CMOS IMPLEMENTATION OF AN ADAPTIVE NOISE CANCELLER INTO A SUBBAND FILTER

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ABSTRACT

In recent years the demand for mobile communication has increased rapidly. While in the early years of mobile phones battery life was one of the main concerns for developers speech quality is now becoming one of the most important factors in the development of the next generation of mobile phones.

This paper describes the CMOS implementation of an adaptive noise canceller (ANC) into a subband filter. The ANC-Subband filter is able to reduce noise components of real speech without prior knowledge of the noise properties. It is predestined to be used in mobile devices and therefore, uses a very low clock frequency resulting in a small power consumption. This low power consumption combined with its small physical size enables the circuit also be used in hearing aids to efficiently reduce noise contained in the speech signal.

1. INTRODUCTION

The origin of noise in a speech signal can be either ambient noise or quantisation noise by analogue to digital convertion. Noise degrades the quality of speech signals and must therefore be removed. If the noise is known a priori, it could be simply subtracted from the noisy speech signal. The result would be a perfectly clean signal. This principle is used in headsets for pilots where a reference microphone is installed at a sufficient distance to the pilot that only picks up the noisy signal which is then subtracted form the microphone on the headset [1]. However, this is not practicable in mobile devices because a second microphone can not be placed at a sufficient distance from the speech microphone due to the small size of the devices. Therefore, another approach was found to extract the noise signal and the clean speech signal from one source as described in [2]. The quasi-periodic property of voiced speech is exploited to achieve this goal.

The design described in this paper consists of an Adaptive Noise Canceller (ANC) [3] and a Subband Filter [4]. The main part of the ANC is a Finite Impulse Response (FIR) Filter. Its output is defined by the equation

$$y(n) = \sum_{i=0}^{L} b_i x(n-i-T) \tag{1}$$

where x is the noisy speech signal, L is the filter order, b_i are the filter coefficients and T is the pitch period of the speech signal. In account of the changing conditions of speech signals, the frequency responce needs to be updated permanently. The Normalised Least Mean Square (NLMS) [5] Algorithm is used to update the coefficients. The estimation error of the filter $e(n) = x(n) - y(n)$ is defined as the difference between input signal and estimated filter output. This error is used in the LMS [2] algorithm

$$b_{n+1} = b_n + 2\mu \cdot e(n) \cdot X_{n-T} \tag{2}$$

to update the coefficients and thus to adjust the filter response. The stepsize μ controls stability as well as the rate of convergence. It is a fixed value for a LMS implementation, but for the NLMS algorithm used in this filter μ is constantly updated using the following equation.

$$\mu = \frac{a}{b + \sum_{i=0}^{L} x_i^2} \quad \text{with} \quad 0 < a < 2 \tag{3}$$

Here x_i are the filter inputs and L is the filter order. This dynamic update provides a fast rate of convergence while keeping the filter stable [5]. A block diagram of this approach is shown in Figure 1.

The pitch period for the adaptive filter is provided by the pitch detector [6] for every input frame. Pitch detection is only possible for voiced speech due to the quasi-periodic character of voiced speech. The required voiced/unvoiced decision is also performed by the pitch detector for each input frame. To distinguish between voiced and unvoiced frames three different methods are implemented in the pitch detector of the ANC. These are Short-Term Energy measure, Zero Crossing detection [7] and the AMDF (Average Magnitude Difference Function) Min/Max Ratio of a pitch detector output frame. The pitch period for voiced frames is extracted using the AMDF [8]. A block diagram of the ANC is shown in Figure 2.

The schematic of the subband filter is shown in Figure 3. The subband filter divides the input signal into four frequency bands. The bitrate in each of these subbands is by a factor of four smaller than the input bitrate. One ANC is embedded in each subband. Therefore, each ANC has to cancel noise in one subband only. To split and combine the signals the subband filter uses tree structured analysis and synthesis FIR filter banks [5].

The analysis bank contains decimation filters which split the input signal into subband signals with a lower information content then the input signal. Each stage of the analysis bank consists of a low pass and a high pass FIR filter. Both the upper and lower channels are then down-sampled by a factor of two. At an input sample rate of 8 kHz each of the four suband results in a sample rate of 2 kHz.

The synthesis bank is necessary to combine the subband signals and form the output of the filter. Each stage of the synthesis bank consists of two up samplers with attached interpolation filter. These interpolation filters are required to reduce the effect of imaging in the up-sampled signals. At the end of each synthesis stage the upper and lower bands are are added together.

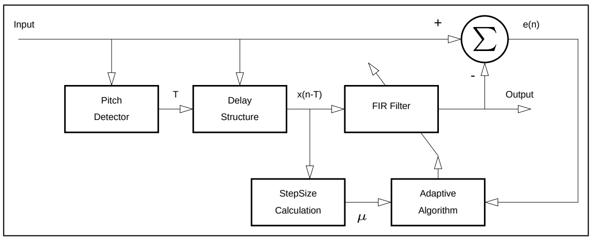

Figure 1: Overview of the ANC.

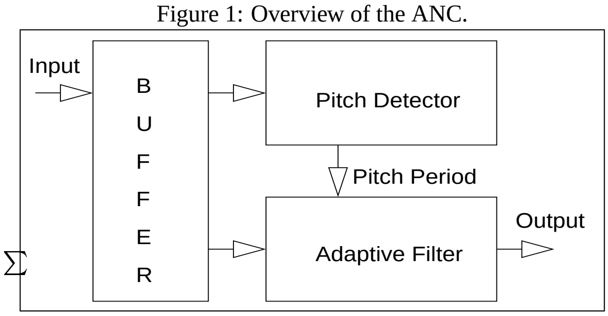

Figure 2: Block Diagram of the ANC.

2. IMPLEMENTATION

The circuit was implemented using the European Silicon Structures $0.7\mu m$ CMOS technology. All designs were written in VHDL and synthesised using the Synopsys Design Compiler without any design constrains. Figure 4 shows the block diagram of the structure of the ANC-Subband filter. The design consists of three main sections. The analysis bank, which incorporates three decimation filters, four ANCs, one in each subband and finally three stages of interpolation filters and adders for the synthesis bank.

The device uses three different clock frequencies. An 8 kHz clock to synchronise to the input samples which arrive at this frequency. An 800 kHz clock is used by the subband filter to perform the computations needed to split up the signal into the four subbands. The third clock oscillates at 5.7 MHz and it is used by the ANC to perform all necessary calculations to clear the input signal from noise. The silicon area of the overall filter structure is 70nm^2.

3. RESULTS

To benchmark the ANC-Subband Filter real speech signals were used. These signals provide variable pitch frequencies as well as voiced and unvoiced speech. All signals were distorted by White Gaussian Noise (WGN) with a Signal to Noise Ratio (SNR) of 10 dB to 0 dB.

Spectrograms of the results are presented in Figure 5 to Figure 14. These spectrograms show the energy distribution of a signal over the frequency depending on the time. Figure 5 and Figure 8 show the undistorted speech signals of a female ("Her wardrobe consists of only skirts and blouses") and a male ("Are you looking for employment?") speaker. These two sentences are used in this paper to illustrate the behaviour of the subband filter. However, additional tests have been carried out with over 25 speech samples with similar results.

Figure 6 and Figure 7 show the sample of female speech before and after filtering, the SNR of the distorted input sample is 10 dB. Figure 9 and Figure 10 show the sample of male speech respectively. The results of a 0 dB SNR input speech of the same speakers are presented in Figure 11 and Figure 12 as well as Figure 13 and Figure 14. It should be noted that darker colours in the spectrograms represent areas of lower energy.

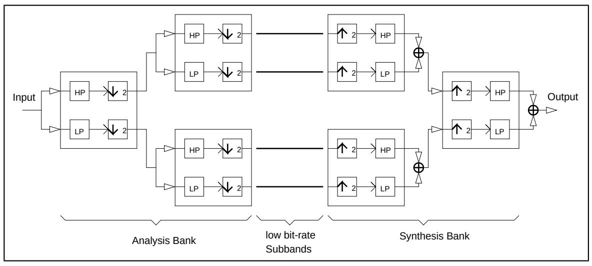

Figure 3: Block Diagram of the Subband Filter.

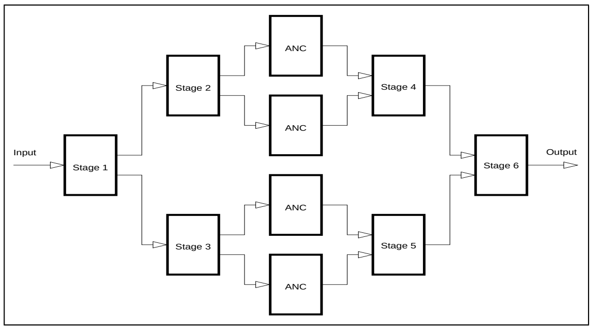

Figure 4: Block Diagram of the Combined ANC - Subabnd Filter.

It can be seen that the noise component is significantly reduced in all filtered signals. Additionally, the spectral shape in the voiced speech sections is preserved. Thus, the perceptual quality of the speech is improved. Additionally, extensive qualitative auditory tests confirmed that after processing the quality of a noisy signal was significantly improved.

4. CONCLUSIONS

The objective of this paper was to present a hardware implementation of an ANC-Subband Filter to significantly improve the audibility of noisy speech. Four ANCs were incorporated into the four subbands of a subband filter with additional hardware optimisation of the overall structure. Due to the parallelisation of four ANCs the maximum clock frequency which is necessary to perform all computational steps in real time could be reduced from 22.7 MHz to only 5.7 MHz, resulting in nearly 4 times reduction in power consumption.

By using objective speech measure as well as qualitative auditory tests it has been shown that the filter is able to reduce noise in a speech signal, thus increasing the quality of the signal. The subband filter can therefore, be included in mobile communication devices such as hearing aids or mobile phones. In conclusion, this paper presented a CMOS based ANC-subband filter providing good noise cancelling performance under real world conditions.

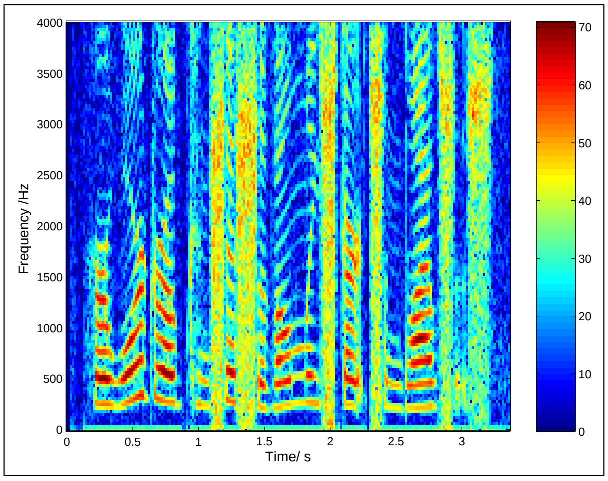

Figure 5: Spectrogram, Female Reference Speech.

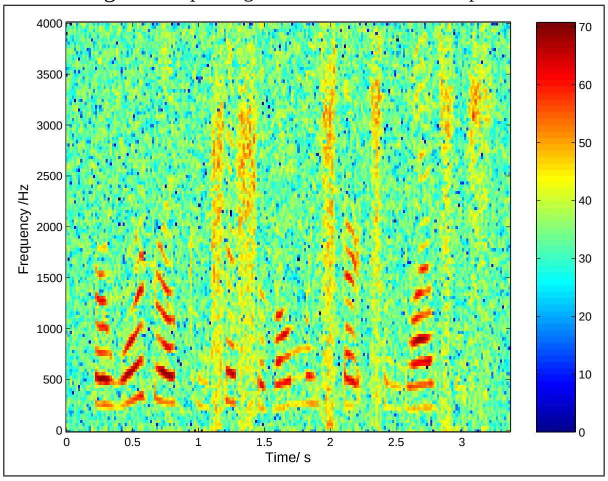

Figure 6: Spectrogram, Female 10 dB SNR, Input.

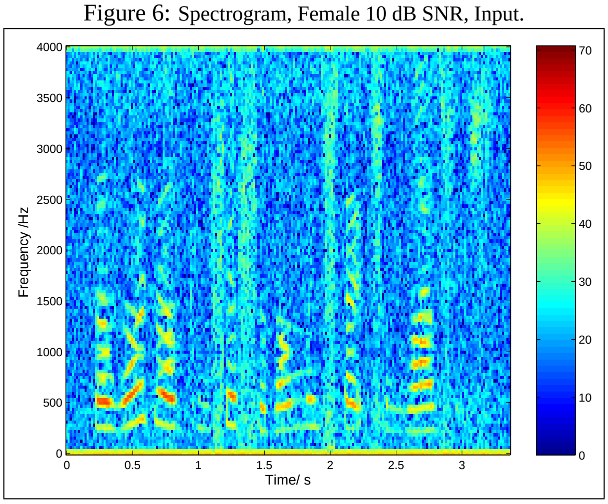

Figure 7: Spectrogram, Female 10 dB SNR, Output.

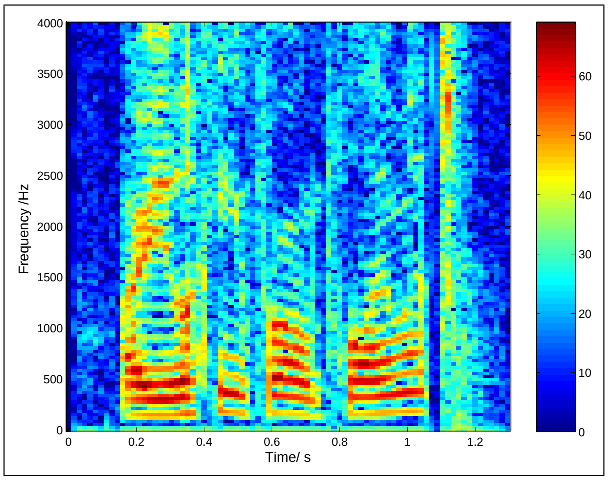

Figure 8: Spectrogram, Male Reference Speech.

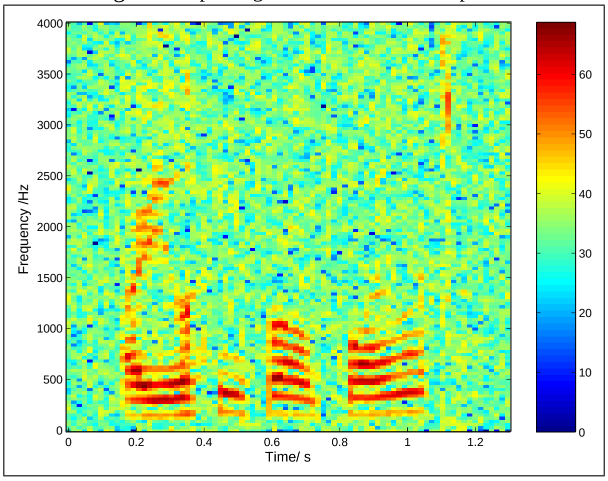

Figure 9: Spectrogram, Male 10 dB SNR, Input.

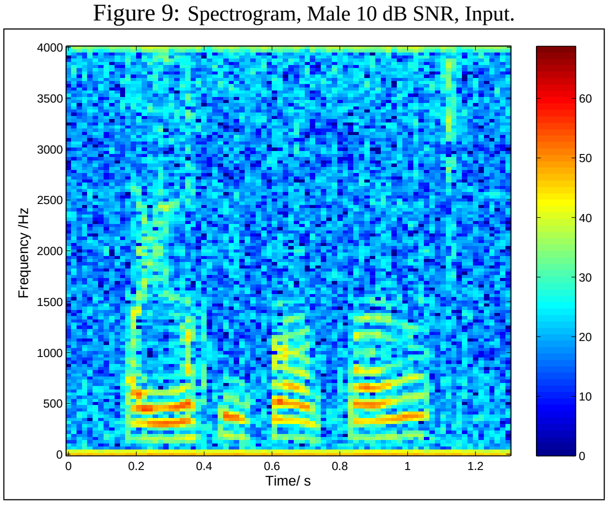

Figure 10: Spectrogram, Male 10 dB SNR, Output.

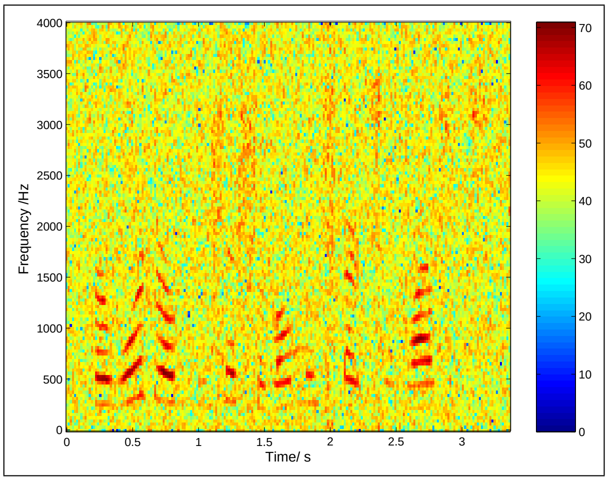

Figure 11: Spectrogram, Female 0 db SNR, Input.

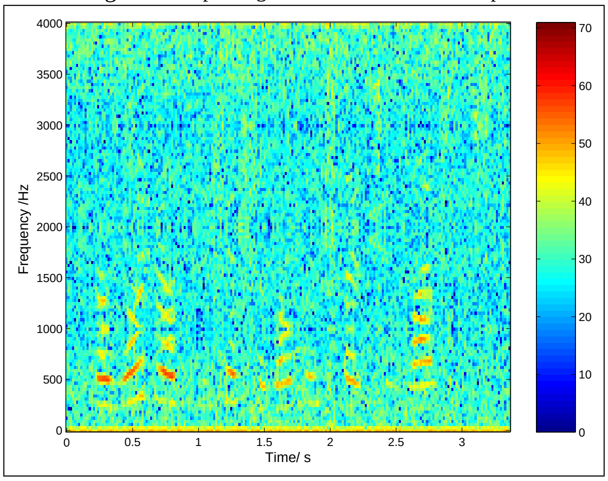

Figure 12: Spectrogram, Female 0 dB SNR, Output.

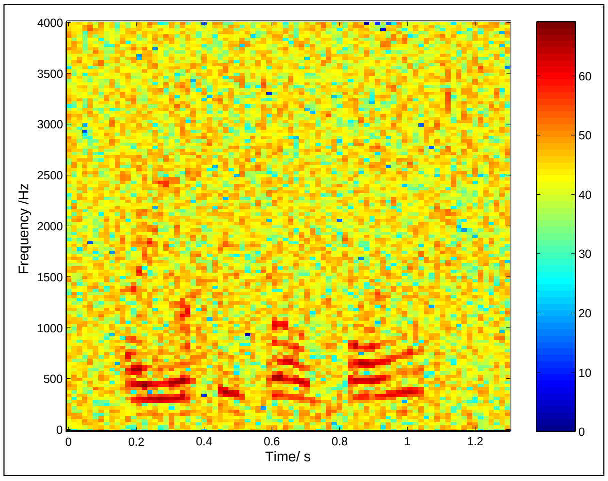

Figure 13: Spectrogram, Male 0 dB SNR, Input.

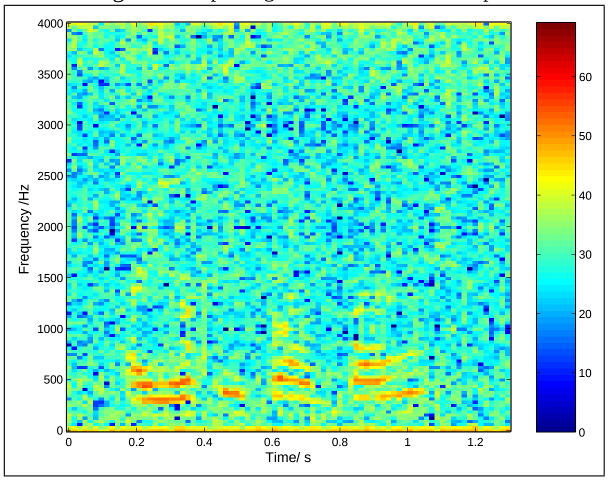

Figure 14: Spectrogram, Male 0 dB SNR, Output.

5. ACKNOWLEDGEMENTS

The authors would like to thank Prof. Egon Hilt of Deutsche Telekom University of Applied Sciences, Leipzig, Germany for his many useful sugestions regarding the implementation of the circuit.

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