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One Method for Design of Narrowband Low-Pass Filters

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INTRODUCTION

Stearns and David (1996) states that "for many diverse applications, information is now most conveniently recorded, transmitted, and stored in digital form, and as a result, digital signal processing (DSP) has become an exceptionally important modern tool." Typical operation in DSP is digital filtering. Frequency selective digital filter is used to pass desired frequency components in a signal without distortion and to attenuate other frequency components (Smith, 2002; White, 2000). The pass-band is defined as the frequency range allowed to pass through the filter. The frequency band that lies within the filter stop-band is blocked by the filter and therefore eliminated from the output signal. The range of frequencies between the pass-band and the stop-band is called the transition band and for this region no filter specification is given.

Digital filters can be characterized either in terms of the frequency response or the impulse response (Diniz, da Silva, & Netto, 2000). Depending on its frequency characteristic, a digital filter is either low-pass, high-pass, band-pass or band-stop filters. A low-pass (LP) filter passes low frequency components to the output while eliminating high-frequency components. Conversely, the high-pass (HP) filter passes all high-frequency components and rejects all low-frequency components. The band-pass (BP) filter blocks both low- and high-frequency components while passing the intermediate range. The bands-stop (BS) filter eliminates the intermediate band of frequencies while passing both low- and high-frequency components.

In terms of their impulse responses, digital filters are either infinite impulse response (IIR) or finite impulse response (FIR) digital filters. Each of four types of filters (LP, HP, BP, and BS) can be designed as an FIR or an IIR filter (Grover & Deller, 1999; Ifeachor & Jervis, 2001; Oppenheim & Schafer, 1999).

The design of a digital filter is carried out in three steps (Ingle & Proakis, 1997):

- Define filter specification;
- Approximate given specification;
- Implement digital filter in hardware or software.

The topic of filter design is concerned with finding a magnitude response (or, equivalently, a gain) which meets the given specifications. These specifications are usually expressed in terms of the desired pass-band and stop-band edge frequencies w_p and w_s, the permitted deviations in the pass-band (pass-band ripple) R_p, and the desired minimum stop-band attenuation A_s, (Mitra, 2001). Here, we consider the specifications given in dB. In this case, the gain in decibels is

$$\text{Gain in dB} = 20\log_{10}\left|H(e^{j\omega})\right|. \tag{1}$$

Figure 1 illustrates a typical magnitude specification of a digital low-pass filter.

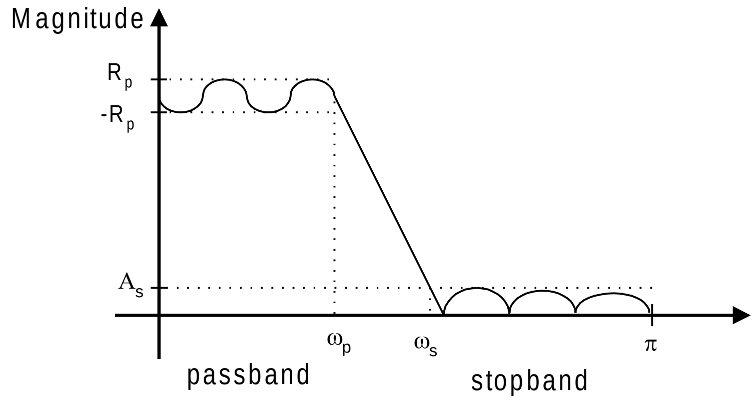
Due to their complexity, narrowband low-pass FIR filters are difficult and sometimes impossible to implement using conventional structures (Milic & Lutovac, 2002). The interpolated finite impulse response (IFIR) filter proposed by Neuvo, Cheng-Yu, and Mitra (1984) is one efficient realization for the design of narrowband FIR filters.

The IFIR filter $H(z)$ is a cascade of two filters,

$$H(z) = G(z^M)I(z), \tag{2}$$

where $G(z^M)$ is an expanded shaping or model filter, $I(z)$ is an interpolator or image suppressor, and M is the interpolator factor. In this manner, the narrowband FIR prototype filter $H(z)$ is designed using lower order filters, $G(z)$ and $I(z)$. For more details on the IFIR structure, see Neuvo, Cheng-Yu, and Mitra (1984) and Jovanovic-Dolecek (2003).

Figure 1. Low-pass filter magnitude specification



An increase in the interpolation factor results in the increase of the interpolation filter order as well as in the decrease of the shaping filter order.

The design goal in Jovanovic-Dolecek and Diaz-Carmona (2003) is to decrease the shaping filter order as much as possible, and to efficiently implement the high order interpolator filter. To do so, we propose to use a sharpening recursive running sum (RRS) as an interpolator in the IFIR structure.

BACKGROUND

Sharpening Technique

The sharpening technique, which was first proposed by Kaiser and Hamming (1984), attempts to improve both the pass-band and stop-band of a linear FIR filter by using multiple copies of the same filter based on the Amplitude Change Function (ACF). An ACF is a polynomial relationship of the form $H_0(w) = P[H(w)]$ between the amplitude responses of the overall and the prototype filters, $H_0(w)$ and $H(w)$, respectively. The improvement in the pass-band, near H=1, or in the stop-band, near H=0, depends on the order of the tangency of the ACF m and n at H=1 or H=0, respectively. The expression proposed by Kaiser and Hamming for an ACF giving the mth and nth order tangencies at H = 1 and H = 0, respectively, is given by

$$H_0 = H^{n+1} \sum_{k=0}^{m} \frac{(n+k)!}{n!\,k!}(1-H)^k$$
$$= H^{n+1} \sum_{k=0}^{m} C(n+k,k)(1-H)^k , \qquad (3)$$

where $C(n+k,k)$ is the binomial coefficient.

Hartnett and Boudreaux (1995) proposed an extension of this method by introducing the slope d of tangency at H=0 and the slope s of tangency at H=1. Samadi (2000) proposed an explicit formula for the filter sharpening polynomial.

We use the following ACF polynomials:

1) $$P\left[H(\omega)\right] = 3H^2(\omega) - 2H^3(\omega); \delta = \sigma = 0; m = n = 1.$$

2) $$P\left[H(\omega)\right] = 6H^2(\omega) - 8H^3(\omega) + 3H^4(\omega); \delta = \sigma = 0; m = 2; n = 1.$$

3) $$P\left[H(\omega)\right] = 10H^2(\omega) - 20H^3(\omega) + 15H^4(\omega) - 4H^5(\omega); \delta = \sigma = 0; m = 3, n = 1.$$

(4)

The plot of each one of these ACF is shown in Figure 2. Note that the third ACF has the best passband improvement and the smallest stopband attenuation, whereas, the first ACF has the worst pass-band improvement but the best stopband attenuation.

The three sharpening structures are presented in Figure 3, where the resulting number of multipliers per output sample (MPS) is equal to three, four and five for the first, second and third structure, respectively.

SHARPENING RRS FILTER

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

$$g(n) = \frac{1}{M}\sum_{k=0}^{M-1} g(n-k). \qquad (5)$$