Simple Methods for Design of Narrowband High-Pass FIR Filters

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INTRODUCTION

Digital signal processing (DSP) is an area of engineering that "has seen explosive growth during the past three decades" (Mitra, 2001). Its rapid development is a result of significant advances in digital computer technology and integrated circuit fabrication (Smith, 2002; Jovanovic-Dolecek, 2002). Diniz, Silva, and Netto (2002) state that "the main advantages of digital systems relative to analog systems are high reliability, suitability for modifying the system's characteristics, and low cost."

The main DSP operation is digital signal filtering, that is, the change of the characteristics of an input digital signal into an output digital signal with more desirable properties. The systems that perform this task are called digital filters. The applications of digital filters include the removal of the noise or interference, passing of certain frequency components and rejection of others, shaping of the signal spectrum, and so forth (White, 2000; Ifeachor & Jervis, 2001).

Digital filters are divided into finite impulse response (FIR) and infinite impulse response (IIR) filters. FIR digital filters are often preferred over IIR filters because of their attractive properties, such as linear phase, stability, and the absence of the limit cycle (Mitra, 2001; Diniz, Silva & Netto, 2002). The main disadvantage of FIR filters is that they involve a higher degree of computational complexity compared to IIR filters with equivalent magnitude response (Mitra, 2001; Stein, 2000). In past years, many design methods have been proposed to reduce the complexity of FIR filters (Jou, Hsieh & Kuo, 1997; Kumar & Kumar, 1999; Webb & Munson, 1996; Lian & Lim, 1998; Bartolo & Clymer, 1996; Kuo, Chien & Lin, 2000; Abeysekera & Padhi, 2000; Yli-Kaakinen & Saramaki, 2001; Coleman, 2002; Jovanovic-Dolecek, 2003).

We consider high-pass (HP) linear-phase narrowband filters. It is well known that one of the most difficult problems in digital filtering is the implementation of narrowband filters. The difficulty lies in the fact that such filters require a high-order design with a large amount of computation, making them difficult to implement (Mitra, 2001; Grover & Deller, 1999; Stein, 2000).

We propose an efficient design of high-pass linear-phase narrowband digital filters based on the corresponding low-pass (LP) filter. In this design we use the interpolated FIR (IFIR) structure and the sharpening recursive running sum (RRS) filter (Jovanovic-Dolecek, 2003). In the next section we describe the transformation of a low-pass into a high-pass filter, followed by descriptions of an IFIR structure, RRS filter, and the sharpening technique. Finally, we present the design procedure, along with an example.

BACKGROUND

Transformation of LP into HP Filter

Instead of designing a high-pass filter by brute force, we can transform it into a low-pass filter. We replace the desired cutoff frequencies of the high-pass filter ω_p and ω_s, by the corresponding low-pass specifications as follows:

$$\begin{aligned} \omega_p' &= \pi - \omega_p \\ \omega_s' &= \pi - \omega_s \end{aligned} . \tag{1}$$

Given these specifications, a low-pass FIR filter can be designed. From this auxiliary low-pass filter, the desired high-pass filter can be computed by simply changing the sign of every other impulse response coefficient. This is compactly described as:

$$h_{HP}(n) = (-1)^n h_{LP}(n), \tag{2}$$

where $h_{HP}(n)$ and $h_{LP}(n)$ are the impulse responses of the high-pass and the low-pass filters, respectively. In that way the IFIR structure proposed for design of LP filters can also be used for HP filters.

IFIR Structure

The interpolated finite impulse response structure, proposed by Neuevo, Dong, and Mitra (1984), is an efficient realization of a high-order liner-phase LP FIR filter. Instead of designing one high-order liner-phase filter $H(z)$, two lower order liner-phase LP filters are computed. One of them is called the shaping or model filter $G(z)$, and the other one is the interpolator filter $I(z)$.

Suppose that the specifications of the original filter $H(z)$ are: pass-band edge ω_p,stop-band edge ω_s, pass-band ripple R_p, and minimum stop-band attenuation A_S. The specification of the LP filter $G(z)$ can then be expressed as follows:

$$\begin{aligned} \omega_p^G &= M\omega_p \\ \omega_s^G &= M\omega_s \\ R_p^G &= R_p / 2\,, \\ A_s^G &= A_s \end{aligned} \tag{3}$$

where M is the interpolation factor, and the upper index G stands for the filter $G(z)$. The expanded filter $G(z^M)$ is obtained by replacing each delay z^{-1} in the filter $G(z)$ with z^{-M} . In the time domain this is equivalent to inserting M-1 zeros between two consecutive samples of the impulse response of $G(z)$. The expansion of the filter $G(z)$ introduces M-1 images in the range [0, 2π], which have to be eliminated. This is why the filter interpolator $I(z)$ is needed. The general low-pass IFIR structure is given in Figure 1.

The high-pass filter design depends on the parity of the interpolation factor. For M even, there is an image at high frequency. If all other images along with the original spectrum are eliminated, the high-pass filter results. Therefore, the interpolator filter is a high-pass filter with the following specifications (Jovanovic-Dolecek, 2003):

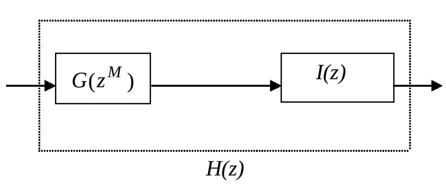
$$\begin{aligned} \omega_p^I &= \pi - \omega_p^{'} \\ \omega_s^I &= \frac{2\pi}{M}\frac{M-2}{2} + \omega_s^{'} = \frac{\pi(M-2)}{M} + \omega_s^{'} \\ R_p^I &= R_p / 2 \qquad , \\ A_s^I &= A_s \end{aligned} \tag{4}$$

where upper index I stands for the interpolator $I(z)$. Now, the resulting IFIR filter is the desired high-pass filter.

For M odd, however, there are no images at high frequency. In order to obtain the low-pass filter, all images have to be eliminated using the interpolator filter, so that only the original spectrum remains. In this case the interpolator $I(z)$ has the following specifications (Jovanovic-Dolecek, 2003):

$$\begin{aligned} \omega_p^I &= \pi - \omega_p^{'} \\ \omega_s^I &= \frac{2\pi}{M} - \omega_s^{'} \\ R_p^I &= R_p / 2 \qquad . \\ A_s^I &= A_s . \end{aligned} \tag{5}$$

Figure 1. IFIR structure



Since the resulting IFIR filter is a low-pass filter, we apply transformation (2) to achieve the desired high-pass filter.

To further simplify the proposed design, a simple RRS filter, described in the next section, is used as the interpolator in the IFIR structure.

Recursive Running-Sum Filter

The simplest low-pass FIR filter is the moving-average (MA) filter. Its impulse response $g(n)$ is given by:

$$g(n) = \frac{1}{M}\sum_{k=0}^{M-1} g(n-k)\,. \tag{6}$$

All impulse response coefficients are equal to 1, and so the filter requires no multiplications.

Its transfer function is given by:

$$G(z) = \frac{1}{M}[1 + z^{-1} + ... + z^{-(M-1)}] = \frac{1}{M}\sum_{k=0}^{M-1} z^{-k} \quad . \tag{7}$$

A more convenient form of the above transfer function for realization purposes is given by:

$$H_{RRS}(z) = [G(z)]^K \quad = \left[\frac{1}{M}\frac{1-z^{-M}}{1-z^{-1}}\right]^K, \tag{8}$$

which is also known as a recursive running-sum filter (RRS) (Mitra, 2001), or a boxcar filter. The scaling factor $1/M$ is needed to provide a dc gain of 0 dB, and K is the number of the cascaded sections of the filter. The magnitude response of the filter can be expressed as:

$$\left|H_{RRS}(e^{j\omega})\right| = \left| \quad \left[\frac{\sin(\omega M/2)}{M\sin(\omega/2)}\right]^K \right|. \tag{9}$$