A NEW METHOD FOR WAVELETS GENERATION

A. Martínez-González, L. Ortiz-Balbuena, H. Pérez-Meana, E. Sánchez-Sinencio and J. C. Sánchez-García*
Universidad Autónoma Metropolitana Iztapalapa ,Depart.of Electrical Engineering, CBI Division
Av. Michoacán y Purísima. Col. Vicentina, Iztapalapa. C.P. 09340 México, D.F. México
Tel: (525) 725 46 35; Fax: (525) 725 49 02
* Texas A & M, Department of Electrical Engineering, College e Station, Texas, U.S.A.
e-mail: leob@xanum.uam.mx

ABSTRACT

Wavelets operators are very important in most practical applications. Implementation of these operators in software and in commercial DSP hardware are popular. We are presenting an alternative hardware implementation of wavelets operators using mixed-mode signal techniques, that is, a judicious combination of analog and digital hardware implementations. The approach is general and can be applied to a number of wavelets types.

1 INTRODUCTION

Wavelets operators are very important in many areas of signal processing. Although wavelets analysis is a young field, several applications has been developed for image processing, data compression, subband coding, multiresolution analysis, speech processing, etc. The most common way to implement wavelets operators is using the DSP techniques in which is relatively easy get a robust solutions to particular problem. However, for some applications like cellular telephony, the idea to implement channel equalizer, subband analog encryption using FFT algorithms and analog to digital/digital to analog conversion or make data compression for digital cellular transmission is very attractive but it has some limitations. To implement an efficient channel equalizer is required at least 15 taps, that means two DSP with RLS algorithm working at the same time.This situation could be impractical because cellular phones must be smaller as they can and the power consumption has to be minimum. In similar fashion, to implement a subband analog encryption required analog to digital conversion and Fourier transformation in order to mix the bands that involves again a DSP.

On the other hand, adaptive filter can be used to synthesize any arbitrary impulse response [1], and although almost all of them are implemented using DSP. However, it can be implemented in analog way too [2] [3]. So the main idea is, that an adaptive analog filter can reproduce wavelets operator with a previous training.

2 PROPOSED STRUCTURE

The proposed structure is showed in the block diagram of figure 1. First of all, the analog adaptive filter has to be trained before it can reproduce the wavelet transform of any input signal. The wavelet transfer function W(z) and the training signal X(n) are generated by a DSP for several reasons. The first one, is that currently there is no way to generated analog wavelets transform, second is that there is no mathematical expression for some wavelets like Malvar wavelets and the only way that can be generated is using an algorithm, the third one is that is relative easier generate wavelets using a DSP. Regarding to the adaptive filter, it is implemented with an IIR transversal structure in which the delay block Z-1 has been replaced with Laguerre functions using a bilineal transformation [4] [5], that is:

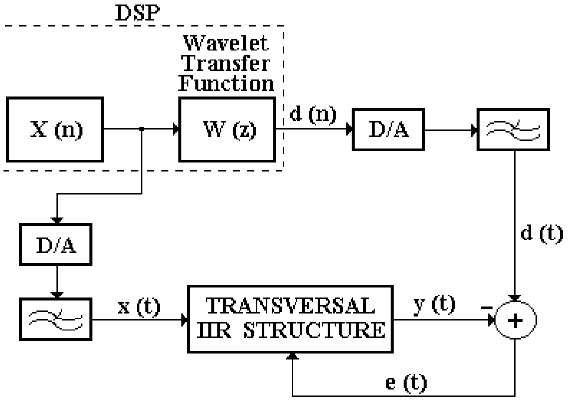

Figure 1. Proposed structure.

$$z = \frac{s+p}{s-p} \qquad (1)$$

$$L_n(s) = \frac{\sqrt{2p}\,(p-s)^n}{2\pi\,(s+p)^{n+1}} \qquad (2)$$

Equation (2) Laguerre functions, represents the transfer function of a cascade of all pass analog filters [4]. The figure 1 is working in a very well know identification configuration. In this way, when signal X(n) is fed in to wavelet transfer function, adaptive filter identify this transfer function W(z). Ones the process is completed, the coefficients of adaptive filter are kept and filter is ready to calculate wavelet transform of any input signal. The adaptive algorithm used for this structure is an analog LMS. The discrete LMS algorithm can be expressed by equation (3) and making the partial derivative it can be expressed as equation (4)

$$a_n(kT) = a_n((k-1)T) - \mu \frac{\partial e^2 (k-1)T}{\partial a_n (k-1)T} \tag{3}$$

$$a_n(T) = a_n(0) + 2\mu\, e(0)\, x_n(0) \tag{4}$$

$$a_n(2T) = a_n(T) + 2\mu\, e(T)\, x_n(T)$$

A generalization of equation (4) is given by (5). At this point if the sample period T become more and more short, in other words taking the limit when T approach zero, equation (5) change to expression given by (6) which is the analog LMS algorithm.

$$a_n(kT) = a_n(0) + 2\mu \sum_{i=0}^{k-1} e(kT)\, x_n(kT) \tag{5}$$

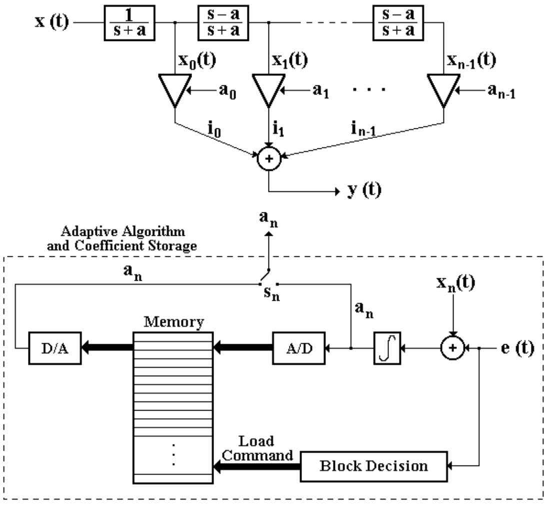

Figure 2. Complete proposed structure showing the analog adaptive filter and the storage structure.

$$a_n(t) = a_n(0) + 2\mu \int_0^t e(\tau)\, x_n(\tau)\, d\tau \tag{6}$$

Previous works [2], [3], [5] shows the possibility to implement an analog adaptive structure in which simulation results shows that is possible to reach faster convergence time, around 0.5 μs and it depends of application. This structure can be integrated onto a single analog chip with the advantage that the power consumption is reduced and it can operate at higher speeds than DSP because there is no sample time and the speed limit is given by technology. With this in mind, figure 2 shows a block diagram of proposed structure.

In figure 2, the delay line is implemented with all pass analog function [6] except the first block which is a low pass analog function according to Laguerre polinomies. The coefficients are calculated with analog LMS algorithm. In figure 2, signal $x_n(t)$ is multiplied with error signal e(t) which is produced with the difference between y(t) signal and desired signal d(t). This product is integrated yielding coefficient a_n. Ones the error has been minimized, the block decision produce the load command to memory which store the corresponding digital value of coefficient a_n. At this point, all coefficients an has been stored and the can be used to reproduce the system that adaptive filter has been identified moving switch S_n to left position.

3 RESULTS

Figure 3, shows a Gabor wavelets using computer simulation, ones the Gabor wavelets was generated, a Laguerre adaptive filter structure in discrete time too,

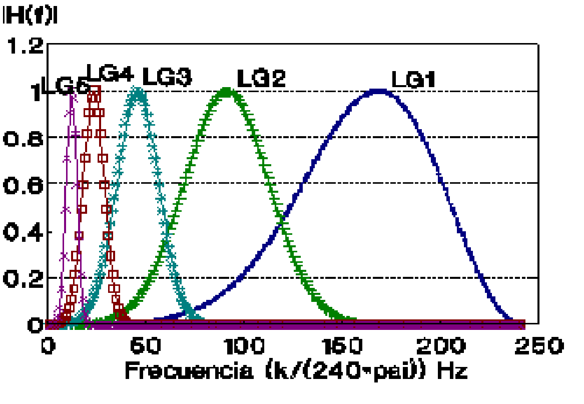

Figure 3. Gabor wavelets.

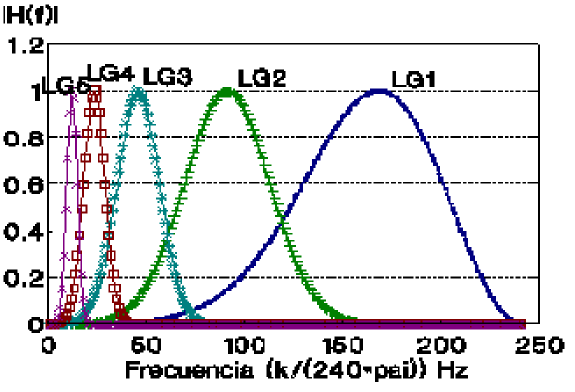

Figure 4. Gabor wavelets produced by discrete Laguerre filter.

was added to previous simulation in a system identification configuration. Figure 4 shows that Laguerre structure can identify the Gabor wavelets.

After this simulations, three analog band pass functions was simulated in PSPICE with the following characteristics:

$$\omega_1 = 333\ {}^{rad}\!/_{s},\ BW_1 = \frac{\omega_1}{3\sqrt{\ln 2}} = 133\ {}^{rad}\!/_{s}$$

$$\omega_2 = 666\ {}^{rad}\!/_{s},\ BW_1 = \frac{\omega_2}{3\sqrt{\ln 2}} = 266\ {}^{rad}\!/_{s}$$

$$\omega_3 = 1000\ {}^{rad}\!/_{s},\ BW_3 = \frac{\omega_1}{3\sqrt{\ln 2}} = 400\ {}^{rad}\!/_{s}$$

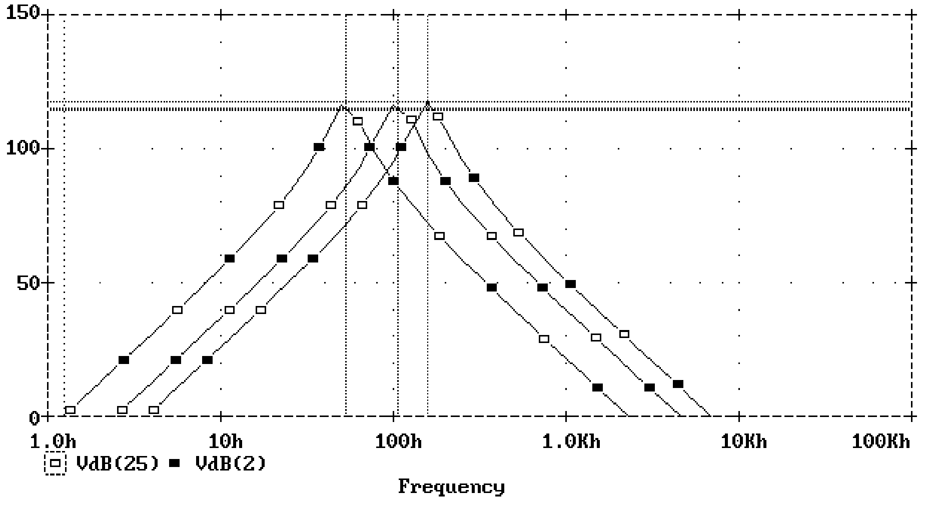

Figure 5. Output filter signal follows reference signal for three band pass filters.

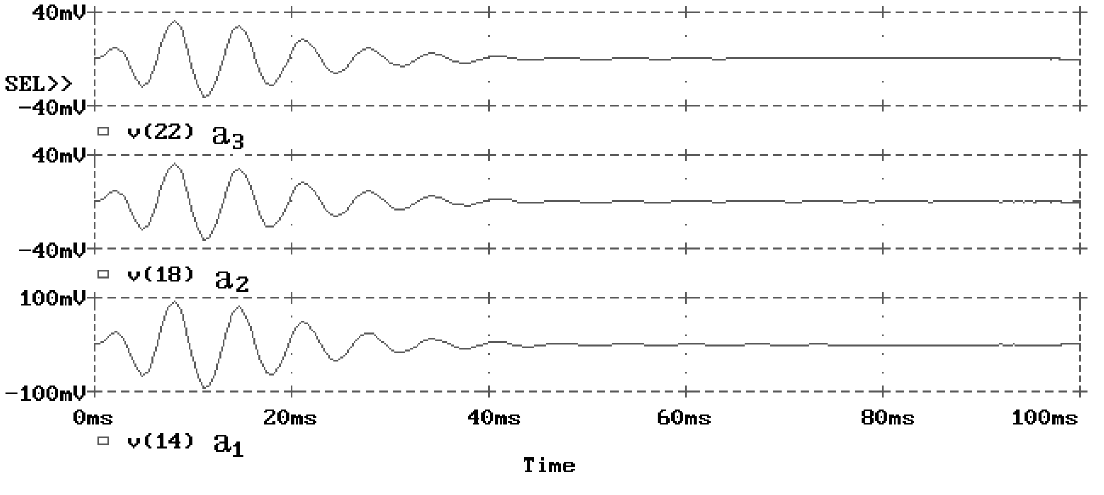

Figure 6. Coefficients convergence.

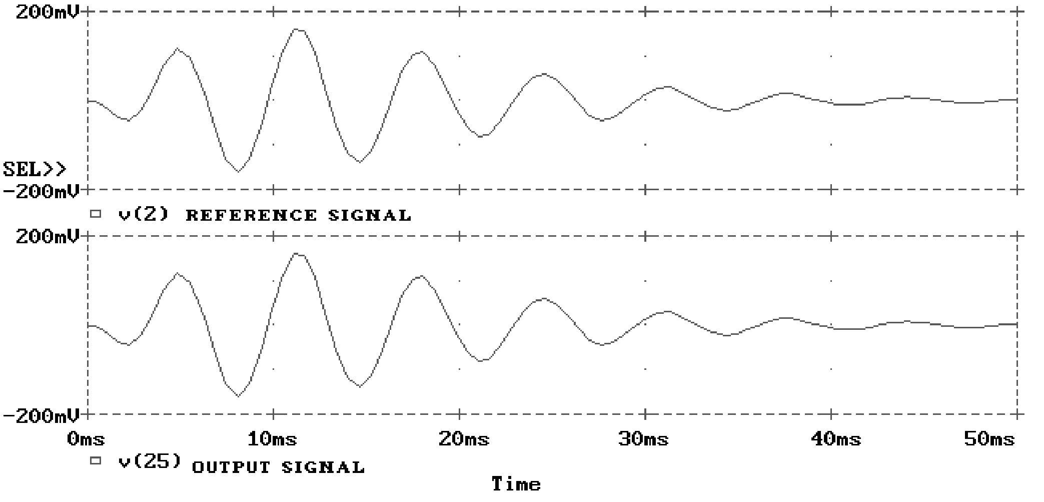

Figure 7. Time response of the reference signal and the outtput filter signal

In the same simulation, the adaptive analog Laguerre filter was implemeted with 3 taps put it in a identification configuration. Figure 5 shows that the analog adaptive filter can identify the transfer function of the band pass filter that in this case are the wavelets. In this figure we can see too how the output filter follows the reference signal. Figure 6 shows the convergence of the coefficients a_n.

Figure 7 shows the time response for desired signal d(t) an output signal from the adaptive filter y(t) of the one band pass filter.

4 CONCLUSION

This partial results shows that is possible to implement a wavelet transform in analog way. This could represent some advantage over implementations with DSP because the power consumption in a single analog chip with embedded memory must be lower that DSP.

Another one is that is possible to process signals with higher frequency and the size could be smaller than DSP implementation. Actually Laguerre analog adaptive fil ter VLSI structure is under construction. Of course, is necessary to probe its functionality and develop its possible applications.

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