Reconfigurable Filter Banks for Software Defined Radio Receivers – An Alternative Low Complexity Design to Conventional DFT Filter Banks

R. Mahesh and A. P. Vinod

School of Computer Engineering,
Nanyang Technological University,
Nanyang Avenue,
Singapore, 639798.
{mahe0001, asvinod}@ntu.edu.sg

Abstract

A new approach to implement computationally efficient reconfigurable filter banks for software defined radio receivers is presented in this paper. If the coefficients of a finite impulse response (FIR) filter are decimated by M, i.e., if every M-th coefficient of the filter is kept unchanged and remaining coefficients are replaced by zeros, a multi-band frequency response will be obtained. The frequency response of the decimated filter will have bands at centre frequencies at $2\pi k/M$, where k is an integer ranging from 0 to $M-1$. If these multi-band frequency responses are subtracted from each other or selectively masked using inherently low complex wide transition-band masking filters, we can obtain different low-pass, high-pass, bandpass, and bandstop frequency bands. The resulting filter bank, whose bands' centre frequencies are located at integer multiples of $2\pi/M$, is a low complexity alternative to the well known uniform Discrete Fourier Transform (DFT) filter banks. We also show that the proposed filter bank is more flexible and easily reconfigurable than the DFT filter bank. Furthermore, the proposed filter bank is able to receive channels of multiple standards simultaneously, where as separate filter banks would be required for simultaneous reception of multi-standard channels in a DFT filter bank based receiver.

1. Introduction

Software Defined Radio (SDR) is evolving as a promising technology in the area of wireless communications. The basic idea of SDR is to replace most of the analog signal processing in the transceivers with digital signal processing in order to provide the advantage of flexibility through reconfiguration or reprogramming [1]-[3]. This will enable different air-interfaces to be implemented on a single generic hardware platform. The SDR receiver typically employs a channelizer, which comprises of digital filter banks to extract multiple radio channels (frequency bands) from the wideband input signal for further processing [1, 2]. A first-order estimate of resources needed to implement a wideband receiver shows that the channelizer is the most computationally intensive part of a wideband receiver [4] since it operates at the highest sampling rate in the digital front-end. The compatibility of the channelizer with different communication standards is guaranteed by its reconfigurability. Therefore, a reconfigurable low complexity filter bank is a vital part for efficient implementation of SDR channelizers.

In this paper, we propose a new method for designing reconfigurable filter banks based on the multirate signal processing concept of decimation. In our design called coefficient decimation-based filter bank design, we first design an N-tap lowpass FIR filter as the modal filter. Subsequently, every M-th coefficient of the modal filter is retained and all other coefficients are replaced with zero values. The frequency response of the resulting decimated modal filter will have replicas of the passband of the modal filter at integer multiples of $2\pi/M$. The desired channels of different bandwidths can be extracted from the identical bandwidth spectrum replicas of the decimated modal filter using one or more of following operations - subtraction, frequency masking or complementary filtering. As all the desired channels are obtained using the same modal filter, a fixed-coefficient implementation is feasible for the modal filter, which leads to a low complexity filter bank. Furthermore, as coefficient decimation reduces the number of nonzero coefficients of the original modal filter to N/M, the complexity of the resulting filter bank is substantially reduced. The proposed design also has the advantage that the passband widths and the centre frequencies of passbands can be easily changed to extract the channels located almost anywhere in the available bandwidth of wideband signal. Thus the proposed coefficient decimated filter bank can be employed as a low complexity and highly flexible alternative to inherently less flexible and high complex DFTFB.

2. Design of Proposed Filter Bank

In this section, we present the proposed filter bank design. If the coefficients of an FIR filter are decimated by M, i.e., every M-th coefficient is kept unchanged and all others are replaced by zero values, we get a frequency response similar to images created during upsampling. Our definition of *coefficient decimation* is that unused coefficients (i.e., coefficients other than every M-th coefficient) are replaced by zero values as opposed to the conventional notion of discarding unused samples in the decimation of a signal.

The proposed design can be illustrated with the help of Fig. 1. The frequency response of the modal filter with normalized (with respect to sampling frequency) passband and stopband specifications of $f_p = 0.05$ and $f_s = 0.075$ is shown in Fig. 1(a). Let the passband and stopband ripple specifications (δ_p and δ_s respectively) are 0.1 dB and -55 dB. Fig. 1(b) represents frequency response for $M = 2$, i.e., the case when every 2nd coefficient is kept unchanged and remaining coefficients are replaced by zero values. Note that the frequency response is obtained by scaling the coefficients by $M = 2$. In the proposed CDFB implementation (architecture is explained later), this can be achieved by scaling the output of the filter by $M = 2$. This is possible because convolution holds good for $(M \times h) \otimes x = M \times (h \otimes x)$, where x is the input and h represent the filter coefficients. As seen from (7) and Fig. 1(b), for $M = 2$, the frequency responses are obtained at $2\pi k/2=\pi k$, for k=0 and 1. Similarly, Figures 1(c) and 1(d) represent the case $M = 3$ and $M = 4$ respectively. It can be seen from Figures 1(b) to 1(d) that the stopband attenuation reduces as M increases. But it should be noted that the transition band width remains unaltered for any M. Therefore, based on the desired stopband attenuation specification of the channel to be extracted, the original modal filter is designed with larger stopband attenuation keeping in account of the deterioration of the final filter bank's stopband attenuation. The complexity of the CDFB reduces as M increases. An N-tap FIR filter will have N multiplications whereas the filter whose coefficients are decimated by M will have only N/M multiplications. In our design example, the modal filter in Fig. 1(a) has 120 taps ($N = 120$) and consequently 120 multiplications, but for the case $M = 2$, the number of multiplications is only 60. Therefore, even if the initial modal filter in CDFB is designed with more taps (compared to the length of prototype filter in a DFTFB) taking into account of the stopband attenuation reduction after coefficient decimation, the effective filter length and overall multiplication complexity of the CDFB is less than that of DFTFB as coefficient decimation reduces filter length by a factor of M.

The proposed method can be extended to develop a filter bank to extract multiple frequency bands simultaneously, i.e., parallel reception of several channels. This can be illustrated as follows. If the frequency responses in Figures 1(a) to 1(d) are obtained simultaneously (using the architecture in Fig. 2), then by subtracting the outputs of Fig. 1(b) and Fig. 1(c) from Fig. 1(a) and the output of Fig. 1(d) from Fig. 1(b), different frequency bands located at integer multiples of $2\pi/M$ can be extracted. A generalized architecture for the proposed FB is shown in Fig. 2. The frequency responses in Figures 1(a) to 1(d) are obtained using the filter bank structure in Fig. 2 as outputs y_1 to y_4 respectively. Also, the responses in Figures 3(a) to 3(c) are obtained as outputs y_2-y_1, y_3-y_1 and y_4-y_2 in the architecture shown in Fig. 2. Thus the proposed CDFB is capable of extracting channels corresponding to the frequency responses in Figures 1(a) to 1(d) and 3(a) to 3(c) simultaneously without the need of any extra filters or modulation operations. Note that neither the passband width nor the transition band width is altered while obtaining all the above frequency responses. An alternative method to spectral subtraction for obtaining individual frequency responses from the multiband response is to employ frequency masking filters to mask out unwanted bands. In Fig. 2, it is shown that the channels y_{41C} and y_{42C} are obtained from the multiband response y_{4C} using masking filters HM1 and HM2 respectively. The masking filters HM1 and HM2 have wide transition band widths and therefore their complexities are low as they can be realized using short filters.

In our method, since the frequency responses (replicas) obtained are at integer multiples at $2\pi/M$, down sampling by a factor of M would avoid the need of a digital down converter to shift the bandpass and high pass channels to the baseband. This can be explained with the help of Fig. 4. In Fig. 4(a), the complete down conversion process is shown. The output of bandpass filter $H_k(z)$, which is located at $2\pi k/M$, is down converted to baseband by using exponential frequency shift function, $e^{-j2\Pi kn/M}$ (This can be achieved by employing modulator such as DFT). Then the sample rate is reduced using down sampler. Note that the only modulator outputs not discarded by the down sampler are those with time index $n=mM$, where m is an integer. But for these outputs, the modulator has the value, $e^{-j2\Pi kmM/M} = e^{-j2\Pi km} =1$, and thus it can be ignored. The resulting system is shown in Fig. 4 (b). The same methodology can be employed in the proposed filter bank to down convert the bandpass channels to baseband without the need of any down converter. Since all the bandpass channels obtained by the proposed filter bank are located at integer multiples at *2π/M*, down sampling by *M*, will shift all the bandpass channels to baseband.

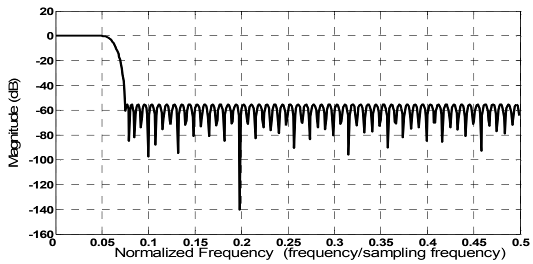

Figure 1(a). Frequency response of original Modal filter.

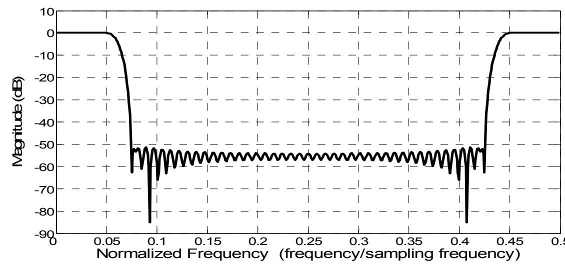

Figure 1(b). Frequency response of Modal filter with M=2.

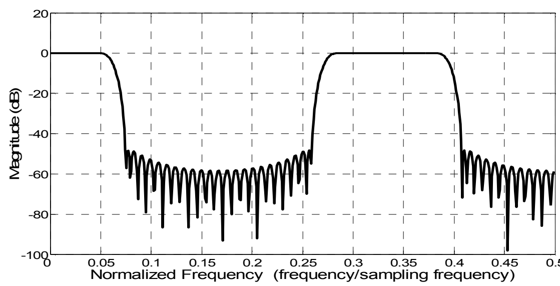

Figure 1(c). Frequency response of Modal filter with M=3.

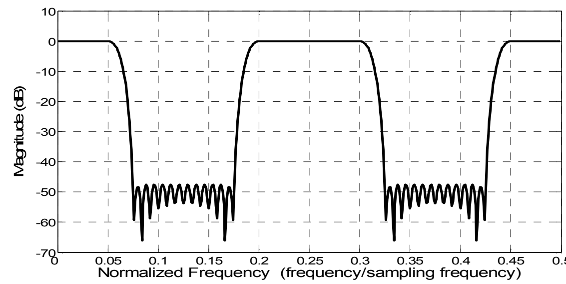

Figure 1(d). Frequency response of Modal filter with M=4.

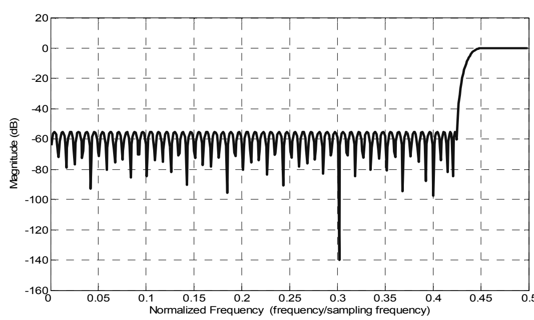

Figure 3(a). Frequency response at y_2-y_1.

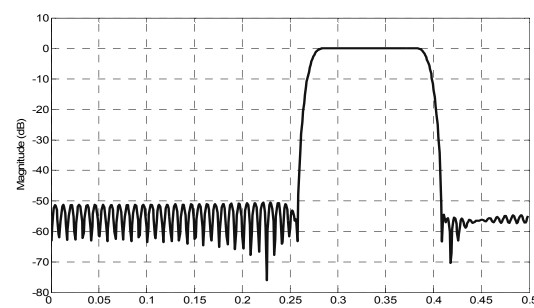

Figure 3(b). Frequency response at y_3-y_1.

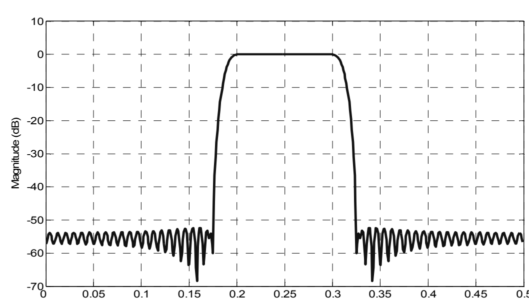

Figure 3(c). Frequency response at y_4-y_2.

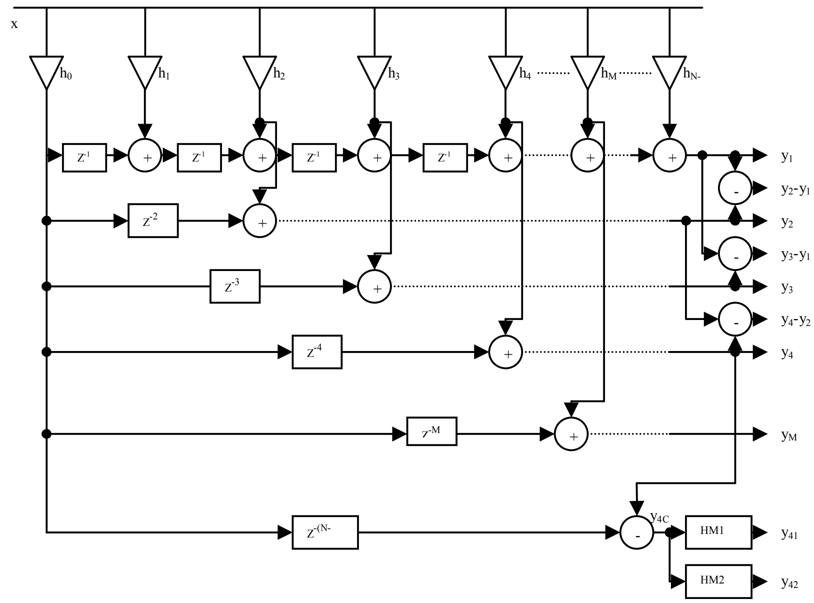

Figure 2. Architecure of the proposed filterbank.

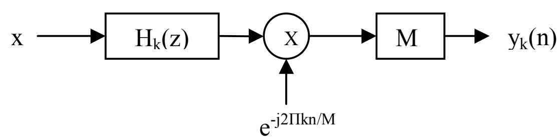

Figure 4(a). Down conversion to baseband after bandpass filtering.

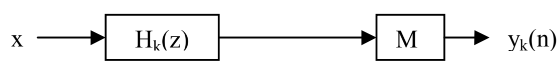

Figure 4(b). Modified Down conversion to baseband after bandpass filtering.

3. Conclusion

In this paper we have proposed a new methodology for the implementation of reconfigurable filter banks. We have shown that the proposed filterbank can be employed as a low complexity substitute to conventional DFTFB. It was also shown that, the proposed filterbank is easily reconfigurable compared to existing filter banks. In future, the design tradeoff of the proposed filterbank will be studied in a more detailed manner and the proposed filterbank will be implemented and tested on Virtex FPGA.

4. References

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