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Adaptive noise filtering based on artificial hydrocarbon networks: An application to audio signals

Hiram Ponce *, Pedro Ponce, Arturo Molina

Graduate School of Engineering, Tecnologico de Monterrey, Campus Ciudad de Mexico, 14380 Mexico City, Mexico

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ABSTRACT

Many audio signal applications are corrupted by noise. In particular, adaptive filters are frequently applied to white noise reduction in audio. Recent work provides that there exist some insights on using an artificial intelligence method called artificial hydrocarbon networks (AHNs) for filtering audio signals. Thus, the scope of this paper is to design and implement a novel approach of artificial hydrocarbon networks on adaptive filtering for audio signals. Three experiments were developed. Results demonstrate that AHNs can reduce noise from audio signals. A comparison between the proposed algorithm and a FIR-filter is also provided. The short-time objective intelligibility value (STOI) and the signal-to-noise ratio (SNR) were used for evaluation. At last, the proposed training method for finding the parameters involved in the AHN-filter can also be used in other fields of application.

1. Introduction

One of the problems in audio signal applications is the addition of noise signals that interfere with the original ones arriving in poor performance of audio. For human ears, this causes non-intelligibility of audio and for signal analysis it causes imprecise data information that blinds important features; which it implies a tradeoff between noise supression and signal distortion. In order to face it, different audio filters have been developed in audio signals (Parks & Burrus, 1987; Smith, 1997; Winder, 2002).

In general, audio filters are analog or digital (Winder, 2002). The following work considers digital audio filters, e.g. filters implemented in software. In digital signal processing, audio filters can be classified into finite impulse response (FIR) filters and infinite impulse response (IIR) filters. The first one uses a filter kernel that reaches a finite zero frequency response while the second one uses a filter kernel that responds in frequency with infinite exponential decaying sinusoidal functions (Smith, 1997). In practice, convolution implements FIR filters and recursion implements IIR filters. In advance, FIR filters use convolution of the input signal while IIR filters uses convolution of the input but also of the output signals (Smith, 1997). In so many cases, FIR filters are more used than IIR filters, primary based on their phase response in frequency domain (Clark, 2005). For instance, FIR and IIR filters can be compared (in frequency domain) in their magnitude and phase responses. As an example, consider the filter response of a sinc function as an input signal like both FIR and IIR filter responses shown in Fig. 1. Ideally, the frequency response in magnitude corresponds to a rectangular-shaped function (Clark, 2005). However, as shown in Fig. 1, the response of the IIR filter is smooth while the response of the FIR filter contains ripple, and both filters try to exhibit a rectangular-shaped function. As observed, IIR filters might be better used, but in phase response FIR filters present a desired linear characteristic as shown in Fig. 2. Notice that FIR filters (Fig. 2(a)) present a linear phase response while IIR filters (Fig. 2(b)) do not. This behavior is crucial when choosing for FIR or IIR filters because nonlinear phase responses might not be demodulate correctly (Clark, 2005).

Several audio filtering applications consider two interests: improving either time domain response or frequency domain response. Smoothing, noise reduction and DC removal are typical problems in time domain while separating frequencies is typical in frequency domain (Smith, 1997). However, it is very difficult to improve time and frequency responses with the same filter. Thus other techniques have been proposed. For example, the moving average and windowed-sinc filters are widely used in time domain like reported in Khan, Okuda, and Ohba (2003), Smith (2007) and Thede (1996). In contrast, the single pole filter and Chebyshev's filter are used in frequency domain (Bai & Lu, 2005; Smith, 1997).

The above examples fall into linear filters. These kinds of filters have a linear response from the input signal. However, there are other prominent filters that respond in a nonlinear way from the input signal. Classical nonlinear filters are based on the time-varying Wiener's filter (Chen, Benesty, Huang, & Doclo, 2006), probabilistic filters (Burnshtein & Gannot, 2002; Huda, Yearwood, &

* Corresponding author. Tel.: +52 5554832020.
E-mail addresses: hiram.ponce@itesm.mx (H. Ponce), pedro.ponce@itesm.mx (P. Ponce), armolina@itesm.mx (A. Molina).

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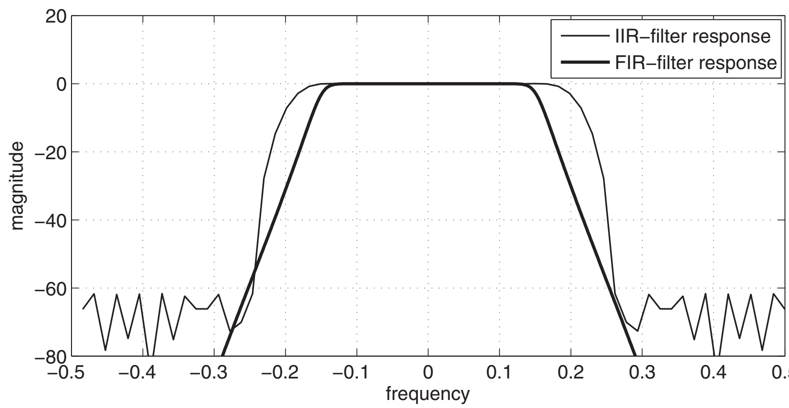
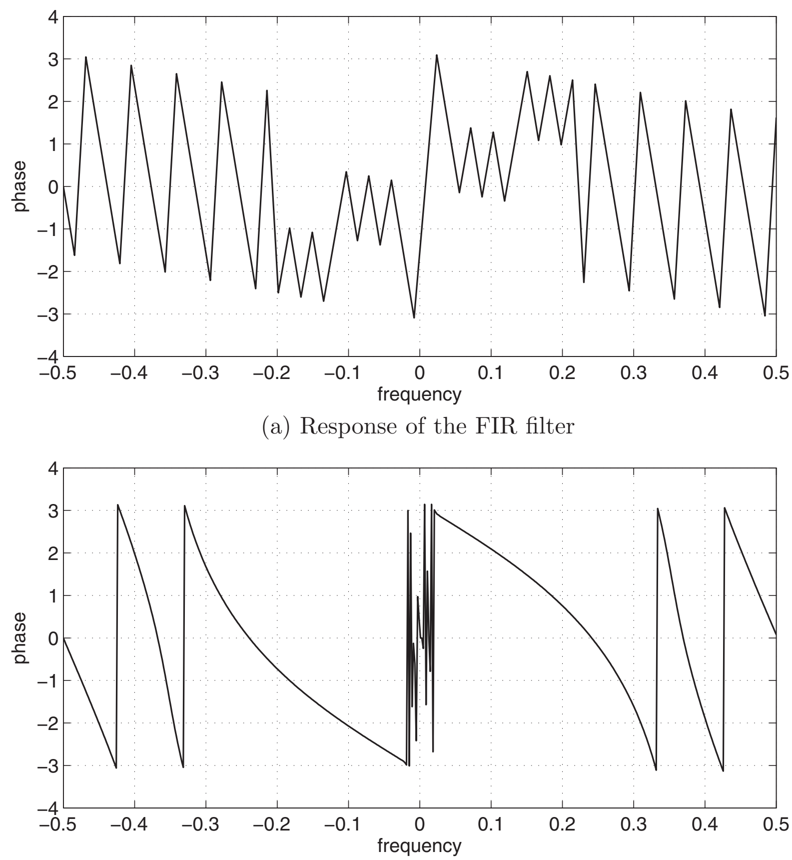


Fig. 1. Comparison between FIR and IIR responses in magnitude.



(a) Response of the FIR filter

(b) Response of the IIR filter

Fig. 2. Comparison between FIR and IIR responses in phase.

Togneri, 2009; Vermaak & Niranjan, 1998) and homomorphic filters (Smith, 1997). Examples of these algorithms can be found in Comon and Laocume (1986), Manolakis, Kalouptsidis, and Carayannis (1983), Smith (1997) and Diniz (2008).

Consider a special kind of noise so-called white or uniform noise. In particular, white noise reduction cannot be handled easily in audio signals because it contains similar low magnitude components in all the spectrum of frequencies that is typically confusing or overlapping with mainly voice signals and musical signals (Smith, 1997, 2007; Thede, 1996). In that sense, linear filters cannot clean the signal efficiently and nonlinear filters for these purposes have been proposed (Diniz, 2008; Ling, 2007). For example, Baravdish, Evangelista, Svensson, and Sofya (2012) reported an audio denoising using a singular value decomposition (SVD) matrix to determine the most significant components in audio and then to eliminate the less significant ones. In fact, their approach also considers the usage of the inverse of partial differential equations that actually smooth the signal, resulting in a good performance denoising signal in a nonlinear fashion. However, computational effort increases for nonlinear cases. On the other hand, LeRoux and Vincent (2013) proposed a consistent Wiener filtering for audio denoising and source separation. Their proposal considers to improve the Wiener filtering applied as a short-time Fourier transform (STFT) in order to relate short-time responses of STFT of different bins such that the redundancy present in STFT responses will act as a hard constraint or as a soft penalty. They found that this technique can improve source separation, and that it can be exploited in audio denoising. Other example applications can be found in