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Analog wavelet transform using multiple-loop feedback switched-current filters and simulated annealing algorithms

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ABSTRACT

A new approach for implementing continuous wavelet transform (CWT) based on multiple-loop feedback (MLF) switched-current (SI) filters and simulated annealing algorithms (SAA) is presented. First, the approximation function of wavelet bases is performed by employing SAA. This approach allows for the circuit implementation of any other wavelets. Then the wavelet filter whose impulse response is the wavelet approximation function is designed using MLF architectures, which is constructed with SI differentiators and multi-output cascade current source circuits. Finally, the CWT is implemented by controlling the clock frequency of wavelet filter banks. Simulation results of the proposed circuits and the filter banks show the advantages of such new designs.

1. Introduction

The continuous wavelet transform (CWT) is a widely used signal processing technique, particularly for local analysis of nonstationary and fast transient signals [1–3]. In wearable and implantable biomedical devices, such as wearable detector and pacemakers, power consumption is a critical issue due to the limited energy density and the lifetime of currently available portable batteries. This problem implies that the design of such devices has to be optimized for low-power dissipation. Traditionally, systems employing the CWT are implemented using digital signal processing devices. However, it is not suitable to implement the CWT due to the high power consumption and real-time associated with the required analog-to-digital (A/D) converters. From the power consumption and real-time perspectives, an excellent alternative is to use analog circuits and to implement the CWT instead.

In the past decades, there have been significant advances in the analog implementations of the CWT using continuous-time and sample-data circuits and their practical applications [4–21]. Especially, the analog sample-data circuits for the design of CWT system have caused much attention [14–21]. The main reason of using sample-data circuits for implementing CWT is that dilations of a given filter may be easily and very precisely controlled by both the component parameter ratios and the clock frequency. In [14,15], the switched-capacitors (SC) circuits are used to implement CWT in an analog way. SC circuits are suited for this application since the dilation constant across different scales of the transform can be implemented and controlled by both the capacitor ratios and the clock frequency. However, the reduced supply voltage of advanced CMOS technologies has imposed new challenges on the design of SC circuits especially for high-speed applications. Furthermore, SC is not fully compatible with current trends in digital CMOS process, because they require good linear floating capacitors. Consequently, the design approach for implementing the CWT by means of switched-current (SI) circuits is developed [16–21]. In SI methods of CWT realization, the circuits consist of analog SI filters whose impulse response is the approximated wavelet. So the performance of the implementation of the analog CWT depends largely on the accuracy of the approximation. Firstly, Padé approximation [7,16] is used to approximate the Laplace transform of the desired filter transfer function by a suitable rational function. The main advantage of the Padé method is its computational simplicity and its general applicability. However, there are also some disadvantages limiting its practical applicability. Among others, one important issue is stability. The stable transfer function of a wavelet filter does not automatically result from this approach. Another drawback is that the choice of the degrees for the numerator and denominator polynomials may yield an inconsistent system of equations. Subsequently, more accurate L_2 approach has been reported [4,6,18–20,22]. The advantage of the L_2 method over Padé approximation is that L_2 approximation offers a more accurate approximation. Also, the approximation can be performed directly

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in time domain. A major drawback of this approach is the possible existence of local optima. Moreover, the choice of a good starting point becomes very important and difficult. Recently, a differential evolution algorithm has been used to calculate the transfer function of the filter [17,21]. However, in this algorithm setting proper parameters is difficult and takes longer operation time. The other key issue of CWT circuit realization is the design of wavelet filter topology. In general, there are cascade [16,21] and parallel [17] architectures for implementing the wavelet filter based on SI integrators. However, these structures have high sensitivity [23], in particular as filter order increases. One of the methods for obtaining low sensitivity in the filter design is to use MLF networks. A MLF structure for implementing CWT with two-input multiple-output SI bilinear integrator as building blocks has been proposed in [20]. However, the feedforward and feedback coefficients in the circuit cannot be directly determined by the transfer function. Moreover, for the proposed circuit, it is difficult to adjust the coefficient values in order to design new filters.

In this paper, a new CWT circuit is presented using MLF SI filter and SAA method. In order to obtain the transfer function of the wavelet filters, the wavelet base is approximated in time domain employing SAA method, which overcomes the shortcomings of Padé, L_2 approach and other complex intelligent algorithms. The CWT circuit consists of a wavelet filter bank whose impulse response is the approximation wavelet and its dilations. The wavelet filter design is based on a MLF structure with SI differentiators and cascade current source. The coefficients of the filter can be directly obtained by the transfer function. In addition, by changing the output currents of the cascade current source, the new filter is easy to be designed.

This paper is structured as follows. In Section 2, it is argued how the CWT can be implemented with analog filters. In Section 3, the SAA method is used to compute the transfer function which describes a certain wavelet base that can be implemented as an analog filter. Section 4 describes the complete filter design issues. Some results obtained by simulations are given in Section 5. Finally, the conclusions are presented in Section 6.

2. Principle of wavelet transform design

The CWT is a linear operation that decomposes a signal into components that appear at different scales. The CWT of a continuous time signal $f(t)$ at scale a and position τ is defined as

$$W_\psi(\tau, a) = \frac{1}{\sqrt{a}} \int_{-\infty}^{\infty} f(t)\psi^* \left(\frac{t-\tau}{a}\right) dt \quad (1)$$

where $\psi^*(t)$ is the complex conjugation of the given admissible wavelet function, called the wavelet base $\psi(t)$. When the signal $f(t)$ is passed through a linear time-invariant filter, the filter output is the convolution of $f(t)$ with the impulse response $h(t)$ of the filter:

$$(f * h)(\tau) = \int_{-\infty}^{\infty} f(t)h(\tau - t)dt \quad (2)$$

From (1) and (2), it is well known that analog computation of the CWT $W_\psi(\tau, a)$ can be achieved through the implementation of a linear filter with impulse response

$$h(t) = \frac{1}{\sqrt{a}}\psi\left(\frac{-t}{a}\right) \quad (3)$$

For obvious physical reasons only the hardware implementation of causal stable filters is feasible. However, for a given wavelet base $\psi(t)$, the transfer function $H(s)$ will usually be non-rational and non-causal. Therefore, we consider approximations of $H(s)$ by transfer function that have all their poles in the complex left half plane and which are strictly proper rational. Note that $h(t)$ will be zero for negative t, so that the time-reversed wavelet base $\psi(-t)$ which does not have this property must be time-shifted to facilitate an accurate approximation of its CWT. In the next section, we will discuss the wavelet approximation that is suitable for analog implementation.

3. Wavelet function approximation in time domain

3.1. The model of wavelet function approximation

The first stage in analog filter design is the definition of the respective transfer function. However, a linear differential equation having a desired impulse response does not always exist. Thus, we must employ a suitable approximation method. From linear systems' theory, it is known that any strictly causal LTI filter of finite order n can be represented as a state-space system (A, B, C, D), corresponding to a system of associated first-order differential equations:

$$\dot{x}(t) = Ax(t) + Bu(t) \quad (4)$$

$$y(t) = Cx(t) + Du(t) \quad (5)$$

where $x(t)$ is the state vector, $u(t)$ is the input signal and $y(t)$ is the output of the filter. The matrix D is set to zero to achieve causality. The associated impulse response $h(t)$ and its Laplace transform $H(s)$ of the system are given by

$$h(t) = Ce^{At}B \quad (6)$$

$$H(s) = C(sI - A)^{-1}B \quad (7)$$

For the generic situation of stable systems with distinct poles, the impulse response function $h(t)$ is a linear combination of damped exponentials and exponentially damped harmonics. The impulse response function $h(t)$ of N order filter may typically have the following form [4,17,21]

$$h(t) = \sum_{i=1}^{m} a_i e^{b_i t} + \sum_{j}^{n} c_j e^{d_j t} \cos(\rho_j t) + f_j e^{g_j t} \sin(\rho_j t) \quad (8)$$

where the parameters a_i, b_i, c_j, d_j, ρ_j, f_j and g_j are real numbers. m and n correspond to the number of real poles, and $m+n=N$. For instance, a 5th order approximation function may be described as

$$h(t) = r_1 e^{r_2 t} + r_3 e^{r_4 t} \cos(r_5 t) + r_6 e^{r_4 t} \sin(r_5 t) + r_7 e^{r_8 t} \cos(r_9 t) + r_{10} e^{r_8 t} \sin(r_9 t) \quad (9)$$

where the parameters r_2, r_4 and r_8 must be strictly negative for reasons of stability.

As an example, the first derivative of Gaussian approximation is considered in the following. The first derivative of a Gaussian wavelet is expressed as

$$\psi(t) = -2te^{-t^2} \quad (10)$$

As stated before, if $h(t)$ is used to approximate a time-shifted and time-reversed wavelet function $\psi(t_0 - t)$, the output of the linear filter is the approximate wavelet transform $W_\psi(\tau, a)$. The selection of the time-shift t_0 is an important process. If t_0 is chosen too small, the truncation error will be too large and the overall approximation performance will decrease. If t_0 is chosen too large, the function to be approximated will become very flat near $t=0$. This effectively induces a large time-delay. Fig. 1 shows the waveforms of different time-shifting in time-reversed wavelet function. In [17,21], the selection problem of time-shift t_0 is not discussed and the value of time-shift is directly provided. For a selected time-shift t_0, the computation of an accurate approximation can be achieved using various approaches. Any approximation method should be associated with some measure of error. Therefore, we define an error