

VERSATILE BUILDING BLOCKS FOR MULTIRATE PROCESSING OF BANDPASS SIGNALS

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ABSTRACT

In this paper we develop a special down-converting bandpass decimator structure which can be used for reducing the sampling rate by a factor two without causing any disturbing aliasing effects for a bandpass signal located originally around quarter of the sampling rate ($\pi/2$). Based on this special bandpass filter design and usual halfband lowpass and highpass decimation filters, an efficient stepwise decimation and mixing approach is developed for filtering out and down-converting a narrowband signal located in frequency domain anywhere within a wideband digitized signal. This kind of down-conversion and decimation approach finds applications in communications receivers where the role of DSP-based solutions is becoming more and more important.

1. INTRODUCTION

There are several important applications for multirate processing of bandpass signals. Especially, in communications signal processing, more and more of the receiver and transmitter front-end processing is carried out by digital signal processing techniques. In receivers, one of the central tasks is to filter out a (relatively) narrowband bandpass signal and to translate it down, e.g., to a complex baseband signal (I/Q-signal) [1]. Due to high sampling rates utilized in advanced receiver architectures, it is very important to find efficient techniques for this task.

One straightforward approach is to mix the signal with a complex exponential at the center frequency (carrier) of the desired signal in order to translate it into a baseband I/Q-signal. Then efficient narrowband lowpass decimation techniques can be used for the channel selection filtering [1,2]. However, in this approach the complex multiplication for down-conversion needs two general multiplications to be carried out at the input sampling rate. In addition, a numerically controlled oscillator (NCO) is needed for generating the digital local oscillator signal at the input sampling rate. Furthermore, two real decimation filters are needed for processing the baseband I/Q-signal.

Here we are considering alternative approaches, where the down-conversion and selectivity filtering are carried out stepwise, using decimation and very simple mixing operations. We consider here the case of decimation by two, but the basic principle can be generalized to other decimation factors as well. Furthermore, the same ideas

can be used in the transmitter side for up-converting a signal and increasing the sampling rate.

In Section 2 we describe briefly half-band decimation filters and point out their limitations in the applications we are considering. Section 3 develops a special down-converting bandpass decimator structure. This, together with ordinary half-band decimators, makes it possible to handle in an efficient way any narrowband bandpass signal within a wideband digitized signal. A stepwise decimation and mixing approach to do this is described in Section 4.

2. HALF-BAND FILTERS IN DOWNCONVERSION

Efficient FIR and IIR half-band filters have been developed for decimation by two [3, 4]. The lowpass case is considered widely in the literature. In the example of Fig. 1, the lowpass decimator can handle all narrowband signals in the frequency band $[0, \omega_p]$. Likewise, the highpass decimator can handle all signals in the band $[\pi - \omega_p, \pi]$. In this case, the complementary halfband highpass filter is used for clearing the low-frequency band $[0, \omega_p]$ where the desired frequency band is aliased into in the sampling rate reduction. In this way the desired frequency band is translated towards baseband. Also in this highpass case the aliased components can be attenuated as much as desired.

However, using this approach for signals that are close to the frequency $\pi/2$ results in excessive computational complexity and other difficulties in implementing the halfband filter because the transition band would become

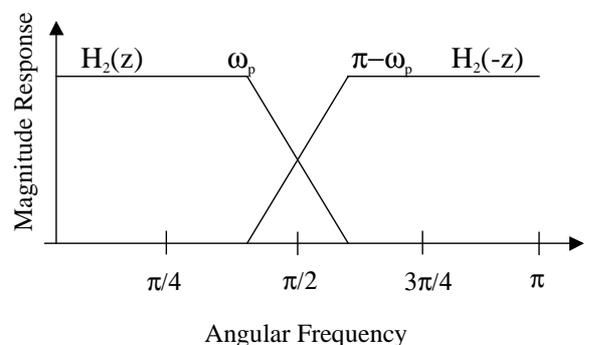


Fig. 1. Ideal frequency responses for halfband lowpass and highpass filters.

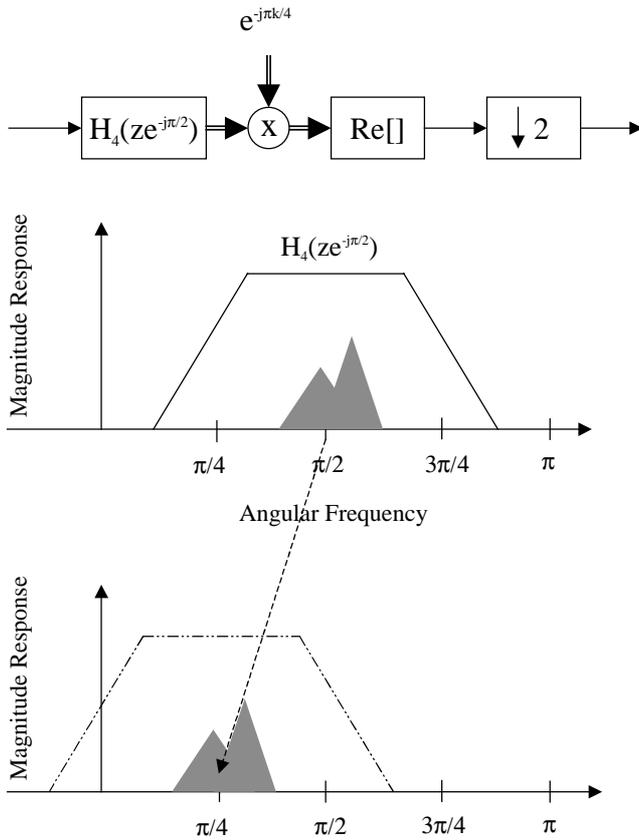


Fig. 2. The proposed concept of decimation by two for bandpass signals in the mid-frequency range.

very narrow. This is due to the fact that, in decimation by two, aliasing takes place from frequency $\pi/2 + \epsilon$ to frequency $\pi/2 - \epsilon$. This means also that if the interesting frequency range extends to both sides of $\pi/2$, sampling rate reduction by 2 can not be used for down-converting a real signal.

3. SPECIAL DOWN-CONVERTING BANDPASS FILTER DESIGN

We have examined the principle illustrated in Fig. 2 for handling bandpass signals in the mid-frequency range. The idea is to use a special type of complex bandpass filter for frequency selection and then translate the bandpass-filtered signal down, as a complex signal, by a fixed amount of $-\pi/4$ before decimation by 2. In this approach, if the transition bands of the bandpass filter are symmetric around $\pi/4$ and $3\pi/4$, aliasing will take place across the transition bands but there will be no (significant) aliasing to the passband region. Also the spectral mirror image produced by taking the real part has stopband in the passband region of the down-converted signal. Consequently, the spurious spectral components in the passband region due to both decimation by two and taking the real part can be made arbitrarily small by designing the bandpass filter to have sufficient stopband attenuation.

A suitable bandpass filter for this application can be obtained from a fourth-band FIR or IIR filter [3, 4] by translating it to the center frequency of $\pi/2$. The translation is carried out by replacing each delay element z^{-1} by the element jz^{-1} . Fig. 3(a) shows the initial filter structure based on polyphase representation of the fourth-band filter. The scaling factor of $1/4$ has been omitted for simplicity. The polyphase branch filters are either FIR filters or allpass IIR filters.

This structure of Fig. 3(a) can be developed into an equivalent but much more efficient form shown in Fig. 4. The optimized structure can be obtained by utilizing some basic equivalencies of multirate signal processing [3, 5]. This is illustrated in Fig. 3, which shows a sequence of equivalent forms. In Fig. 3(a), a unit delay has been included at the input to facilitate later developments.

In Fig. 3(b), the decimator block has been moved over the real-part block and the mixer, and the mixing sequence has been modified accordingly. In Fig. 3(c), the polyphase branches have been reordered and combined to the real and imaginary parts. Also the decimation and mixing blocks have been moved to the real and imaginary branches.

In Fig. 3(d), the multiplier j has been combined with the mixing sequence of the imaginary branch. Furthermore, the imaginary values in the mixing sequences have been replaced by zeros, since the imaginary parts would be removed anyway by the real-part block. Also the decimators have been moved to the inputs of the polyphase branches and the delays in the polyphase branches have been changed accordingly (i.e., each pair of unit delays has been replaced by a single delay).

The upper branch of Fig. 3(e) is obtained by noting that the cascade of decimation-by-two and interpolation-by-two blocks corresponds to mixing with the sequence $\{1, 0, 1, 0, \dots\}$ and that this mixer in cascade with a mixer with sequence $\{1, 1, -1, -1, 1, 1, \dots\}$ is equivalent to a single mixer with sequence $\{1, 0, -1, 0, 1, \dots\}$. The lower branch is obtained in a similar way, observing that the one-sample offset in the mixing sequence can be taken into account by moving the mixer in front of the unit delay.

In Fig. 3(f), the two decimators on the output side have been moved to the input of the polyphase branches and the delays have been changed accordingly. Finally, the structure of Fig. 4 is obtained from the structure of Fig. 3(f) by reordering the polyphase branches, replacing the delays and the decimators at the input side with a distributor, and by replacing the two interpolators and the unit delay at output with a combiner [3, 5].

From the structure Fig. 4, it can be observed that all filter operations are done at quarter of the input sampling rate. No complex (I/Q) signal processing is needed. The structure is simple and quite efficient to implement, and also the hardware speed limitations are relaxed.

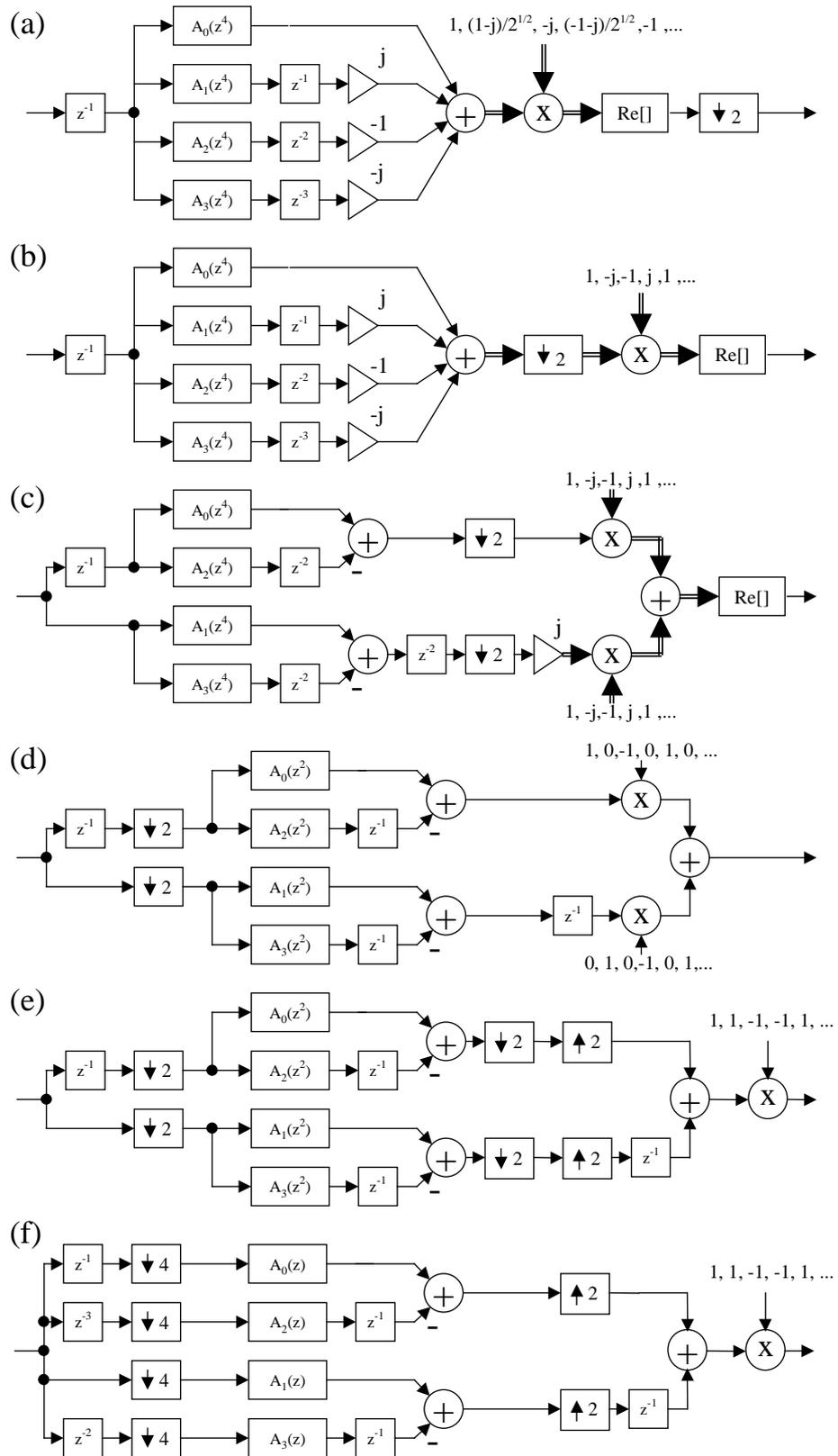


Fig. 3. Transforming the down-converting bandpass filter to computationally efficient form.

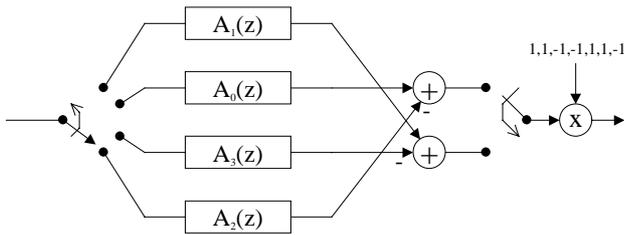


Fig. 4. The final, computationally efficient realization for the down-converting bandpass filter.

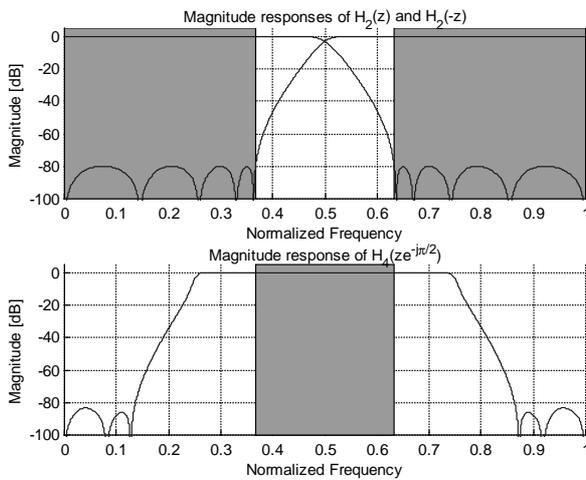


Fig. 5. Frequency responses of the example design.

Design Example

We consider the proposed type of decimation-by-two filter stages utilizing IIR Nth-band filters [4]. In this case the polyphase branches are composed of first-order allpass sections in cascade. Then optimizing the design in such a way that all the polyphase branches (both for the halfband and for the fourth-band filters) have two first-order allpass sections, the resulting attenuation of the aliased components is about 78 dB. The computational requirement is then to implement twelve simple fixed-coefficient first-order allpass sections at quarter of the input sampling rate.

4. STEPWISE DECIMATION AND MIXING APPROACH

The general approach for decimation of arbitrary narrowband bandpass signals is then as follows:

- Use a cascade of decimation-by-two -stages.
- In each stage, depending on the frequency where the desired signal is located, use a lowpass halfband decimator, highpass halfband decimator, or the special down-converting bandpass filter described above.

The edge frequencies of the halfband decimators and the bandpass decimator should be chosen to minimize the overall complexity of each stage. One approach is to make the design in such a way that all the polyphase branches have the same filter orders, as it was done in the previous example.

To accommodate signals close to the edge frequencies, the passbands of the halfband filter and the bandpass filter should be slightly overlapping. Continuing the previous example, we redesign the system using three allpass sections in all the polyphase branches while all the filters have the same 78 dB stopband attenuation. Then the bandwidths can be extended in such a way that any bandpass signal whose bandwidth is no more than 10 percent of the input sampling rate is completely within the passband of one of the filters (lowpass, bandpass, or highpass). It is obvious that when the relative bandwidth of the desired signal gets wider, higher filter orders are needed.

5. CONCLUSIONS

We have proposed a multistage bandpass decimator structure that can be used for down-converting any narrowband bandpass signal towards baseband. This approach could be used, e.g., as the first part of the receiver signal processing after wideband IF sampling. At a later stage, a fine-tuning down-conversion by mixing would be carried out to obtain the final baseband signal.

The proposed approach can be generalized in a straightforward way to the dual interpolation case. Also possibilities to apply the same approach with other sampling rate conversion factors than two are under study.

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