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# A STUDY ON LOW POWER RECONFIGURABLE FIR FILTERS WITH DYNAMIC CHANGE IN FILTER ORDER

**[1]PRIYA STALIN and [2] C. ARUN**
[1]Assistant Professor,Department of Electronics and Communication Engineering,
Sree Sastha Institute of Engineering and Technology, CHENNAI.
[2]Professor, Department of Electronics and Communication Engineering
RMK College of Engineering and Technology, CHENNAI.
E-mail: priyastalinme@gmail.com

**ABSTRACT**

An architectural approach to design low power reconfigurable finite impulse response (LPRFIR) filter. The LPRFIR is well suited when the filter order is fixed and not changed for particular applications and efficient trade-off between power savings and filter performance can be implemented using the proposed architecture. Generally, FIR filter has large amplitude variations in input data and coefficients. Considering the amplitude of both the filter coefficients and inputs, proposed FIR filter dynamically changes the filter order. Mathematical analysis on power savings and filter performance degradation and its experimental results shows that the proposed approach achieves significant power savings without seriously compromising the filter performance. The power savings is up to 20.5% with minor performance degradation and the area overhead of the proposed scheme is less than 5.3% compared to the conventional approach.

**Keywords:** *Approximate filtering, low power filter, reconfigurable design, high speed filter.*

## 1. INTRODUCTION

**T**HE demand for low power digital signal processing (DSP) systems has increased due to explosive growth in mobile computing and portable multimedia applications . One of the most widely used operations performed in DSP is finite impulse response (FIR) filtering. The input-output relationship of the linear time invariant (LTI) FIR filter can be expressed as the following equation:

$$Y(n)=\sum_{k=0}^{N-1} c_k x(n-k) \quad (1)$$

where N represents the length of FIR filter, c *k*the *kth* coefficient, and *x(n – k)* the input data at time instant*(n-k)* . In many applications, in order to achieve high spectral containment and/or noise attenuation, FIR filters with fairly large number of taps are necessary. Many previous efforts for reducing power consumption of FIR filter generally focus on the optimization of the filter coefficients while maintaining a fixed filter order [1]–[3]. In those approaches, FIR filter structures are simplified to add and shift operations, and minimizing the number of additions/subtractions is one of the main goals of the research. However, one of the drawbacks encountered in those approaches is that once the filter architecture is decided, the coefficients cannot be changed; therefore, those techniques are not applicable to the FIR filter with programmable coefficients. Approximate signal processing techniques [4] are also used for the design of low power digital filters [5], [6]. In [5], filter order dynamically varies according to the stop band energy of the input signal. However, the approach suffers from slow filter-order adaptation time due to energy computations in the feedback mechanism. Previous studies in [6] show that sorting both the data samples and filter coefficients before the convolution operation has a desirable energy-quality characteristic of FIR filter. However, the overhead associated with the real-time sorting of incoming samples is too large. Reconfigurable FIR filter architectures are previously proposed for low power implementations [7]–[9] or to realize various frequency responses using a single filter [10]. For low power architectures, variable input word-length and filter taps [7], different coefficient word-lengths [8], and dynamic reduced signal representation [9] techniques are used. In those works, large overhead is incurred to support reconfigurable schemes such as arbitrary nonzero digit assignment [7] or programmable shift [8]. In this paper, we propose a simple yet efficient low

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power reconfigurable FIR filter architecture, where the filter order can be dynamically changed depending on the amplitude of both the filter coefficients and the inputs. In other words, when the data sample multiplied to the coefficient is so small as to mitigate the effect of partial sum in FIR filter, the multiplication operation can be simply canceled. The filter performance degradation can be minimized by controlling the error bound as small as the quantization error or signal to noise power ratio (SNR) of given system. The primary goal of this work is to reduce the dynamic power of the FIR filter, and the main contributions are summarized as follows. 1) A new reconfigurable FIR filter architecture with real-time input and coefficient monitoring circuits is presented. Since the basic filter structure is not changed, it is applicable to the FIR filter with programmable coefficients or adaptive filters. 2)We provide mathematical analysis of the power saving and filter performance degradation on the proposed approach. The analysis is verified using experimental results, and it can be used as a guideline to design low power reconfigurable filters. The rest of the paper is organized as follows. In Section II, the basic idea of the proposed reconfigurable filter is described. Section III presents the reconfigurable hardware architecture and circuit techniques used to implement the filter. Discussions on the design considerations and mathematical analysis of the proposed reconfigurable FIR filter are presented in Section IV. Section V shows the numerical results, followed by conclusions in Section VI.

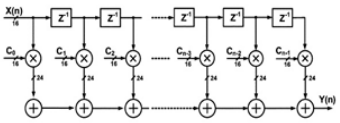

*Figure 1.1: Architecture Of Direct Form FIR Filter*

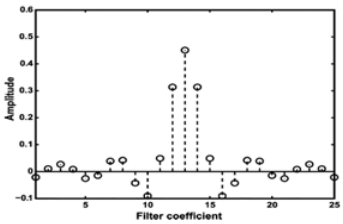


*Figure 1.2: Amplitude Of The 25-Tap Equi-Ripple Filter Coefficient*

## 2. RECONFIGURABLE FIR FILTER TO TRADE OFF FILTER PERFORMANCE AND COMPUTATION ENERGY

As shown in Figure. 1.1, the weighted values of input sequences sumed up in the FIR Filter operation is known as convolution sum. These are frequently used to implement selection of frequency such as low-pass, high-pass, or band-pass filters. Commonly, the result of summation and its related power of FIR filter are directly proportional to the filter order. The changes of filter order by turning off some of the booth multipliers are done to save the power in it. Even though we save the power, performance degradation should be carefully considered. when we change the filter order. Figure.1.2 exemplary shows the coefficients of a typical 25-tap low-pass FIR filter. The coefficient in centre has the largest value—the coefficient $c_{12}$ has the largest value in the 25-tap FIR filter and the amplitude of the coefficients generally decreases as they become more distant from the centre tap. The data inputs of the filter, which are multiplied with the coefficients also have large variations in amplitude. Therefore, the basic idea is that if the amplitudes of both the data input and filter coefficient are small, the multiplication of those two numbers is proportionately small; thus, turning off the booth's multiplier which has negligible effect on the filter performance. For example, since two's complement data format is widely used in the DSP applications, if one or both of the booth's multiplier input has negative value, multiplication of two small values give rise to large switching activities, which is due to the series of 1's in the MSB part. By canceling the multiplication of two small numbers, considerable power savings can be achieved with negligible filter performance degradation. In the fixed point arithmetic of FIR filter, full operand bit widths of the booth's multiplier outputs is not generally used. In other words, as shown in Figure.1.1, when the bit-widths of data inputs and coefficients are 16, the booth's multiplier generates 32-bit outputs. However, considering the circuit area of the following adders, the LSBs of booth's multipliers outputs are usually truncated or rounded off, (e.g., 24 bits are used in Figure.1.2) which incurs quantization errors. When we turn off the booth's multiplier in the FIR filter, if we can carefully select the input and coefficient amplitudes such that the multiplication of those two numbers is as small as the quantization error, filter performance degradation can be made negligible .In the following, we denote threshold of input and threshold of coefficient as $x_{th}$ and $c_{th}$, respectively.

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By threshold, we mean that when the filter input *x(n)*and coefficient $c_k$ are smaller than $x_{th}$ and $c_{th}$, respectively, the multiplication is canceled in the filtering operation. When we determine $x_{th}$ and $c_{th}$, the trade-off between filter performance and power savings should be carefully considered.

## 3. FIR FILTER ARCHITECTURE

In this section, we present a direct form (DF) architecture of the reconfigurable FIR filter, which is shown in Figure. 3.1. The speed of the filter can be increased significantly, by replacing the conventional multiplier by a booth's multiplier. In order to monitor the amplitudes of input samples and cancel the right multiplication operations, amplitude detector (AD) in Figure 3.2 is used. When the absolute value of *x(n)* is smaller than $x_{th}$ the threshold , the output of AD is set to "1". The design of AD is dependent on the input threshold $x_{th}$, where the fan in's of AND and OR gate are decided by $x_{th}$. If it has to be changed adaptively due to designer's considerations, AD can be implemented using a simple comparator. Dynamic power consumption of CMOS logic gates is a strong function of the switching activities on the internal node capacitances. In the proposed reconfigurable filter, if we turn off the booth's multiplier by considering each of the input amplitude only, then, if the amplitude of input *x(n)* abruptly changes for every cycle, the booth's multiplier will be turned on and off continuously, which incurs considerable switching activities. Booth's multiplier control signal decision window (MCSD) in Figure.3.1 is used to solve the switching problem. Using *ctrl* signal generator inside MCSD, the number of input samples consecutively smaller than $x_{th}$ are counted and the booth's multipliers are turned off only when *m* consecutive input samples are smaller than $x_{th}$. Here, *m* means the size of MCSD [in Figure 3.1, is equal to 4].

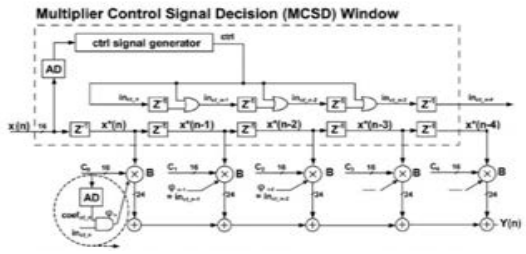


*Figure 3.1: Proposed Reconfigurable FIR Filter Architecture*

Note :ADs and AND gates for each coefficient monitoring are required only in adaptive filter case The above figure shows the *ctrl* signal generator design. As an input smaller than $x_{th}$ comes in and AD output is set to "1", the counter counts up. When the counter reaches *m*, the *ctrl* signal in the figure3.1 changes to "1", which indicates that *m* consecutive small inputs are monitored and the

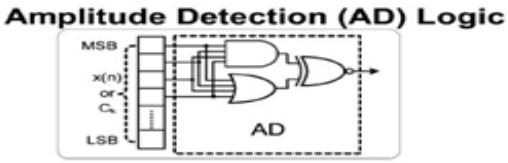


*Figure 3.2: Amplitude Detection Logic(AD)*

booth's multipliers are ready to turn off. One additional bit *inct_n*, in Figure. 3.1 is added and it is controlled by *ctrl*. The *inct_n* accompanies with input data all the way in the following flip-flops to indicate that the input sample is smaller than $x_{th}$ and the multiplication can be canceled when the coefficient of the corresponding booth's multiplier is also smaller than $c_{th}$. *inct_n* signal is set inside MCSD, the signal does not change outside MCSD and holds the amplitude information of the input. A delay component is added in front of the first tap for the synchronization between *x*(n)* in Figure. 3.2 since one clock latency is needed due to the counter in MCSD. In case of adaptive filters, additional ADs for monitoring the coefficient amplitudes are required as shown in Figure. 3.2 .However, in the FIR filter with fixed or programmable coefficients, since we know the amplitude of coefficients ahead, extra AD modules for coefficient monitoring are not needed, extra AD modules for coefficient monitoring are not needed. When the amplitudes of input and coefficient are smaller than the threshold, the booth's multiplier is turned off by setting $\varphi_n$ signal in Figure . 3.1 to "1". Based on the simple circuit technique [11] in Figure . 3.2 the booth's multiplier can be easily turned off and the output is forced to "0". As shown in the figures, when the control signal $\varphi_n$ is "1", since PMOS turns off and NMOS turns on, the gate output is forced to "0" regardless of input. When $\varphi_n$ is "0", the gate operates like standard gate. Only the first gate of the booth's multiplier is modified and once the $\varphi_n$ is set to "1", there is no switching activity in the following nodes and booth's multiplier output is set to "0". The area overheads of the proposed reconfigurable filter are flip-flops for *inct_n* signals, AD and *ctrl* signal generator inside MCSD and the modified gates in Figure 3.1 for turning off booth's multipliers. Those overheads can be implemented using simple logic gates, and a single AD is needed

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for monitoring input *x(n)* as specified in Figure . 3.1. Consequently, the overall circuit overhead for implementing reconfigurable filter is as small as a single booth's multiplier.

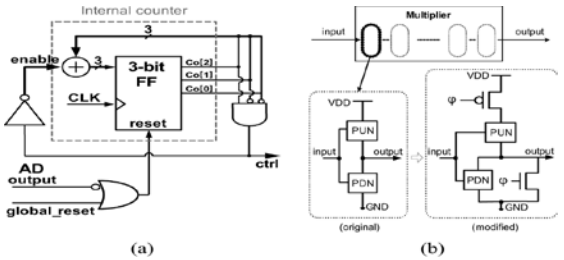


*Figure 3.3(A) Schematic Of Ctrl Signal Generator. Internal Counter Sets Ctrl Signal To "1" When All Input Samples Inside MCSD Are Smaller Than Xth (M= 4 Case).*

*Figure 3.3 (B) Modified Gate Schematic To Turn Off Booth's Multiplier.*

## 4. DESIGN CONSIDERATIONS AND MATHEMATICAL ANALYSIS ON THE RECONFIGURABLE FIR FILTER

In this section, we present design considerations on the proposed reconfigurable FIR filter. Mathematical analysis which describes the trade-off between power savings and filter performance degradation

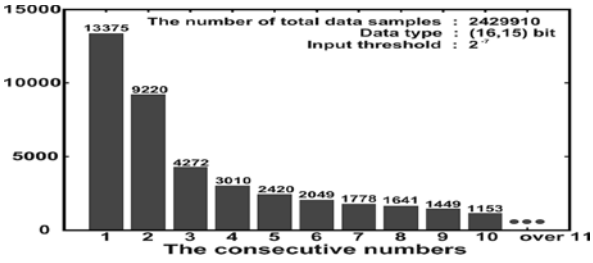


*Figure 4.1 Graph Between The Consecutive Numbers And Input Threshold*

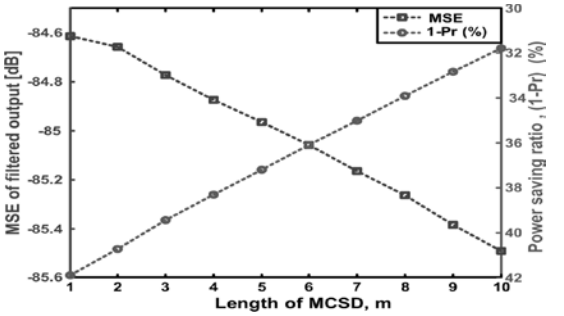


*Figure 4.2 Graph Between Length Of MCSD And MSE Of Filtered Output.*

### 4.1.Design Considerations

In following discussions, as a metric of power savings, we use the power consumption ratio,P*r* , which means the ratio of the reconfigurable filter power consumption to the conventional filter power(*Preconf*/*Pconven*). As a measure of filter performance degradation, we use mean-square error (MSE) between the proposed reconfigurable filter output and original filter output. The most important factors that have a large effect on the proposed filter performance and power consumption are $x_{th}$ and *cth*. When $x_{th}$ and $c_{th}$are set too large, it can give rise to large power savings with considerable distortion in the filter output. On the other hand, if $x_{th}$ and $c_{th}$are too small, power savings become trivial. The other one to be considered is *m*,the length of MCSD. Figure 4.1 shows the number of input samples whose *m*(*x* axis) consecutive input values are smaller than input threshold. The input signals used in the simulation are more than ten samples of sounds and speeches. In Figure.4.2, if we choose a specific value in the axis, the total number of canceled multiplications is the accumulated number of samples from the selected value to the right. Therefore, if *m* becomes larger, the number of input samples that make booth's multipliers turned off decreases; then, power reduction becomes smaller and filter performance degradation becomes lower as well. Figure 4.2 shows the trade-off between the power saving ratio *(1-Pr)* and the MSE for different *m* values in case of a 75-tap equi -ripple filter with $x_{th}$ and $c_{th}$ of $2^{-7}$.

### 4.2. Mathematical Analysis

Mathematical modeling's on the power savings and performance degradation of the proposed reconfigurable FIR filter are presented in this subsection. Assuming the input signal as stationary random Markov process as commonly used in communication systems [12], where the future state is independent of its past states and conditionally dependent only on the present state, power saving ratio, *(1-Pr)* , can be expressed as,

$$1\text{-}P_r = \mathrm{Pr}_c \mathrm{Pr}_x P^{m-1}{}_{cut2cut} \quad (2)$$

Where

$$P_{rc}(=\mathrm{Pr}\{|C_k| \le C_{th}\}) \text{ and} \quad (3)$$

$$Prx(=Pr\{|x(n-k)| \le X_{th}\}) \quad (4)$$

are the probability that the amplitude of the coefficient and input signal are less than the given thresholds, i.e.,

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$c_k \le c_{th}$ ,and $|x(n-k) \le x_{th}|$, **(5)**

respectively. $P_{cut2cut}$ is the conditional probability meaning that future state of input samples is smaller than $x_{th}$ under present state being also smaller than $x_{th}$ ,

$$\mathbf{Pr\{|\ x(n)\ | \le x_{th} \ ||\ x(n\text{-}1)\ | \le x_{th}\ \},\ \forall_n} \quad \textbf{(6)}$$

note that power savings of the proposed reconfigurable filter are directly affected by the probabilities of inputs and coefficients being smaller than *xth* and *cth* respectively, as well as the conditional probability P*cut2cut* which is dependent upon time correlation characteristics of the input signal. It is also a function of the MCSD window size *m*. As a metric of filter performance degradation,

the MSE of the filter output $\sigma_e^2$ is described as,

$$\boldsymbol{\sigma}_{e} = \mathbf{Pr}_c\ \boldsymbol{\sigma}_x \quad \textbf{(7)}$$

$$\sigma_e^{\,2} = \mathbf{Pr}_c^{\,2}\sigma_x^{\,2} \sum_{k=0}^{N-1} \sum_{h=0}^{N-1} \quad \textbf{(8)}$$

$$\times C_h C_k r_x(h\text{-}k)\, prob_{cut}(x_{th,}\ |h\text{-}k|,\ m)$$

Where $\sigma_x^2 (= E\{x^2(n)\})$ is the average input signal power, $r_x(h\text{-}k)$ is time correlation of input signal spaced by *(h-k)*, and $k \epsilon C_{tc}$ denotes a set of *k* where the $k_{th}$ filter coefficient is smaller than $c_{th}$ , i.e, $|c_k| \le c_{th.}$ .

Here, $prob_{cut}(x_{th}$ , $|h\text{-}k|$ , *m)* is the probability that the input samples at *(n-h), (n-h-1),…,(n-h-m+1)* and *(n-k), (n-k-1),…,(n-k-m+1)* are smaller than $x_{th}$, which is represented as

$$\mathbf{Prob_{cut}(x_{th,}\ |h\text{-}k|\ ,m);}$$

$$\mathbf{Prob_{cut}(x_{th}\ ,\ |h\text{-}k|\ ,m)}$$

$$\approx \begin{cases} Pr_x P_{cut2cut}^{\,m-1} & ,when\ |h\text{-}k| > m \\ Pr_x P_{cut2cut}^{\,|h-k|+(m-1)} & ,when\ |h\text{-}k| \le m \end{cases} \quad \textbf{(9)}$$

Above equation shows that the MSE is mainly determined by the filter coefficient, input signal power, auto-correlation of the input signal, and probability of input samples being smaller than $x_{th}$.

## 5. NUMERICAL RESULTS

### 5.1. Reconfigurable FIR Filter Specifications

Following are the specifications on the FIR filters implemented. Input sequence and coefficients are 16-bit data with fractional part of 15 bit. Hence, the data range is [-1,1]. The outputs of the booth's multiplier in the FIR filter are quantized into 16 bit and the final filter output is 24 bit. We use $2^{-7}$ as an input threshold ,$x_{th}$, and coefficient threshold, $c_{th}$ . The values of MCSD window length, *m* , are differently assigned for the filters. The values of $x_{th}$ ,$c_{th}$ and can be controlled by users considering the performance degradation and power savings trade-off presented in Section IV-B.

### 5.2. Numerical Results

The proposed reconfigurable FIR filters are verilog coded and synthesized using TSMC 0.25µm CMOS technology. The first gate of each of the booth's multipliers is replaced with the modified gates as shown in Figure 5.1. Power consumption is measured in the spice level simulations using [13] an operating frequency of 100-MHz, 2.5-V supply voltage. Table I shows the average power saving ratio, and average MSE of filter output for the implemented FIR filters.

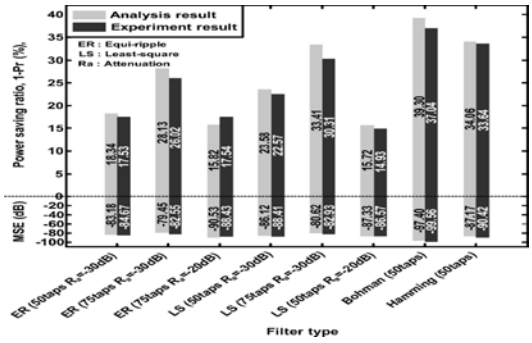


*Figure 5.1 Graph Between Filter Type And Power Saving Ratio.*

(MSE) was also compared with the experimental results for various reconfigurable filters as seen in Figure 5.1. It is noticeable from the figure that the differences between the mathematical modeling and experimental results are very small. To analyze the filter performance degradation, we define signal power to MSE ratio of the filter output (SMR) considering the effective ratio of the desired signal and the distorted error signal power as

$$\mathbf{SMR} = \sigma_y^{\,2} / \sigma_x^{\,2}$$

$$\approx \sum_{k=0}^{N-1} \sum_{h=0}^{N-1} c_k c_h r(h\text{-}k)$$

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$$\mathbf{Pr}_c \sum_{k=0,\ k \in Ctc}^{N-1} \sum_{h=0,\ h \in Ctc}^{N-1} c_h c_k r_x(h\text{-}k)\, prob_{cut}(x_{th}\ |h\text{-}k|, m) \quad \textbf{(10)}$$

Table II also presents the SMR results of various filters. For most of the cases, the SMR is

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larger than 45 dB, meaning that MSE is almost ignorable. For given applications, if SMR is comparable or larger than the signal to quantization error power ratio or the SNR of a given system, which are usually less than 30 dB [12], the performance degradation of proposed reconfigurable FIR filter can be considered negligible. Power saving ratio of each filter is normalized with the number of taps, the precision of input samples and coefficients, process technology, supply voltage, and clock frequency using the following equation[7]:

$$P(tap) = \frac{Total\ power}{\#taps} \times \frac{16}{\#bits\ coeff} \times \frac{16}{\#bits\ sample} \times \frac{(2.5|vdd)^2}{Tech} \times 0.25 \times \frac{Clk\ freq}{100} \quad (11)$$

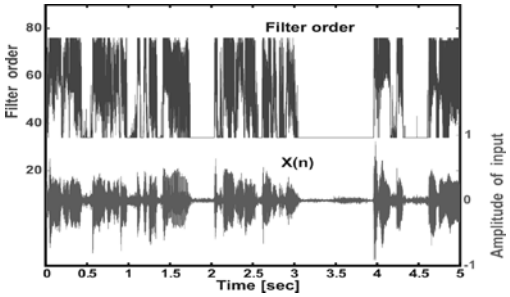


*Figure 5.2: Graph Between Order Of Filter And Time Measured In Seconds.*

*Table I: Filter Parameters Are Discussed Such As Delay, Power Consumption, Increase In Area, Increase In Speed And Power Savings For Different Parameters*

| Filter Parameters | Delay | Power Consumption | Increase In area | Increase In speed | Pwr Savings % |
|---|---|---|---|---|---|
| Existing Sys | 4.210ns | 334mW | 5.3% | 4.67% | 15% |
| Proposed Sys | 4.015ns | 285mW | | | |

*Table II: Average Power Saving Ratio (1-Pr)(%), The Average MSE And SMR For Various Filter Types In Speech Signal Case With 25taps.*

| Type | 25taps($\omega_p$=0.10, $\omega_s$=0.38,$R_a$= -70dB) | | | |
|---|---|---|---|---|
| | m | 1-$P_r$(%) | MSE | SMR(dB) |
| Equi-ripple | 3 | 19.18% | -88.74 | 49.30 |
| least squares | 3 | 19.78% | -88.74 | 49.30 |
| Type | 25taps($\omega_p$=0.10, $\omega_s$=0.20,$R_a$= -30dB) | | | |
| | m | 1-$P_r$(%) | MSE | SMR(dB) |
| Equi-ripple | 3 | 7.61 % | -95.12 | 55.34 |
| least squares | 3 | 7.02% | -94.33 | 54.54 |
| Type | 25taps($\omega_p$=0.10, $\omega_s$=0.20 ,$R_a$= -20dB) | | | |
| | m | 1-$P_r$(%) | MSE | SMR(dB) |
| Equi-ripple | 3 | 14.68% | -92.17 | 51.86 |
| least squares | 3 | 7.02% | -93.61 | 53.90 |
| Type | 25taps($\omega_c$=0.12) | | | |
| | m | 1-$P_r$(%) | MSE | SMR(dB) |
| Equi-ripple | 4 | 24.54% | -91.42 | 51.63 |
| least squares | 3 | 24.94% | -99.56 | 59.79 |

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| Type | 50taps($\omega_p$=0.10, $\omega_s$=0.26,$R_a$= -70dB) | | | |
|---|---|---|---|---|
| | m | 1-$P_r$(%) | MSE | SMR(dB) |
| Equi-ripple least squares | 3 | 31.84% | -88.29 | 48.75 |
| | 3 | 31.83% | -88.49 | 48.95 |
| Type | 50taps($\omega_p$=0.10, $\omega_s$=0.155,$R_a$= -30dB) | | | |
| | m | 1-$P_r$(%) | MSE | SMR(dB) |
| Equi-ripple least squares | 3 | 17.53 % | -84.67 | 45.16 |
| | 3 | 22.57% | -88.41 | 48.79 |
| Type | 50taps($\omega_p$=0.10, $\omega_s$=0.13 ,$R_a$= -20dB) | | | |
| | m | 1-$P_r$(%) | MSE | SMR(dB) |
| Equi-ripple least squares | 3 | 9.73% | -94.22 | 54.92 |
| | 3 | 14.93% | -86.57 | 46.96 |
| Type | 50taps($\omega_c$=0.12) | | | |
| | m | 1-$P_r$(%) | MSE | SMR(dB) |
| Hamming | 4 | 33.64% | -90.42 | 50.72 |
| Bohman | 3 | 37.04% | -86.47 | 46.75 |

*Table III: Average Power Saving Ratio (1-Pr)(%), The Average MSE And SMR For Various Filter Types In Speech Signal Case With 50 Taps*

*$\Omega_p$: Normalized Passband*

*$\Omega_s$ : Stopband Edge Frequency*

*$\Omega_c$ : Cut-Off Frequency $R_a$ : Stopband Attenuation*

Our proposed filter shows larger power savings than the filters in [7] and [9]. The reconfigurable filter in [5] consumes less power than our proposed filter; however,

| Type | 75taps($\omega_p$=0.10, $\omega_s$=0.20,$R_a$= -70dB) | | | |
|---|---|---|---|---|
| | m | 1-$P_r$(%) | MSE | SMR(dB) |
| Equi-ripple least squares | 3 | 39.45% | -84.77 | 45.19 |
| | 3 | 41.19% | -86.29 | 46.71 |
| Type | 75taps($\omega_p$=0.10, $\omega_s$=0.135,$R_a$= -30dB) | | | |
| | m | 1-$P_r$(%) | MSE | SMR(dB) |
| Equi-ripple least squares | 5 | 26.02 % | -82.55 | 42.71 |
| | 5 | 30.31% | -82.93 | 43.25 |
| Type | 75taps($\omega_p$=0.10, $\omega_s$=0.12 ,$R_a$= -20dB) | | | |
| | m | 1-$P_r$(%) | MSE | SMR(dB) |
| Equi-ripple least squares | 3 | 17.64% | -88.43 | 48.60 |
| | 5 | 28.60% | -80.20 | 40.57 |
| Type | 75taps($\omega_c$=0.12) | | | |
| | m | 1-$P_r$(%) | MSE | SMR(dB) |
| Hamming | 4 | 41.98% | -80.78 | 41.08 |
| Bohman | 3 | 42.42% | -87.00 | 47.37 |

*Table IV: Average Power Saving Ratio (1-Pr)(%), The Average MSE And SMR For Various Filter Types In Speech Signal Case With 75 Taps*

MSE is even larger. Phonetic signal processing system is a good application to use the proposed reconfigurable filtering approach. Voice signal is usually composed of considerable portions of data samples with small amplitude. Though we focused on the FIR filters with fixed coefficients in this work, our proposed approach can be extended to the adaptive filter cases, where both data inputs and coefficients amplitude should be monitored simultaneously.

## 6. CONCLUSION

In this paper, we propose a low power reconfigurable FIR filter architecture to allow efficient trade-off between the filter performance

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and computation energy. In the proposed reconfigurable filter, the input data are monitored and the booth's multipliers in the filter are turned off when both the coefficients and inputs are small enough to mitigate the effect on the filter output. Therefore, the proposed reconfigurable filter dynamically changes the filter order to achieve significant power savings with minor degradation in performance. According to the mathematical analysis, power savings and filter performance degradation are represented as strong functions of MCSD window size, the input and coefficient thresholds, and input signal characteristics. Numerical results show that the proposed scheme achieves power savings up to 41.9% with less than around 5.34% of area overhead with very graceful degradation in the filter output. The proposed approach can be applicable to other areas of signal processing, where a proper trade-off between power savings and performance degradation should be carefully considered. The idea presented in this paper can assist in the design of FIR filters and its implementation for low power applications.

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