Reprint From: 3rd Int. Symposium on Communication Systems, Networks and DSP 2002, Stafford, Untied Kingdom, pp. 9-12, July 2002

VLSI Implementation of an Adaptive Noise Canceller

A.Th. Schwarzbacher[1,++] and J. Timoney[2]

[1]Dublin Institute of Technology, Dublin, Ireland
[2]NUI Maynooth, Ireland

Abstract: Adaptive Noise cancelling is a speech-specific filtering technique to suppress additive interference. It is a sensible choice for speech-orientated devices in situations where the designer has no knowledge of the properties of interference that could potentially corrupt the input speech to a point beyond intelligibility. A good example of devices subject to such noise is mobile telephones, where the user may be communicating in a variety of environments. However, constraints appear when implementing a noise-cancelling algorithm on such a portable device, and issues of power consumption and silicon area become prominent. This paper discusses an adaptive noise cancelling scheme and presents a VLSI implementation strategy. It then considers some of the benefits of the implementation and its potential area for improvement.

Keywords: Adaptive noise cancelling, VLSI design, speech enhancement

1. Introduction

The advent of mobile communications over the past decade has dramatically affected the way people live and communicate. Mobile telephones have become an everyday item using ever-increasing amounts of the available bandwidth. Equipment manufacturers are spending more and more money researching ways to maximise the usage of the limited amount of available bandwidth allocated to wireless communications, whilst examining methods to improve signal to noise ratios and to increase battery life time.

Mobile phones are often used in noisy environments, the result being additional unwanted noise signals are transmitted with the speaker's voice over the channel. At the receiving end the speaker's voice can become unintelligible amongst the noise.

Presently, bandwidth limitation is the only technique applied to the voice signal and its effectiveness is inadequate in the presence of high to severe noise. However, if adaptive noise cancelling devices were to be incorporated into future mobile communication equipment, users would enjoy better communication with the added benefit of extended battery life. Service providers would gain by being able to take advantage of better signal to noise ratios, which would allow them to support more users per mobile cell and reduce the transmission power required to achieve a suitable SNR.

In this paper a VLSI implementation of an adaptive noise canceller (ANC) circuit is presented. This LMS driven adaptive filter structure is one of many possible methods of speech enhancement. It performance relative to its computational complexity makes it a good choice for portable applications. Furthermore, the ANC device has been shown to provide best performance in the presence of low SNRs.

1.1 Principle

Unlike many adaptive filtering solutions, as shown in Figure 1, the ANC device examined here forms a noise reference signal from the input, thus removing the need for more than one microphone [1] [2]. This is possible as voice speech is a quasi-periodic signal [3]. Therefore, if a portion of the input signal (noise and voice) is delayed by an amount corresponding to one period the voice component will be highly correlated, while the additive noise will be uncorrected, under the assumption of Gaussian properties [1].

++ Author to whom correspondence may be directed: andreas.schwarzbacher@dit.ie
Andreas Schwarzbacher, Dublin Institute of Technology,
School of Electronic and Communications Engineering, Kevin Street, Dublin 8, Ireland

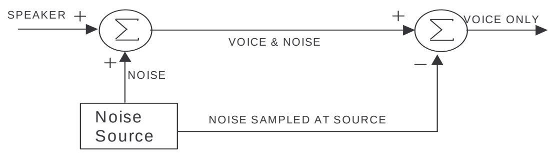

Figure 1: Solution to Eliminate the Noise in a Signal

A block diagram of the ANC is shown in Figure 2. The input speech *s(n)* and its delayed version *s(n-T)*, are highly correlated, where *T* is its pitch period. *w(n)* represents additive white noise and is thus uncorrelated with *w(n-T)* or the speech signal. The role of the adaptive filter is to minimise the energy of the system output *w'(n)* and thus produce a signal *s'(n)* that is, in the LMS sense, the optimum estimate of *s(n)*. Both the speech and noise are time varying and thus the filter must adapt to changes in the properties of the input, and it is the LMS that determines the trajectory of the changing filter weights in order to keep the energy of its output at a minimum [4]. Furthermore, the dynamics of speech means its pitch does not remain constant and thus, for best performance, this has to be estimated in an iterative manner by some pitch estimation module. In the case of unvoiced speech however, no pitch value exists and the simplest solution is to freeze the filter weights at their current value until voiced speech occurs again [1].

2.1 Pitch Estimation

Pitch estimation of the voiced speech was carried out using the Average Magnitude Difference Function (AMDF) [5] and associated decision logic was included to determine if the sound is voiced or unvoiced [6]. The AMDF is defined by the relation [5];

$$D_\tau = \frac{1}{L}\sum_{j=1}^{L}\left|S_j - S_{j-\tau}\right| \qquad \tau = 0,1, \ldots \tau_{max}$$

Where

S_j are the samples of the input speech

$(S_j) = (S_1, S_2, \ldots S_L)$

$S_{j-\tau}$ are the samples shifted τ seconds (1)

The AMDF was chosen mainly because no multiply operations are required for its calculation and thus its computational efficiency make it suitable for implementation on a programmable processor or in special purpose hardware [3].

A block diagram of the hardware implementation of the AMDF and voiced/unvoiced decision module [8] is given in Figure 3. This figure shows the individual modules as well as the local controller. However, to keep the diagram readable the control structure is indicated in a simplified manner. The designs of this project were fully described in VHDL and synthesised using a ES2 0.7µm technology.

2.2 Adaptive Filter

The adaptive filter section of the ANC device is FIR and the filter coefficients are updated using the LMS algorithm. When considering an implementation of an adaptive filter, FIR filters are much more preferable to IIR filters because they have a linear phase response, they exhibit low sensitivity to finite wordlength effects and roundoff and quantisation errors have less impact on their performance [4]. Choosing the number of taps for the FIR filter is of critical importance for efficiency. If there is a large number of coefficients, more multiplications per sample are required. The LMS algorithm [4] shown below

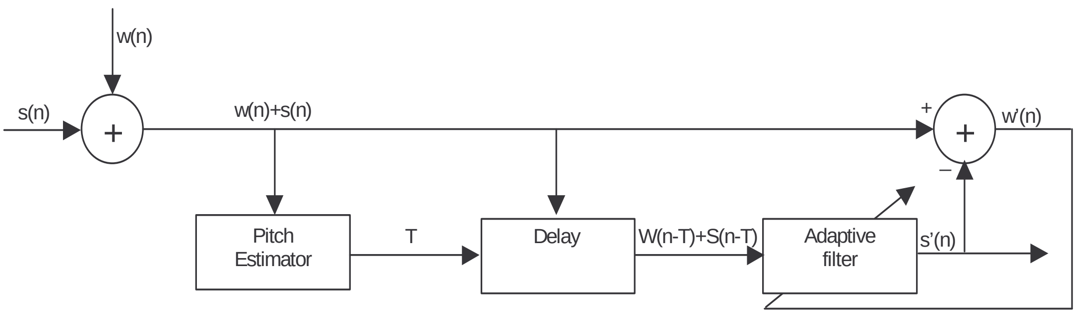

Figure 2: Adaptive Noise Canceller

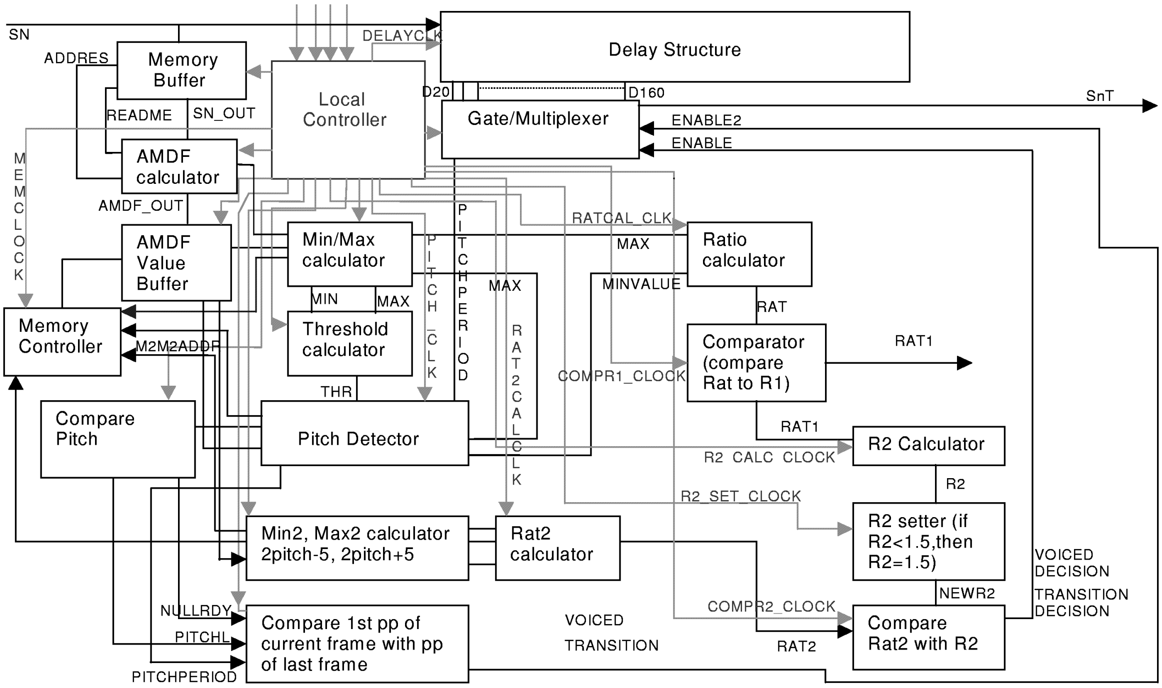

Figure 3: Block Diagram of the Pitch detection Section of the ANC

provides the mathematical method of calculating the filter coefficients for each tap;

LMS ALGORITHM :

$$B(n+1) = Bn + 2.MU.e(n).X(n-T)$$

Where

B(n+1) = Next filter coefficient

Bn = present coefficient

MU = step size

X(n-T) = Previous delayed sample

Where the step size (*MU*) controls the rate of convergence of the filter.

Despite the apparent simplicity of the LMS algorithm there are some limitations in practice. For example, a practical problem exists in a rapidly changing environment where the error signal has just been minimised and the coefficients optimised by the algorithm for that sample but then the next sample changes radically and the error becomes large again [4]. Fortunately, under the assumption of Gaussian statistics for the noise and knowing that the properties of speech signals do not fluctuate rapidly, this effect is not a significant issue.

Following extensive simulations, the filter design implemented utilised an 11 tap, 10 delay line structure The response for an 11 tap filter compares favourably with the response for a 20 tap structure. Such a large structure would be unfeasible due to the number of multiplications that would be involved and the fact that if implemented it would occupy twice the silicon area and hence twice the cost with no justifiable improvement over an 11 tap design.

Figure 4 illustrated the block diagram of the adaptive filter implementation. Due to the slow data throughput rates of speech the filter as well as the coefficient multiplier is implemented sequentially to save silicon area.

2 Conclusions

This paper presented the outline a hardware implementation structure of an adaptive noise canceller for speech signals. When presenting the individual blocks, two main aspects were highlighted. First the computational efficiency of the algorithms used and the corresponding ease of implementations, and second the robustness of the theoretical algorithms and the robustness of the resulting implementation towards effects of finite wordlength. This has resulted in an implementation that can be adapted for various applications such as mobile phones, portable voice controlled devices, and possibly hearing aids.

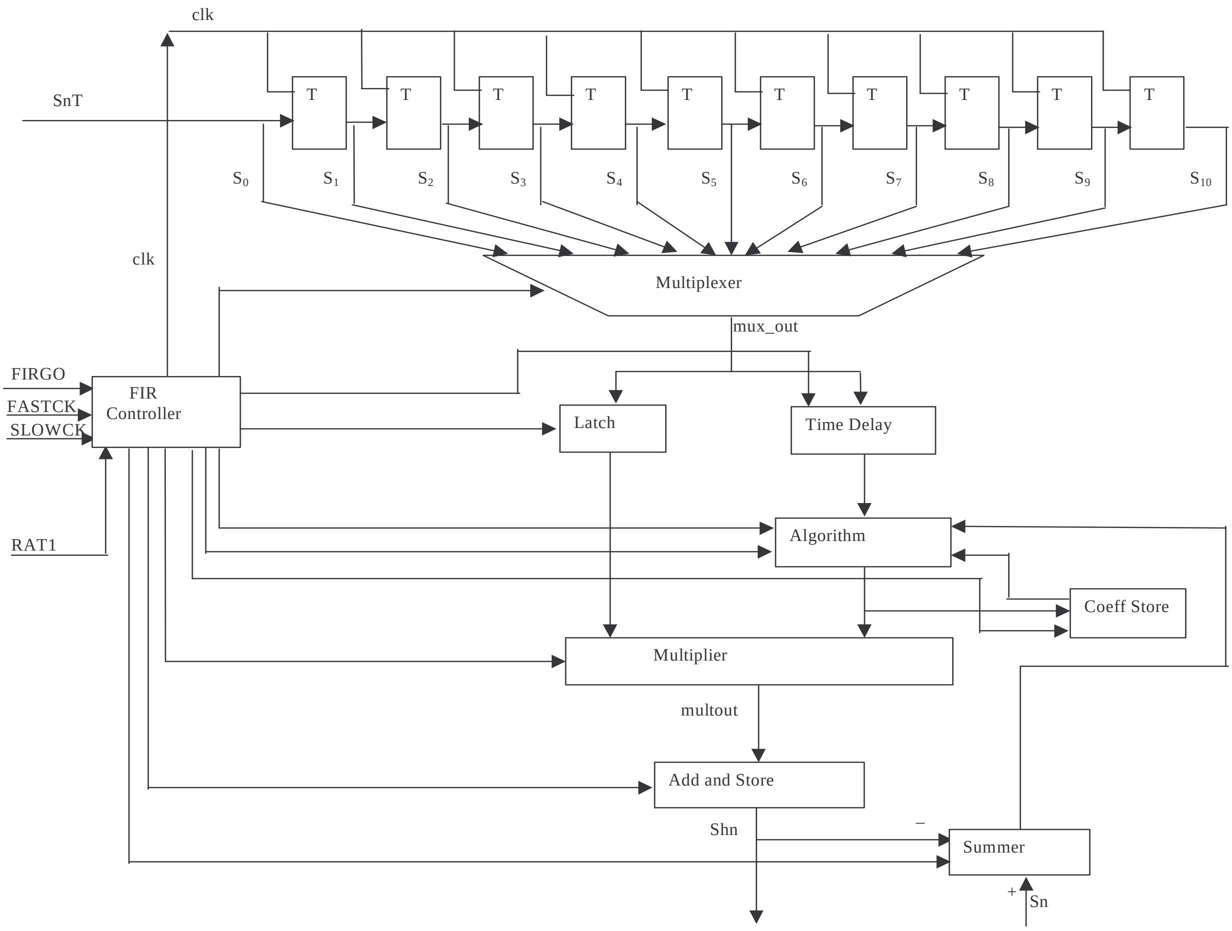

Figure 4: Block Diagram of the Adaptive Filter

In the next stage of the project the design will be extended to embrace the benefits of the sub-band adaptive filtering structure, and furthermore, perceptual issues in relation to the parameters of sub-band filters will also be examined [7]. Once this is established, the design will be rigorously tested and studied to find non-standard implementations of elements that will bring reductions in the power consumption and silicon area. The final aim is to produce a robust prototype that could be transformed into a commercial product.

References

[1] M. R. Sambur, ``Adaptive Noise Canceling for Speech Signals," *IEEE Trans. on Acoustics, Speech and Signal Processing*, 26(5):419--423, October 1978.

[2] M. R. Sambur, "LMS Adaptive Filtering for Enhancing the Quality of Noise Speech," *Proc. IEEE ICASSP*, 1978, pp 610-613.

[3] L.R. Rabiner and R.W. Schafer, *Digital processing of speech signals*, Prentice-Hall, Englewood Cliffs, NJ, 1978.

[4] B. Widrow et al, *Adaptive signal processing*, Prentice-Hall, London, 1985.

[5] G.S. Ying, L.H. Jamieson, and C.D. Mitchell, "A Probabilistic Approach to AMDF Pitch Detection,", *Proceedings of the Fourth International Conference on Spoken Language Processing*, Philadelphia, PA, October 1996.

[6] W. Hess. *Pitch Determination of Speech Signals*. Springer-Verlag, Berlin, 1983.

[7] W. Kellermann, "Analysis and Design of Multirate Systems for Cancellation of Acoustical Echoes," *Proc. IEEE ICASSP*, 1988, pp. 2570-2573.

[8] C. Eroglu, "Average magnitude difference function (AMDF)," June 1996, Http://www.ee.bilkent.edu.tr/~eroglu/report/node3.html