Chapter 57

Comb Filters Characteristics and Current Applications

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ABSTRACT

The comb filter is a very popular linear-phase filter due its simplicity (i.e., all its coefficients are equal to unity). As a consequence, it does not require multipliers or coefficients storage. This characteristic makes this filter attractive for many applications, as for example, in decimation, communications, digital audio, among others. However, the comb filter presents passband droop and a poor attenuation in the stopband region. In this proposal, the comb filter characteristics are reviewed and illustrated with one example. Additionally, the selected methods commonly used to improve the magnitude characteristics of a comb filter will be described and illustrated with examples.

INTRODUCTION

A comb filter is a Linear Time-Invariant (LTI) digital filter, where linear means that its output to a scaled sum of input digital signals is equal to the scaled sum of the outputs to every one of these input signals (i.e., the filter satisfies the superposition principle) and time-invariant means that, for any input signal that has a given delay, the output undergoes the same delay as the input (Antoniou, 2006). The name comb is derived by the fact that its magnitude response resembles the teeth of a comb. Since there are several filters having magnitude responses with such characteristic, the term comb filter is rather general. The duration of the impulse response of comb filters can be either finite or infinite, i.e., there are Finite Impulse Response (FIR) comb filters and Infinite Impulse Response (IIR) comb filters (Zölzer, 2008).

The simplest FIR comb filter has the following transfer function:

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\[ H_a(z) = 1 + z^{-M} \]  

(1)

This filter adds to a signal a version of that signal delayed by \( M \) sample periods, and it is the basic building block to introduce echo effects in audio signals (Zölzer, 2008). Moreover, if the addition in (1) is replaced by a subtraction, the resulting comb filter is a useful building block to remove DC and harmonics (Diniz, Da Silva & Neto, 2010). The unintentional delay of an audio signal due to the environment is also modeled as a comb filter (Toole, Shaw, Daigle & Stinson, 2001), and this effect may be undesirable in many cases. Similarly, a simple IIR comb filter has the following transfer function:

\[ H_b(z) = \frac{1}{1 - az^{-M}} \]  

(2)

with \( a < 1 \). This filter is a basic building block to model and create reverberation effects or, in general, to artificially reproduce the acoustics of a room (Zölzer, 2008).

One of the most important comb filters for several Digital Signal Processing (DSP) applications is the one based in the FIR filter where all the samples of its impulse response have values equal to one. Unlike the aforementioned comb filters described by \( H_a(z) \) in (1) and \( H_b(z) \) in (2), this comb filter has a low-pass characteristic, which makes it useful to pass a baseband signal and remove unwanted high-frequency spectra (Milic, 2009). The rest of this chapter is dedicated to this particular filter, which will be referred as the comb filter hereafter. The main characteristics of the comb filter, as well as its advantages and disadvantages will be highlighted. Moreover, we will present the selected methods commonly used to decrease the disadvantages of the comb filters with minimum affectation of its advantages.

In the efficient implementation of the comb filter, a comb filter with transfer function based in (1) (just with the addition replaced by a subtraction) is employed. In order to avoid confusion, that filter will be referred hereafter as comb differentiator, since it is based on a simple first-order differentiator.

**Background**

Consider a simple FIR filter that has the following transfer function (Milic, 2009):

\[ H(z) = \frac{1}{M} \sum_{k=0}^{M} z^{-k} \]  

(3)

where \( M \) is the filter order. The coefficients that multiply the variable \( z \) are all equal to 1. Thus, the non-recursive implementation of this filter does not require multipliers for its coefficient’s values. The impulse response of this filter is \( h(k) = 1 \) for \( 0 \leq k \leq M \) and 0 for other values of \( k \). The scaling factor \((1/M)\) is included to provide a normalized gain of 0 dB at frequency equal to zero.

The transfer function of the comb filter arises from expressing the transfer function given in (3) in recursive form as follows (Lyons, 2004):

\[ H(z) = \frac{1 - z^{-M}}{M(1 - z^{-1})} \]  

(4)
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