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A compact digital gamma-tone filter processor

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ABSTRACT

Area consumption is one of the most important design constrains in the development of compact digital systems. Several authors have proposed making compact Cochlear Implant processors using Gamma-tone filter banks. These model aspects of the cochlea spectral filtering. A good area-efficient design of the Gamma-tone Filter Bank could reduce the amount of circuitry allowing patients to wear these cochlear implants more easily. In consequence, many authors have reduced the area by using the minimum number of registers when implementing this type of filter. However, critical paths limit their performance. Here a compact Gamma-tone Filter processor, formulated using the impulse invariant transformation together with a normalization method, is presented. The normalization method in the model guarantees the same precision for any filter order. In addition, area resources are kept low due to the implementation of a single Second Order Section (SOS) IIR stage for processing several SOS IIR stages and several channels at different times. Results show that the combination of the properties of the model and the implementation techniques generate a processor with high processing speed, expending less resources than reported in the literature.

1. Introduction

The development of digital artificial cochlear chips has attracted the interest of engineers for developing portable applications such as pitch detection, speech recognition and audio source localization on mobile devices, or for auditory prostheses [1]. These applications have used a model of the biological cochlea due to its capabilities for processing audio signals including natural sounds [2]. This cochlea functions as a transducer, converting the mechanical vibrations from the middle ear into electrical signals (auditory nerve spikes). These signals are sent to the human auditory system which responds to the information contained in the speech and audio signals.

Several studies show that the sound processing carried out by the cochlea, can be modeled using the over-complete Gamma-tone filter-bank, due to its resemblance to the human auditory system [2,3]. In addition, recently proposed mathematical models, reported in the literature, show that Gamma-tone filter banks designed using the impulse invariant transformation allow digital implementation of the analogue cochlea while employing reasonable computation with negligible distortion [4]. The hardware implementation of the cochlea, whether analogue or digital, is called an *artificial cochlea* chip or *silicon cochlea*, [5]. Because an efficient Cochlea chip is very important in several fields, the development of efficient cochlea chips has been an active research field. One of the first analogue silicon cochleae was developed by Lyon and Mead [4], using analogue VLSI 3 μm technology. It is reported that this cochlea chip, implemented using a cascade of 480 bi-quad filter sections, provides similar behavior than the human cochlea. A silicon cochlea which provides a good approximation of the human cochlea was proposed by Mandal et al. [6]. One of the most recent approaches was focused on building a bio-realistic analog CMOS Cochlea with high tunability and ultra-steep roll-off. The Chip response has high fidelity with respect to physiological experiments on mammalian cochlea and is 0.9 mm^2 in area and consumes 59.5 – 90.0 μW [7].

Analogue implementations of artificial Cochlea chips, such as the above, are potentially efficient in terms of processing speed and area when compared with the digital implementations. However, the analogue approach is susceptible to other factors, such as temperature, transistor mismatch and power supply noise [7]. To solve these problems several digital implementations of cochlear chips have been proposed aiming for efficient sound processors with minimal area. One of the critical factors to be considered in

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the development of artificial cochlea chips is the precision of the variables of the system. This factor has been taken into account in the system proposed by Van Immersel and Peters [8], which supports the emulation of the biquadratic filters with arbitrary precision and uses a scaling of the fixed-point precision as a method to avoid overflow. In [9], a cochlea chip is designed to work in real-time data maximizing the use of Hardware resources.

This paper proposes a digital cochlear processor based on a digital Gamma-tone filter bank to improve the area compared to existing digital designs. Our strategy maximizes the utilization of a single SOS stage to implement any SOS stages and several channels employing a time multiplexing technique. Indeed, the use of the impulse invariant transformation and the application of the normalization method to the filter coefficients allow creating an efficient filter bank processor in terms of processing speed and area resources, respectively.

Evaluation results are provided to show the desirable properties of the proposed system. The rest of the paper is organized as follows. Section 2 presents the gamma-tone filter bank model. Section 3 presents the overview of the Gamma-tone Filter Processor. Evaluation results are provided in Section 4. Section 5 shows the improvement achieved by our proposal when compared with the current approaches. Finally Section 6 provides the conclusion of this work.

2. Review of the gamma-tone filter bank

Gamma-tone filters are defined in the continuous time domain, and can be mathematically modeled, in the discrete time domain, using different techniques of digital signal processing such as the impulse invariant transformation, the Z-matched transform and the bilinear transform. To select an appropriate transformation, in addition to the computational complexity, the introduced distortion must be considered. To this end, several works [8,10,11] show that the model employing the impulse invariant transformation provides the best performance because it has a low computational cost and lower distortion when compared with the Z-matched and bilinear transformations. Thus according to the previous published works [8,11] the impulse response of a Gamma-tone filter of order α, is given by the product of a gamma function multiplied by a cosine function, as follows:

$$\psi_{f_c}^{\alpha}(t) = \frac{1}{(\alpha-1)!} t^{\alpha-1} e^{-2\pi b_m t} \cos(2\pi f_c t) u(t) \tag{1}$$

where α is the filter order, b_m is the m'th filter bandwidth in Hz and f_{cm}, is the resonance frequency. Next using the Euler representation of cosine function in Eq. (1) and taking the Laplace transform of the resulting equation, after some manipulations it follows that the transfer function of the m'th gamma-tone and pass filter $H_m(s)$ can be represented in terms a cascade of α second order band pass filters as follows [8,10,11]:

$$H_m(s) = \left(\frac{K(s+4\pi b_m)}{(s+2\pi b_m - j2\pi f_{cm})(s+2\pi b_m + j2\pi f_{cm})} \right)^{\alpha} \tag{2}$$

Because the proposed system will be implemented in the discrete time domain, Eq. (2) must be transformed to the z-domain. To this end, the impulse invariant transform is used because it provides a discrete time version of Eq. (2) with less frequency distortion as compared with the bilinear and Z-matched transforms. Thus, applying the impulse invariant transform to Eq. (2), it follows that

$$H_m(z) = \left(\frac{2 - 2B_m z^{-1}}{1 - 2B_m z^{-1} + C_m z^{-2}} \right)^{\alpha}, \tag{3}$$

where

$$B = e^{-2\pi b_m T} \cos(2\pi f_{cm} T) \tag{4}$$

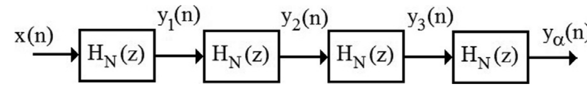

Fig. 1. Block diagram of the Gamma-tone structure implemented as a cascade of SOS IIR identical stages.

$$C_m = e^{-4\pi b_m T} \tag{5}$$

As shown in Eq. (3) the Gamma-tone filter transfer function, can be obtained as a cascade of α filters of second order with complex conjugated poles. Thus it is important to normalize the gain of each stage because it allows us to factorize the transfer function of the gamma-tone filter in α identical SOS stages. This avoids the recalculation the filter coefficients when the value of α is changed. To this end, because each stage represents a second order band pass filter transfer functions, consider the frequency response of a second stage evaluated in $2\pi f_{cm}$, which is given by

$$H_m(f_{cm}) = H_m(z)|_{Z=e^{j2\pi f_{cm}}} = \left. \frac{2 - 2B_m z^{-1}}{1 - 2B_m z^{-1} + C_m z^{-2}} \right|_{Z=e^{j2\pi f_{cm}}} \tag{6}$$

where f_{cm} is the resonance frequency. Thus from Eq. (6) it follows that

$$H_m(f_{cm}) = \frac{2 - 2B_m e^{-j2\pi f_{cm}}}{1 - 2B_m e^{-j2\pi f_{cm}} + C_m e^{-j4\pi f_{cm}}}, \tag{7}$$

whose magnitude is given by

$$|H_m(f_{cm})| = \sqrt{\frac{(2 - 2B_m \cos(2\pi f_{cm}))^2 + 4B_m^2 \sin^2(2\pi f_{cm})}{(1 - (2B_m + C_m)\cos(2\pi f_{cm}))^2 + (2B_m + C_m)^2 \sin^2(2\pi f_{cm})}}. \tag{8}$$

Thus normalizing the magnitude of $H_m(f_{cm})$, from Eqs. (3) – 5 and 8 it follows that

$$H_m^N(z) = \left(\frac{A_m^N - B_m^N z^{-1}}{1 - 2B_m z^{-1} + C_m z^{-1}} \right)^{\alpha}, \tag{9}$$

where

$$A_m^N = \frac{2}{|H_m^N(f_{cm})|} \tag{10}$$

and

$$B_m^N = \frac{2e^{-2\pi b_m T} \cos(2\pi f_{cm} T)}{|H_m^N(f_{cm})|} \tag{11}$$

Next the normalized filter coefficients, A_N and B_N, provide the SOS stages with gain equal to one. Finally taking the inverse Z-transform of Eq. (9) we obtain the output of the gamma-tone filter which is given by

$$y_i(n) = A_i^N y_{i-1}(n) - B_i^N y_{i-1}(n-1) + 2B_i y_i(n-1) - C_i y(n-2) \tag{12}$$

The above synthesis method provides a systematic procedure that allows the implementation of the Gamma-tone filter by cascading identical second order stage independently on the value of α. It is important because it is known that the order of Gamma-tone filter can vary depending on the application and the type of signals to be processed.

Thus, because the filter can be implemented or represented by a cascade of SOS IIR band pass filters, if a given application requires the implementation of a Gamma-tone filter with $\alpha = 4$ it is necessary to use four SOS IIR band pass filters connected in cascade as shown in Fig. 1, where each block represents a second order IIR filter. Consequently the value of α is equal to the