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An improved class of multiplierless decimation filters: Analysis and design ☆

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ABSTRACT

In this paper, we present a class of low-complexity decimation filters for oversampled discrete-time signals. The proposed class of filters improves the frequency response of classical comb filters in two respects. First, it introduces extra-attenuation around the so-called folding bands, i.e., frequency intervals whose spurious signals are folded down to baseband during the decimation process. Second, this class reduces the passband distortion via an effective droop-compensator block, thus increasing the passband of the decimation filters. Like comb filters, the proposed class can be realized through multiplierless architectures, which are also discussed thoroughly in the paper. Unlike comb filters, the proposed filters have superior spurious signal rejection and a greatly reduced droop in the signal passband. These features make the proposed filters suitable for multistage decimation applications, such as reconfigurable software radio receivers, as well as for decimating oversampled digital signals produced by $\Sigma\Delta$ A/D converters.

The paper discusses several useful techniques for designing the proposed filters in a variety of architectures with emphasis on non-recursive architectures. Design examples are discussed to highlight the key frequency features along with implementation issues aimed at reducing the computational complexity of the filters.

1. Introduction

Computationally efficient multistage decimation filters are key components in wide-band, multi-standard, reconfigurable receivers [1–5]. Multistage decimation filters [6,7] are also widely used to decimate signals oversampled by $\Sigma\Delta$ Analog-to-Digital (A/D) converters [8], and for digital down-converters as employed in digital receivers [9,10].

An essential block in a multistage decimation architecture is the comb filter, which can be effectively implemented with only additions and subtractions [11]. However, the magnitude response features poor attenuation around the folding bands and a considerable passband distortion that deteriorates the sampled signal. Several works in the literature have proposed a variety of solutions to improve the frequency response of classical comb filters, including a work addressing the use of comb filters in multirate applications [12].

In [13] the authors proposed the design of decimation filters based on the first 104 cyclotomic polynomials, which was later extended in [14] over the first 200 cyclotomic polynomials. A 3rd-order modified decimation sinc filter was proposed in [15]. The class of comb filters was then generalized in [16] whereby the authors proposed an optimization framework for deriving the optimal zero rotations of Generalized Comb Filters (GCFs) for any filter order and decimation factor M. In [17] generalized comb filters are implemented in non-recursive architecture by exploiting polyphase decomposition. In [18] the authors proposed a multiplierless architecture for the design of 3rd-order GCFs. In [19], the authors proposed a novel two-stage non-recursive architecture for the design of generalized comb filters.

In [20] authors proposed computational efficient architectures for classical comb filters used for multirate applications. Comb filters were used as constituent blocks in Kaiser and Hamming [21] sharpened structures in [22], while in [23] the authors proposed several architectures for comb filters including sharpened structures. In [24] the authors addressed the design of a novel two-stage sharpened comb decimator. In [25] the authors proposed novel decimation schemes for $\Sigma\Delta$ A/D converters based on Kaiser and Hamming sharpened filters. A novel multistage comb rotated sinc filter with sharpened response was proposed in [26]. In [27] the authors proposed a new decimation filter improving the frequency response of Cascaded-Integrator-Comb (CIC) Decimation Filters, while in [28] a simple method to compensate for the passband distortion of CIC decimation filters was proposed.

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A new cascaded modified CIC-cosine decimation filter was proposed in [29]. In [30] the authors addressed the synthesis of very sharp decimators and interpolators using the frequency-response masking technique. Finally, in [31] the authors recently proposed sharpening of CIC filters with a Chebyshev polynomial resulting in equi-ripple and wider controlled stopbands.

The main goal of this paper is to propose a reconfigurable, multiplierless, low-complexity decimation filter architecture that, compared to recursive implementation of comb filters, i.e., CIC architectures, features 1) improved passband magnitude response, 2) improved stopband magnitude response around the folding bands of CIC filters, 3) no integrator section operating at high input sample rate, and 4) low-power consumption. Moreover, the paper provides several hints to reduce the computational complexity of the proposed filters by exploiting polyphase decomposition. Non-recursive architectures are discussed by focusing on specific decimation filters.

The rest of the paper is organized as follows. We first introduce the z-transfer function of the proposed class of filters in Section 2, where we also highlight its key features. In this section we also discuss the design of such filters by addressing the choice of key design parameters. Section 3 presents several examples aimed at contrasting the magnitude response of the proposed filters with classical comb filters. Several effective architectures for implementing the proposed decimation filters are discussed in Section 4, while comparisons with state-of-the-art techniques are presented in Section 5. Finally, Section 6 draws the conclusion.

2. The proposed decimation filters

The objective of this section is to introduce the notation used throughout the paper along with the proposed class of decimation filters and the criteria for the choice of the key filter parameters.

2.1. Notation

Let us briefly describe the notation used throughout the work by referring to the block diagram in Fig. 1. A baseband real input signal $x(t)$ with analog bandwidth $[-B_x, +B_x]$ is oversampled by an A/D converter and converted into the digital signal $x(nT_o)$ with sample rate f_o. The variable n identifies the integer discrete time. The interval T_o is related to the sampling frequency f_o and the signal bandwidth B_x through $f_o = \frac{1}{T_o} = 2\rho B_x$, where $\rho \geqslant 1$ is the oversampling ratio. Notice that $\rho > 1$ for oversampled signals, while $\rho = 1$ for A/D converters operating at the Nyquist frequency. The maximum digital frequency in the input signal is $f_c = \frac{B_x}{f_o} = \frac{1}{2\rho}$, meaning that the analog frequency B_x is mapped to f_c upon sampling. With this setup, the sampled signal $x(nT_o)$ at the input of the first decimation filter $H(z)$ shows frequency components in the range $[-f_c, f_c]$ as pictorially depicted in Fig. 1. The oversampling factor ρ is usually distributed between two, or more, decimation stages; therefore, it is $\rho = M \cdot \nu$ where M and ν are, respectively, the decimation factors of the first and second decimation stages.

Upon considering the sample frequency response in Fig. 1, it is worth noticing that, unlike classical filter design, the design of decimation filters imposes stringent constraints in the folding bands defined as

$$\left[\frac{p}{M} - f_c; \frac{p}{M} + f_c\right], \quad p \in \left\{1, \dots, \left\lfloor \frac{M}{2} \right\rfloor\right\}. \tag{1}$$

On the other hand, the remaining frequency intervals, called *don't care* bands, do not require stringent selectivity since any spurious signal falling within these bands will be filtered out by the subsequent stages in the multistage decimation architecture.

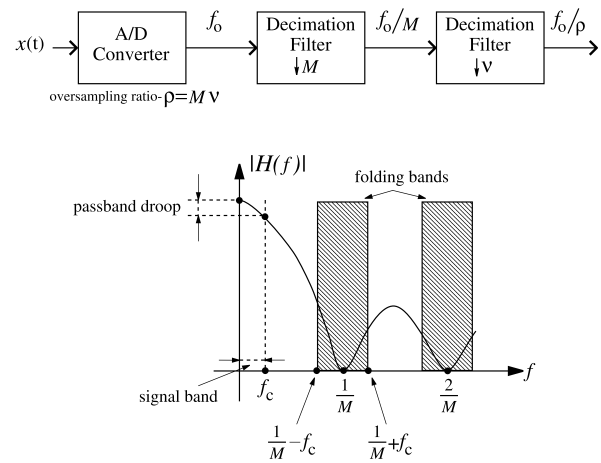

Fig. 1. Conceptual block diagram of a 2-stage decimation architecture along with a pictorial representation of the frequency response of the first decimation filter $H(z)$ and the key frequency intervals to carefully consider in the design.

2.2. The proposed decimation filters

The main goal of this paper is to introduce a new class of decimation filters for improving the magnitude response of classical comb filters while retaining their most important features, namely simple structure, low-power and multiplierless implementation. In order to meet this goal, we propose the following decimation filters with z-transfer function

$$H(z) = H_C(z)H_s(z)G\left(z^M\right). \tag{2}$$

In the previous equation, $G(z^M)$ is the z-transfer function of a passband droop-compensator filter, $H_s(z)$ is the z-transfer function of a filter used to increase folding band attenuation, and $H_C(z)$, the z-transfer function of a Kth-order comb filter decimating by M, is defined as

$$H_C(z) = \left(\frac{1}{M}\frac{1-z^{-M}}{1-z^{-1}}\right)^K. \tag{3}$$

The definitions of the basic building blocks $G(z^M)$ and $H_s(z)$ in (2) are derived in the next sections.

2.3. Droop-compensation filter

The main goal of filter $G(z^M)$ is to achieve passband droop compensation using an as simple as possible multiplierless filter with one free parameter independent from the decimation factor M, while working at the lower rate after decimation by M.

An effective, yet simple, compensation filter has magnitude response defined by [28]

$$\left|G\left(e^{j\omega M}\right)\right| = \left|1 + 2^{-b}\sin^2(\omega M/2)\right|, \tag{4}$$

where b is a suitable integer belonging to the set $\{-2, \dots, 2\}$, and ω is related to the digital frequency f through the relation $\omega = 2\pi f$. Upon using the well-known trigonometric relation $\sin^2(\alpha) = [1 - \cos(2\alpha)]/2$, the z-transfer function in (4) can be rewritten as

$$G\left(z^M\right) = B\left[1 + A \cdot z^{-M} + z^{-2M}\right]. \tag{5}$$

The constant B is a scaling factor ensuring unitary gain at the digital frequency zero. It is defined in power-of-2 form as $B = -2^{-(b+2)}$, whereas $A = -[2^{b+2} + 2]$. The choice of the parameter b is discussed in the following.