tay, et.al (2016) proposed the polyphase structure in the down sampled domain, which is computationally efficient for critically sampled bipartite graph filter banks [12]. It was shown that the signals in the down sampled domain are defined on equivalent sub graphs and the filtering can be defined as spectral filters with respect to these sub graphs. The results from the polyphase analysis reveal analogies of the graph filter banks with 1-D filter banks but some differences also exist. The theoretical results allowed the generalization of the filter bank to bipartite directed graphs and filters with more general base matrices. It was also demonstrated that the use of alternative base matrices can lead to improvement in nonlinear approximation applications. Future work can include developing techniques for decomposing an arbitrary directed graph into a series of bipartite directed graphs.

Yingying Du, et.al (2016) presented that digital down-conversion technology based on the structure of bandpass sampling is one of the key technologies of software radio receiving system. While the traditional digital mixing orthogonal demodulation using multiplier, to a certain extent, increase the computational complexity. This paper proposes a kind of digital down-conversion design based on polyphase filtering structure, effectively reducing the computational complexity [13]. Firstly, conducting odd-even extraction and symbol correction; secondly, using time delay filters to update; finally, the baseband signal is given as output after extracting. The software simulation is applied to prove its feasibility. The digital down-conversion is completed by adopting polyphase filtering structure; output data rate is reduced after ten times extraction. This can effectively alleviate the pressure of the baseband processing, and system resource consumptions are low relatively, so the design scheme is practical.

Yaprak Eminaga, et.al (2015) presented a low complexity high efficiency decimation filter which can be employed in Electro Cardio Gram (ECG) acquisition systems. The destruction filter with a destruction quantitative relation of 128 works in conjunction with a 3rd order letter of the alphabet delta modulator. It is designed in four stages to reduce cost and power consumption [14]. The work reported here provides Associate in nursing economical approach for the destruction method for top resolution medicine conversion applications by using low complexness two-path all-pass primarily based destruction filters. The performance of the projected destruction chain was valid by mistreatment the MIT-BIH heart condition information and comparative simulations were conducted with the state of the art.

Zhao Kongrui, et.al (2015) studied that for the limitation of the processing speed in hardware, the traditional down-conversion doesn't work for GHz high speed sampling signal. In this paper, a novel approach is proposed with polyphase decomposition to achieve the down-conversion of the GHz high speed sampling signal [15]. In this method, the input high speed sampling signal and the corresponding oscillation signal are parallelized to be M sub-channels of low speed sampling with delay and decomposition approach, which can be viewed as polyphase decomposition. The down-conversion operation is performed for every channel of input file. Then the down-conversion of the rate high speed sampling signal comes with the down-conversion results of every channel. The simulations verify the effectiveness of the proposed method.

Yuichi Tanaka et.al (2017) addressed a polyphase1 structure of spectral graph wavelets and filter banks. The two-channel critically sampled graph filter banks are considered here. In classical signal processing, polyphase structure of filter banks is very useful since downsampler (upsampler) can be placed before analysis filtering (after synthesis filtering) [17]. This work theoretically derives that a similar structure is also possible for spectral graph filter banks. The structure can be used for any two-channel critically sampled spectral graph filter banks as long as an underlying graph is bipartite. Similar to classical signal processing, it can move down-sampling and up-sampling operators before and after the analysis and synthesis filtering operations, respectively. The numbers of transformed coefficients in the lowpass and highpass channels are not necessarily to be N/2.

Shilian Zheng et.al (2014) proposed a wideband spectrum sensing procedure based on the idea of group testing in this paper. A new polyphase filter bank (PFB) for which the output subbands have the same bandwidth and contain continuous frequency components is proposed [18]. This PFB is utilized to form a two-stage wideband spectrum sensing method. Cognitive radio has been proposed as a solution to the spectrum underutilization problem by opportunistically accessing the currently unused spectrum licensed to primary users (PUs). Because cognitive radios are considered as secondary users (SUs) of the spectrum, they should not cause harmful interference to PUs. To reach this goal, SUs need to sense a wide spectral range to detect spectrum holes for communications. This process is referred to as wideband spectrum sensing. The simulations show that the proposed method can reduce the number of binary tests greatly when the number of occupied primary channels is low.

Mark L. Fowler, (2011) studied that DSP is a mathematically heavy topic and to fully understand it students need to understand the mathematical developments underlying DSP topics. However, relying solely on mathematical developments often clouds the true nature of the foundation of a result [19]. It is possible that students WHO master the arithmetic should still not really grasp the key ideas of a subject. Furthermore, teaching DSP topics by merely “going through the mathematics” deprives students of learning the art of discovery that will make them good researchers. This paper uses the subject of multiphase devastation and interpolation for instance however it's doable to take care of rigor nonetheless teach mistreatment less mathematical approaches that show students however researchers think when developing new ideas.

P. P. Vaidyanathan, (2013) considered an analog synthesis-analysis filter bank and showed how to convert it into a digital filter bank based on the knowledge that the model signal x(t) has a finite rate of innovations. This conversion is based on the idea of integer oversampling. The advantage of this is that all designs and implementation will benefit from digital filter bank theory, and no semi-infinite matrices are involved [20]. Oversampling is only a convenient theoretical ruse, and in practice the sampling filter bank is still implemented before sampling- no oversampling is really involved. The sampling filter bank can be implemented as a transversal filter, which can be an advantage. Even though we have not addressed any optimization problems due to lack of space, a number of such interesting problems are evident from the framework.

CONCLUSION

The new scheme we propose, tackles non-uniformity along-track within a single look complex (SLC) single channel or post-beam formed SAR collection arising from different PRFs (or from arbitrary sampling). It takes in demodulated SAR data for different acquisitions, which are collected and oversampled at variable PRFs, and delivers resampled data at a lower, constant PRF within each acquisition, and uniformly sampled in the spatial frequency domain (kspace). A new polyphase filter-based implementation allows digital filtering at the lowest possible rate, viz., and the effective output PRF rate. The result can be computationally efficient fully on-board algorithm enabling in-place and real time processing which avoids up/down-link data transfers and bottlenecks. The POLYPHSE method approximately reconstructs the collected data on a uniformly spaced grid along the synthetic aperture, while preserving the resolution and Nyquist constraint within the cross-range extent of interest. The threshold based technique is applied which can remove empty band from the signal. When the empty bands get removed from the signal it leads to increase quality of the speech signal.

References

[1] J.G. Proakis and D.G. Manolakis, "Digital Signal Processing: Principles, Algorithms, and Applications", 2007, 4th Ed., Pearson Prentice Hall, Upper Saddle River, NJ

[2] P.P. Vaidyanathan, "Multirate Systems and Filter Banks", 1993, Prentice-Hall, Englewood Cliffs, NJ

[3] N.J. Fliege, "Multirate Digital Signal Processing: Multirate Systems, Filter Banks, Wavelets", 1994, John Wiley and Sons, West Sussex, England.

[4] R. E. Crochiere and L. R. Rabiner, "Interpolation and decimation of digital signals: A tutorial review," 1981, Proceedings of the IEEE, vol. 69, no. 3, pp. 300–331

[5] S.P. Harris, E.C. Ifeachor, "Automatic design of frequency sampling filters by hybrid genetic algorithm techniques", 1998, IEEE Trans. Signal Process. 46(12), 3304–3314

[6] K. Uesaka, M. Kawamata, "Synthesis of low-sensitivity second-order digital filters using genetic programming with automatically defined functions", 2000, IEEE Signal Process. Lett. 7(4), 83–85

[7] J.T. Kim, "Efficient implementation of polynomial interpolation filters or full digital receivers", 2005, IEEE Trans. Consum. Electron. 51(1), 175–178

[8] Tong Liu, Jiayu Han, Zhigang Li, "Broadband DDC Based on Polyphase Filter and Its FPGA Implementation", 2016 IEEE International Conference on Electronic Information and Communication Technology (ICEICT)

[9] Gopal S. Gawande, Dr. K.B. Khanchandani, "Design, Implementation and Analysis of Power Efficient Polyphase Multirate Filters", 2016 3rd International Conference on Signal Processing and Integrated Networks (SPIN)

[10] Fred harris, Elettra Venosa, Xiofei Chen, Chris Dick, "Interleaving Different Bandwidth Narrowband Channels in Perfect Reconstruction Cascade Polyphase Filter Banks for Efficient Flexible Variable Bandwidth Filters in Wideband Digital Transceivers", 2015, IEEE

[11] Rahul M. Deshmukh, Rashmi Keote, "Design of Polyphase FIR Filter using Bypass Feed Direct Multiplier", IEEE ICCSP 2015

[12] David B. H. Tay, Antonio Ortega, "Bipartite Graph Filter Banks: Polyphase Analysis and Generalization", 2016, IEEE

[13] Yingying Du, Xinjing Ye, Yafei Li, Zhengyu Cai, "Digital Down-conversion Design Based on Polyphase Filtering Structure", 2016, IEEE International Conference on Signal and Image Processing

[14] Yaprak Eminaga, Adem Coskun, Sterghios A. Moschos, and Izzet Kale, "Low Complexity All-Pass Based Polyphase Decimation Filters for ECG Monitoring", 2015, IEEE

[15] Zhao Kongrui, Zhang Chao, "A polyphase digital down converter method for GHz high speed sampling signal", 2015, IEEE 12th International Conference on Electronic Measurement & Instruments

[16] J. Tuthill, G. Hampson, J. D. Bunton, f. j. harris, A. Brown, R. Ferris and T. Bateman, "COMPENSATING FOR OVERSAMPLING EFFECTS IN POLYPHASE CHANNELIZERS: A RADIO ASTRONOMY APPLICATION", 2015 IEEE Signal Processing and Signal Processing Education Workshop (SP/SPE)

[17] Yuichi Tanaka and Akie Sakiyama, "A "Polyphase" Structure of Two-Channel Spectral Graph Wavelets and Filter Banks", 2017, IEEE

[18] Shilian Zheng and Xiaoniu Yang, "Wideband Spectrum Sensing Based on Group Testing Utilizing Polyphase Filter Banks", 2014, Proceedings of the 19thInternational Conference on Digital Signal Processing

[19] Mark L. Fowler, "Polyphase Filters- A Model for Teaching the Art of Discovery in DSP", 2011, IEEE

[20] P. P. Vaidyanathan, "Analog Filter Banks for Sampling: Discretization, Polyphase Form, and Role in Compressive Sensing", 2013, IEEE

