Biomedical Signal Processing and Control 31 (2017) 211–216

Contents lists available at ScienceDirect

Biomedical Signal Processing and Control

journal homepage: www.elsevier.com/locate/bspc

Variable delay digital comb filter extraction of weak phase signals for SSVEP

Chih-Tsung Chang*, Po-Lei Lee, En-Hung Lin

Department of Electrical Engineering, National Central University, No. 300, Jhongda Rd., Jhongli City, Taoyuan County, 32001, Taiwan, ROC

ARTICLE INFO

Article history:
Received 1 August 2015
Received in revised form 14 August 2016
Accepted 17 August 2016

Keywords:
Steady-state visual evoked potential
Electroencephalogram
Quadrature amplitude modulation
Comb filter
Variable delay

ABSTRACT

This study presents denoising of phases signal by comb filter. Signal phase is widely used in biomedical and communications engineering, and has become increasingly important over the past few years. In particular, steady-state visual evoked potential (SSVEP) of electroencephalogram (EEG) and quadrature amplitude modulation (QAM) uses phase to distinguish channel and data. For signal processing, the filter is usually adopted. In this study, the usefulness of comb and bandpass filters in detecting the phase of a signal is compared by using simulated and real signals to verify phase signal denoising. Experimental data show that comb filters had a higher signal to noise ratio (SNR) of phase signal. However, even if a comb filter performs well, the delay time of a digital comb filter will be limited by the sampling rate. It is possible to solve this problem, but doing so is inconvenient within the context of real-time applications. Using a variable delay (VD) approached to fractional delay for digital comb filter, it is both simple and easy to use.

1. Introduction

In biomedical engineering research, teams use electroencephalograms (EEGs) to implement brain computer interfaces (BCIs). These interfaces represent direct communication pathways between the brain and an external device that can translate physiological signals into commands. Such interfaces can help patients to communicate with the outside world. BCI systems have been designed to use brain functions such as sensorimotor rhythm [1], P300 event-related potentials [2], visual evoked potential [3], and steady-state visual evoked potential (SSVEP) [4,5]. Further, these systems use differing signal process and recognition methods. SSVEP, for example, recognizes a sinusoidal occipital EEG signal using Fast Fourier Transform (FFT). In 2010, the authors published a paper on the use of SSVEP in which frequency was combined with phase encoding [6]. Epoch averaging using differential times to classify the phase was used to process a signal. Although the SSVEP had a high signal to noise ratio (SNR) in comparison with other methods, the EEG signal was still very weak and susceptible to noise [7]. As phase encoding has become very popular over the past few years [8–13], phase signal processing has become increasingly important. The signal phase is also used to distinguish digital data in wireless communications systems using methods such as quadrature amplitude modulation (QAM) [14–16]. In this paper, the use of a comb filter is proposed to improve the SNR of the signal phase. Although epoch averaging improves the SNR of the signal phase, this method segments the signal, leading to a degree of discontinuity, which limits the signal processing ability. A comb filter output signal, in contrast, is continuous. Thus, the use of a comb filter will both improve the SNR of a signal phase and result in a continuous signal that can be classified in a wide number of ways. Finally, the authors compare the comb filter results with those of a general filter and demonstrate that general filters do not improve the SNR of the signal phase.

The literature [17] shows that various types of comb filters have been previously investigated. Comb filters determine frequency by including delay time; however, the use of discrete time will restrict delay time because of the sampling rate. In other words, in discrete time sampling, the delay time will be measured as an integer, and this in turn restricts the accuracy with which frequencies can be used in digital comb filter. In the literature [7], it has been proposed that the choice of the frequency range (from 5 to 45 Hz) used in SSVEP results in problems owing to high user variation. In contrast, the digital comb filter frequency is adjusted by the user. Typically, researchers using digital comb filters have adopted resampling techniques to control the sampling rate [18], resulting in an integer delay times. Fractional delay (FD) has been used to achieve non-integer delays [19,20]. Furthermore, Pei [21] designed

---

* Corresponding author.
E-mail address: cool6746@gmail.com (C.-T. Chang).

http://dx.doi.org/10.1016/j.bspc.2016.08.008

a digital comb filter that can use fractional delay times. However, these methods require changing the digital comb filter structure and involve complex designs with difficult-to-use techniques, such as sinc functions and re-sampling, which are generally too complex for use with BCIs. As they are used in real time, complexity is a primary design consideration for BCIs. In this study, the authors propose a variable delay (VD) digital comb filter that is both simple and easy to use and demonstrate that the VD can be used to easily adjust the frequency of a digital comb filter.

2. Materiaks and methods

2.1. Subjects and tasks

Eight volunteers (five males and three females), age ranges between 30+ 5, were recruited for the experiment. Each participant's eyesight was normal or normal after correction, and no subject had any neurological disease. All subjects were informed about the purpose of the study, and informed consent was obtained.

2.2. Control task

Subjects in the study were requested to gaze at the center of the eight flickering LEDs, labeled LEDI, I = 1, 2, 3. . . 8 one after another. The eight LEDs were being gazed one-after-one in the order from LED1-8 for 30 s and each LED collects EEG data for three times and results were being recorded.

2.3. EEG recordings

The VEP data has been collected under an environment without any noise. One electrode (Oz(+)) placed at the Oz position and the another electrode (Oz(−)) placed at the right mastoid, with respect to a ground electrode placed at frontal position (Fpz). (bandpass, 10–50 Hz; MacLab, BioAmp, ADInstruments, Castle Hill, Australia).All EEG electrode placement were based on the international EEG 10–20 system and EEG recordings were digitized at 1 kHz.

3. Feedback digital comb filter with variable delay (VD)

3.1. Proof

The proposed feedback comb filter is represented as

$$y(t) = bx(t) + ay(t-\tau) \tag{1}$$

where $x(t)$ is the analog input signal amplitude, $y(t)$ is the analog output signal amplitude, τ is the delay time, and a and b are the scaling factors. Output $y(t)$ can be re-expressed before converting to discrete time as follows:

$$\begin{aligned} y(t) &= bx(t) + a(bx(t-\tau) + ay(t-2\tau)) \\ &= \mathrm{b}(x(t) + ax(t-\tau)) + a^2 \mathrm{y}(t-2\tau) \\ &= \mathrm{b}(x(t) + ax(t-\tau)) + a^2(bx(t-2\tau) + ay(t-3\tau)) \qquad (2) \\ &\vdots \\ &= \lim_{m\to\infty} \mathrm{b}\sum_{i=0}^{m-1} a^i \mathrm{x}(t - i\cdot\tau) + a^m \mathrm{y}(t - m\cdot\tau) \\ &= \lim_{m\to\infty} \mathrm{b}\sum_{i=0}^{m-1} a^i \mathrm{x}(t - c_i) + a^m \mathrm{y}(t - c_m) \qquad (2) \end{aligned}$$

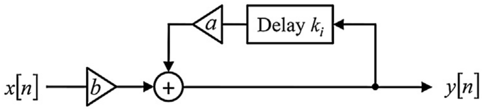

Fig. 1. Feedback comb filter with VD.

where c_i is the cumulative delay time, with $c_i = i.\tau$ and $c_i = c_{i-1} + \tau$. Rewriting (2) in discrete-time representation, we can obtain the following equation:

$$y[n] = \lim_{m\to\infty} b\sum_{i=0}^{m-1} a^i x[n - C_i] + a^m y[n - C_m] \tag{3}$$

The cumulative delay C_i is a rounded sampled value of c_i; that is, $C_i = round(c_i.f_s)$, where $round()$ is a rounding operator that replaces the input value with its nearest integer value, and f_s is the sampling rate. Eq. (3) can be rewritten in the form of the m^{th} delay k_m or the difference between the m^{th} and the $(m-1)^{th}$ cumulative delays (i.e., the discrete time delay $k_m = C_m - C_{m\text{-}1}$ or the analog time delay $\tau = c_i - c_{i\text{-}1}$) as

$$\begin{aligned} y[n] = \lim_{m\to\infty} b\sum_{i=0}^{m-2} a^i x[n - C_i] + a^{m-1} \\ (bx[n - C_{m-1}] \quad + ay[n - C_{m-1} - k_m]) \end{aligned} \tag{4}$$

shows that $y[n - C_{m\text{-}1}] = bx[n - C_{m\text{-}1}] + ay[n - C_{m\text{-}1} - k_m]$

$$\begin{aligned} &\mathrm{y}[n] = \lim_{m\to\infty} \mathrm{b}\sum_{i=0}^{m-2} a^i x[n - C_i] + a^{m-1} \mathrm{y}[n - C_{m-1}] \\ &= \lim_{m\to\infty} \mathrm{b}\sum_{i=0}^{m-3} a^i x[n - C_i] + a^{m-2}(bx[n - C_{m-2}] + ay[n - C_{m-2} - k_{m-1}]) \qquad (5) \\ &\vdots \\ &= \mathrm{bx}[n] + \mathrm{ay}[n - C_1] = \mathrm{bx}[n] + \mathrm{ay}[n - k_1] \end{aligned}$$

Eq. (5) represents a general digital comb filter. Output $y[n]$ can be obtained by using a series of k_i representing discrete delays time. Although the delay time series are all equal for analog signals, this will not necessarily be identical for the series of k_i in discrete time. Therefore, the delay of a digital comb filter can be varied; that is, since differing times n have differing delays k_i, a VD digital comb filter can be used to realize arbitrary frequencies.

3.2. Example

The sample rate used here was 1 kHz; the flickering frequency f was set to 24.9 Hz, resulting in a delay time τ of 1/24.9 or 0.0402 s; and a and b were set at 0.98 and 0.02, respectively. To validate the VD, the delay time was a non-integer in discrete time. A general feedback comb filter structure, shown in Fig. 1, was used. A variable delay was used to develop a comb filter using an arbitrary frequency in discrete time, calculated using $C_i = round(c_i.f_s) = round(i.\tau.f_s)$ and $k_i = C_i - C_{i\text{-}1}$.

Comb filter steps:

(1) If $n = n_1$, find the corresponding cumulative delay, i.e., $C_{i-1} \le n_1 < C_i$.
(2) Using $C_{i\text{-}1}$ and C_i, calculate delay k_i.
(3) $y[n_1] = bx[n_1] + ay[n_1 - k_i]$, when time is $n = n_1$, the delay is k_i.

If $n = 0$, then $i = 1$ and $k_1 = 40$. After $n = 0$, the delay is still $k_1 = 40$ until $n = 40$. When $n = 40$, $i = 2$ and $k_2 = 40$, and so on. These results are summarized in Table 1.