APPLICATION OF MODIFIED SECOND-ORDER FREQUENCY TRANSFORMATIONS

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Abstract - *The frequency transformation based filters (FT filters) give an complete control over the cutoff frequency. On the other hand the cutoff frequency range of the FT filters is inadequate. FTCDM filter has wider cut off frequency range compared to the FT filter. But it also has limited range of cutoff frequency range; To overcome the problem of limited cutoff frequency range using a modified second-order frequency transformation based filter (MSFT filter). The one-to-one mapping condition stuck between the frequency variables and use low-pass to high-pass transformation on the prototype filter to attain wider range of cutoff frequency. By using the MSFT filter the cut off frequency range can be improved.MSFT filter reduce the noise in noisy signal.MSFT filter detect the original noiseless signal.*

***Key Words*: Finite impulse response (FIR) filter, frequency transformations, variable linear-phase digital filter.**

1. INTRODUCTION

variable finite impulse response (FIR) filters (FIR filters whose frequency response can be changed on-the-fly, based on the beloved specifications) are useful in wireless communications, adaptive systems, audio signal processing, and biomedical applications .In multistandard wireless transportation variable filters are required for applications, such as channelization (extraction of individual radio channels from wideband input) and spectrum sensing. Interoperability with the existing and future generation wireless communications standards is the key characteristic of multi standard wireless mobile devices.

These devices should be able to support ultra-narrowband as well as ultra-wideband frequencies. Therefore, the channelizer in a multi standard radio should be able to extract the channels of various bandwidths. (Channel bandwidth specifications of various transportation standards vary from 26 kHz to 23.1 MHz.) Due to its linear phase property, FIR filter is regularly preferred to infinite impulse response filter for the channelization task. Variable FIR filters (filter whose passband width can be varied on-the-fly) are required in these future generation wireless communication devices.

A variable filter should have low area complexity so as to satisfy the stringent area constraint for the handheld, battery-operated terminals at the user end. It should also be able to provide all the desired frequency responses more a large frequency range, without the need of hardware reimplementation. Designing such a variable filter with fine control over the cutoff frequency and with low area complexity is a challenging task, which is the objective of the work presented in this paper. The frequency response of an FIR filter can be changed by completely changing its coefficients or by modifying the impulse response using various operations.

In the case of programmable digital filters, the desired frequency responses are obtained by updating all the filter coefficients that can be stored in the memory. This is a very effortless approach, and in general, such variable-coefficient (or reloadable) filters are optimum in a sense that the filter length for the particular frequency response specifications is minimum. However, when frequency response of a filter needs to be changed frequently, the bulky number of memory access operation would make updating routines of these filters time-consuming. In addition, when the number of frequency responses to be obtained is very large, huge memory is required to store all the filter coefficients corresponding to all the desired responses.

1.1 OBJECTIVE

Main objective is to increase the cut off frequency range and reduces the noise in the required signal.
In the existing system the filter coefficient can be fixed. In the proposed system the coefficients can be generated by different operations. When the frequency response of filter can be changed frequently. Huge memory is required to store all the filter coefficients.

Proposed MSFT filter has two significant changes compared to the FT filter. First, apply the low-pass to high-pass transformation to the prototype filter before applying the second-order frequency transformation. Second, apply the one-to-one mapping condition between the frequency variables to obtain a two band response.

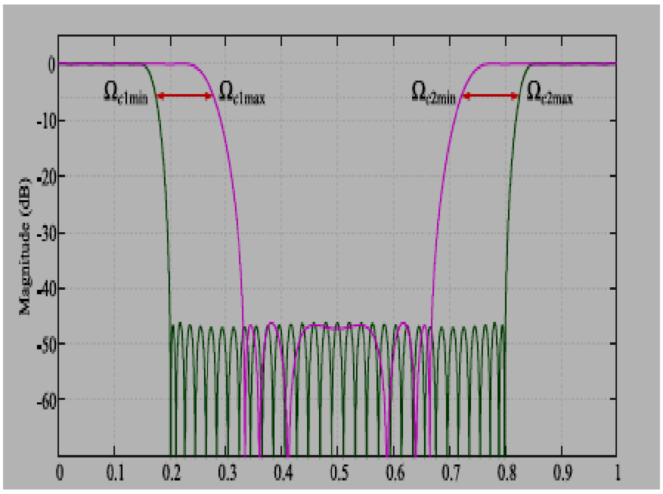


Fig-1: Normalized frequency

The response HL0 obtained from the MSFT filter & the cut off frequency range is 0.35π

2. EXISTING SYSTEM

2.1 FIR FILTERS WITH POWERS-OF-TWO COEFFICIENTS

FIR filters with powers-of-two coefficients is given away that the exponents of filter coefficients can be represent by the canonical signed digit code with *M* ternary digits can be selected from some subsets of {0,1,. . . , M-l}..

Implementation FIR filters with powers of-two coefficients, which are often referred to as 2PFIR filters, have received considerable attention in digital signal processing. By employing only those coefficients that are sums and differences of signed powers-of-two, each multiplication in 2PFIR filtering can be replaced with simple shift-and-add operations. Implementation of a 2PFIR filter is particularly efficient when its coefficients are fixed for dedicated applications. The main disadvantage is coefficients are fixed.

A new hardware-efficient architecture for programmable FIR filters, are efficient high-speed filter architectures, much of this work has focused on filters with fixed coefficients, such as Canonical Signed Digit coefficient filter architectures, multiplierless designs, or memory-based designs. In this paper, we focus on digit-serial, high-speed architectures with programmable coefficients. To achieve high performance goals, we consider both of algorithm level and architecture implementation level of FIR filters. In algorithm level, we reformulate the FIR formulation in bit-level and take the associative property of the addition in both the digit-serial multiplications and filter formulations.
Variable finite impulse response (FIR) filters (FIR filters whose frequency response can be changed on-the-fly, based on the beloved specifications) are useful in wireless communications, adaptive systems, audio signal processing, and biomedical applications.

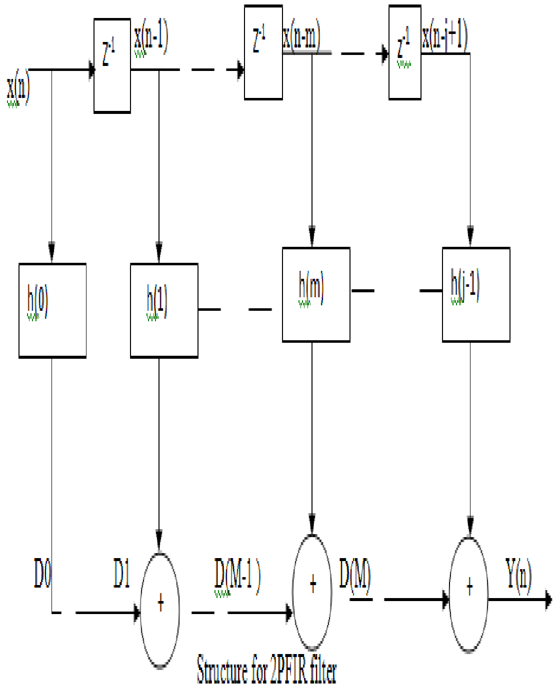


Fig -2.1: FIR filters with powers-of-two coefficients

2.2PROGRAMMABLE FIR FILTERS

Digital signal processing and advances in VLSI technologies have had a great force on the application domains of electronics. Amid the DSP applications, finite impulse response (FIR) filters are important building blocks. Recently, because of increasing demand for video signal processing and transmission, high speed and high-order programmable FIR filters have repeatedly been applied for performing adaptive pulse shaping and signal equalization on the received data in real time .F or this reason, an efficient VLSI architecture for high-speed programmable FIR filters is required. But, high-speed high-order programmable filters are complicated to be implemented efficiently for the reason that of the high implementation cost and the programmability requirements. The order of the filter is given by the number of poles, which is another way of saying the number of roots of the polynomial.

3. PROPOSED SYSTEM

The frequency response of an FIR filter can be altered by completely changing its coefficients or by modifying the impulse response using different operations. In the case of programmable digital filters, the preferred frequency responses are obtained by updating all the filter coefficients that are stored in the memory. This is a very simple advance, and in general, such variable-coefficient filters are optimum in a sense that the filter length for the particular frequency response requirements is minimum. When frequency response of a filter needs to be changed frequently, the large number of memory access operations would make updating routines of these filters time-consuming.When the number of frequency responses to be obtained is very large, huge memory is required to store all the filter coefficients corresponding to all the desired responses.

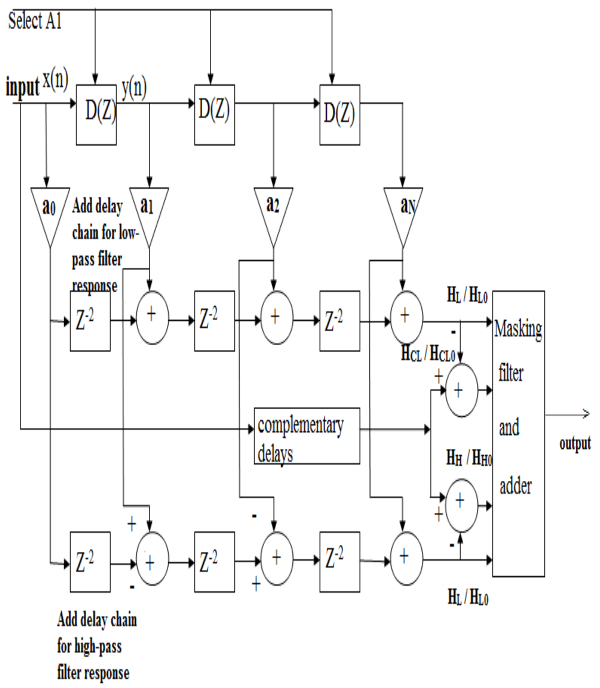


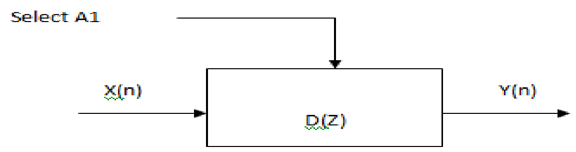
Fig- 3:proposed MSFT Filter

In the proposed system design the variable filter each delay of the filter structure is replaced by M delays to obtain a multiband response and desired band is extracted using a frequency masking filter. In the coefficient decimation method , the impulse response of the fixed coefficient filter is modified by retaining every Dth coefficient of the filter and also replacing the lingering coefficients .In this technique the coefficient can be fixed by the range of frequency.

3.1 SECOND-ORDER FILTER DESIGN

Second Order Filters which are also referred to as VCVS filters, because the op-amp is used as a Voltage Controlled Voltage Source amplifier, are another important type of active filter design because along with the active first order RC filters looked at previously, higher order filter circuits can be designed using them. First order filters can be easily converted into second order filters simply by using an additional RC network within the input or feedback path. Define the second order filters as simply being, two 1st-order filters cascaded together with amplification.

The second order filter has wider range of cut off frequency range compare to the frequency transformation filter and frequency transformation filter with combined co-efficient technique. Second Order Filters which are also referred to as VCVS filters, because the op-amp is used as a Voltage Controlled Voltage Source amplifier, are another important type of active filter design because along with the active first order RC filters looked at previously, higher order filter circuits can be designed using them. First order filters can be easily converted into second order filters simply by using an additional RC network within the input or feedback path.



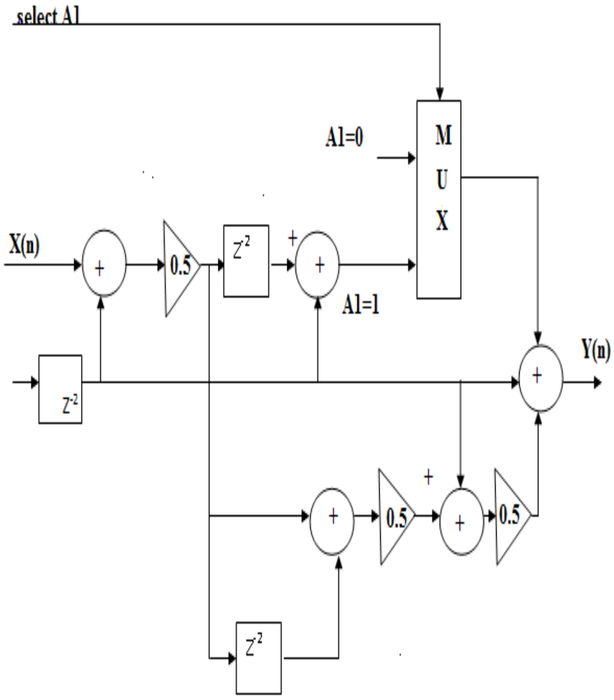


Fig-3.1:second order filter design

4. SIMULATION RESULT

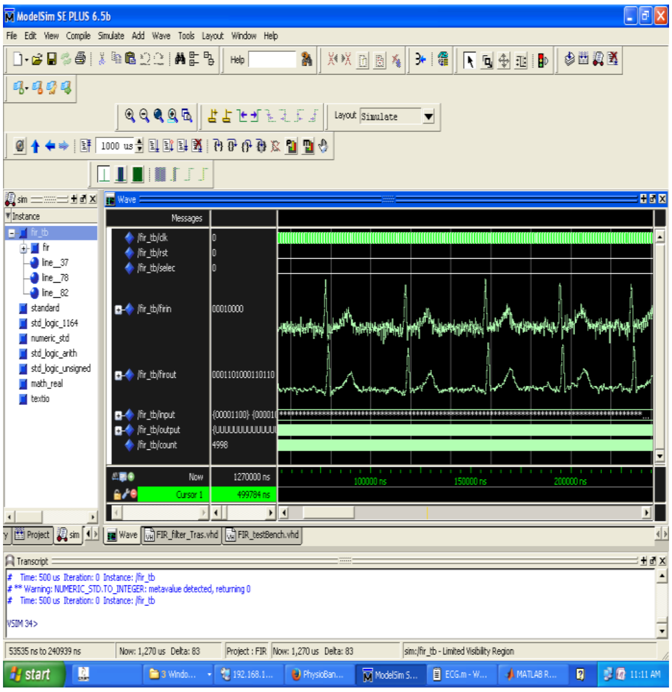


Fig- 4: MSFT filter response

In that proposed system MSFT filter can increase the cut off frequency range of 3.1 &1.23 compared to the existing system the cut off frequency range can be increased.

5. CONCLUSION

The modifications to the second order frequency transformation based filter to increase the cutoff frequency range of low-pass filter responses.The second-order frequency transformation and coefficient decimation based filter (FTCDM filter) has wider compared to the FT filter, but the ratio of transition bandwidths of the transformed and prototype filters is very high. The proposed modified second-order frequency transformation based filter (MSFT filter) has significantly wider cut off frequency range compared to FT and FTCDM filters.

6. FUTURE WORK

The future work will be based on increasing the range of the frequency transformation based filters. The frequency transformation based filters range increased means automatically obtain the low-pass responses over the entire Nyquist band.

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