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64 Bands Hybrid Noise Reduction and Feedback Cancellation Algorithm for Hearing Aid

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Abstract

Both chronic problems of hearing aid have been solved by a probabilistic signal processing algorithm. The algorithm was implemented into a low and single voltage and low power dsp ic chip which can be used as a hybrid of the hearing aid. Both simulated and experimented results show clear noise reduction as well as feedback cancellation. The computational capacity of the physical chip was practically discussed for the limit of the algorithm.

Keywords: *Hearing aid, Noise Reduction, Feedback Cancellation, Speech Enhancement*

1. Introduction

We are always surrounded with many environmental noises, for example, an air conditioner or a laundry machine at home, or the different sounds at a cocktail party, or a crowded market. For a person with normal hearing, the human ear is fine tuned to filter out the unwanted noises. But for a person who depends on a hearing aid device, his or her ear has lost that ability to filter out the unwanted sounds. Thus it is important to have a device that can filter out the unwanted sounds.

Digital hearing aids are getting more widely popular than analog hearing aids, as they promise many advantages over conventional analog hearing aids, among them the increased precision and programmability of DSP techniques and the possibility of adding new functions such as noise reduction, spectral sharpening and feedback cancellation. On the other hand their physical implementation is characterized by very tight requirements in chip size, voltage supply and power consumption. These are very difficult to fulfill due to the complexity and number of functions to be implemented together with the real time requirement and large dynamic range of the input signals [1]. Among such features noise reduction and speech enhancement is one of the most sought after features as it can greatly help in improving the usefulness of the hearing aid to the customer. If properly implemented it suppresses the background noise present in the environment while amplifying the desired speech signals, improving the hearing experience. Much study has been done in this field and a few methods have been proposed along with different variations and modifications by many researchers [2] and [3]. However most of these methods are computationally intensive and are not well suited for implementation in resource constrained environments. In this paper we propose an approach where we determine the input sample with minimum power in a predefined window of samples and use to calculate a smoothing parameter which helps us get the noise estimate. It is then used to calculate the SNR (Signal to Noise Ratio), which determines presence or absence of speech, which in turn determines the gain required. The output seems to have

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minimal distortions and appears clear enough for practical purposes. The platform used was the Ezairo 5920 DSP IC chip which is recently used as a hearing aid hybrid [4]. The advantage of this method in contrast to other popular ones is that the required program code can be implemented in real time and in less memory space.

1.1. Background

Let's begin with an observed noisy speech signal, x(n).

$$x(n) = s(n) + d(n). \tag{1}$$

where, s(n) and d(n) denote discrete time signals of clean speech and noise respectively. The observed signal x(n) is divided into overlapping frames by the application of a window function and analyzed using the Discrete Short Time Fourier Transform (DSTFT) given by

$$X(k,l) = \sum_{n=0}^{N-1} x(n+lM)h(n)e^{-j(\frac{2\pi}{N})kn} \tag{2}$$

where k is the frequency bin index, l is the time frame index, h is an analysis window of size N, and M is the framing step (number of samples separating two successive frames). Using the DSTFT analysis, (1) can be represented as

$$X(k,l) = S(k,l) + D(k,l). \tag{3}$$

1.2. Spectral Subtraction Method

The spectral subtraction method was originally proposed by Boll [5] and since then many variations of the process have been proposed [3]. In this method, firstly the DFT of the input noisy speech samples is calculated. The input signal spectrum is then subjected to subtractive noise suppression analysis. A noise suppressed spectral estimator is calculated by subtracting an estimate of the noise spectrum from the noisy speech signal's spectrum. The noise spectrum is obtained from the signal measured during the time when there is no presence of speech activity. If S(k) represents the speech spectrum, it's spectral subtraction estimate

$$\hat{S}(k) = [|X(k)| - \mu(k)]e^{-j\theta_x(k)}. \tag{4}$$

where the magnitude of the noise spectrum |D(k)| is replaced by μ(k) taken during non-speech activity, and the phase $\theta_D(k)$ of D(k) is replaced by the phase $\theta_x(k)$ of X(k). Since this is an estimator it does not accurately represent the speech signal. The spectral error resulting from this estimator is given by

$$\varepsilon(k) = \hat{S}(k) - S(k) = N(k) - \mu(k)e^{-j\theta_x(k)}. \tag{5}$$

The auditory effects of this spectral error are quite apparent and appear as distortions in the processed output.

2. Noise Reduction Algorithm for Hearing Aids

There are several physical considerations in the beginning. One is that an incoming noisy signal in digital format is the only source of process. The noisy signal means the original voice signal is mixed with an unknown and unwanted noise signal. Therefore we have to classify the voice from the noise by recursively analyzing the history of the sequential noisy data. The second assumption is that the intensity of the noisy signal has the behavior of Gaussian distribution independently in each frequency band, and the voice is not correlated with the noise. The third approach is that we apply short time (ST) fast fourier fransformation (FFT) and inverse FFT as a fundamental analysis tool in order to analyze the spectral characteristics of the noisy signal. From those hypotheses, we aim to estimate the optimal spectral gain, $G\left(\mathrm{k}, \ell\right)$, so that $\hat{X}\left(\mathrm{k}, \ell\right) = G\left(\mathrm{k}, \ell\right) Y\left(\mathrm{k}, \ell\right)$. k is the frequency bin index and ℓ is the time frame index. Y is the ST FFT of the noisy signal and $\hat{X}$ is the optimal spectral amplitude of the voice signal. If X is the original spectral amplitude of the voice signal, we aim to derive G by means of minimizing $E\left\{\left(\left|\hat{X}\left(\mathrm{k}, \ell\right)\right| - \left|X\left(\mathrm{k}, \ell\right)\right|\right)^2\right\}$. From the Gaussian distribution hypotheses, the conditional PDFs of the observed signal are given by [6]

$$pdf\left(Y\left(\mathrm{k}, \ell\right) \mid voice\ absence\right) = \frac{1}{\pi\sigma(k, \ell)^2} exp^{-\frac{\left|Y\left(\mathrm{k}, \ell\right)\right|^2}{\sigma(k, \ell)^2}}. \tag{6a}$$

$$pdf\left(Y\left(\mathrm{k}, \ell\right) \mid voice\ presence\right) = \frac{1}{\pi\left(\sigma_d(k, \ell)^2 + \sigma_x(k, \ell)^2\right)} exp^{-\frac{\left|Y\left(\mathrm{k}, \ell\right)\right|^2}{\left(\sigma_d(k, \ell)^2 + \sigma_x(k, \ell)^2\right)}}. \tag{6b}$$

where $\sigma_d(k, \ell)^2$ and $\sigma_x(k, \ell)^2$ are the variances of the voice absent signal and of the voice present signal respectively.

We leave the detailed mathematical derivation for other papers. The noise reduction algorithm is briefed as the following procedures. At first, the minimum of the local energy, $S_{min}\left(k, \ell\right)$, is tracked as follows:

$$S_{min}\left(k, \ell\right) = min\left\{S_{min}\left(k, \ell - 1\right), \left|Y\left(k, \ell\right)\right|^2\right\}. \tag{7}$$

$$S_{tmp}\left(k, \ell\right) = min\left\{S_{tmp}\left(k, \ell - 1\right), \left|Y\left(k, \ell\right)\right|^2\right\}. \tag{8}$$

If mod(k,L)=0, $S_{tmp}\left(k, \ell\right)$ was reinitialized.

$$S_{min}\left(k, \ell\right) = min\left\{S_{tmp}\left(k, \ell\right), \left|Y\left(k, \ell\right)\right|^2\right\}. \tag{9}$$

$$S_{tmp}\left(k,\ \ell\right)=\left|Y\left(k,\ \ell\right)\right|^{2}. \tag{10}$$

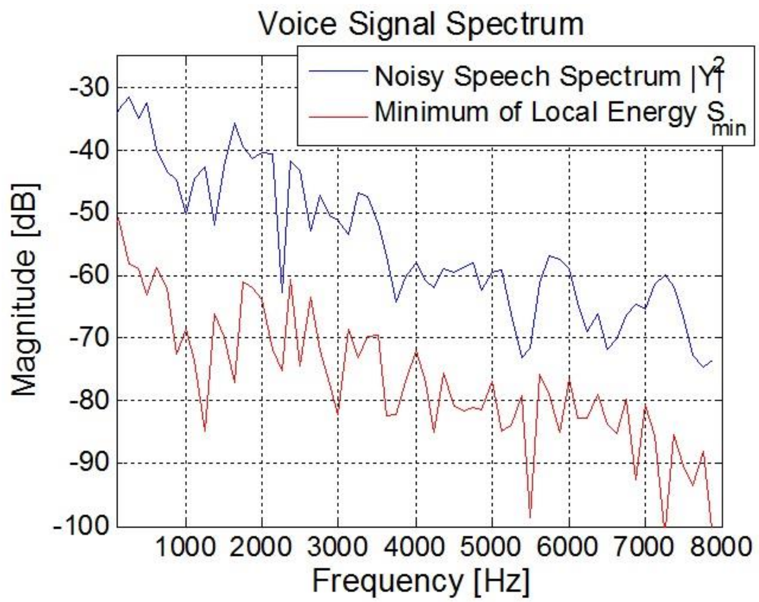

Figure 1. An example of the noisy speech spectrum (*upper blue line*) and the minimum of the local energy (*lower red line*)

Figure 1 shows an example of the noisy speech spectrum and its minimum of the local energy at any time frame. The noisy speech spectrum abruptly changes all the time, while the minimum of the local energy varies slowly. The noise spectrum is estimated based on the following procedure:

The ratio between $\left|Y\left(k,\ \ell\right)\right|^{2}$ and its derived minimum of the local energy is denoted as;

$$S_{r}\left(k,\ \ell\right)\triangleq\left|Y\left(k,\ \ell\right)\right|^{2}/\left|S_{min}\left(k,\ \ell\right)\right|\ . \tag{11}$$

Then, the conditional speech presence probability is estimated as [7];

$$\hat{p}'\left(k,\ \ell\right)=\alpha_{p}\hat{p}'\left(k,\ \ell-1\right)+(1-\alpha_{p})I\left(k,\ \ell\right). \tag{12}$$

where $I\left(k,\ \ell\right)$=1 $S_{r}\left(k,\ \ell\right)>\delta$ and $I\left(k,\ \ell\right)$=0 otherwise. α_{p} is 0.2 and δ is 5.

The noise spectrum $\hat{\lambda}_{d}\left(k,\ \ell\right)$, is recursively estimated while a smoothing parameter, $\tilde{\alpha}_{d}\left(k,\ \ell\right)$, is updated with the conditional speech presence probability, $\hat{p}'\left(k,\ \ell\right)$,

$$\tilde{\alpha}_d\left(k, \ell\right) \triangleq \alpha_d + (1-\alpha_d)\hat{p}'\left(k, \ell\right). \quad (13)$$

$$\hat{\lambda}_d\left(k, \ell\right) = \tilde{\alpha}_d\left(k, \ell\right)\hat{\lambda}_d\left(k, \ell - 1\right) + [1 - \tilde{\alpha}_d\left(k, \ell\right)]\left|Y\left(k, \ell\right)\right|^2. \quad (14)$$

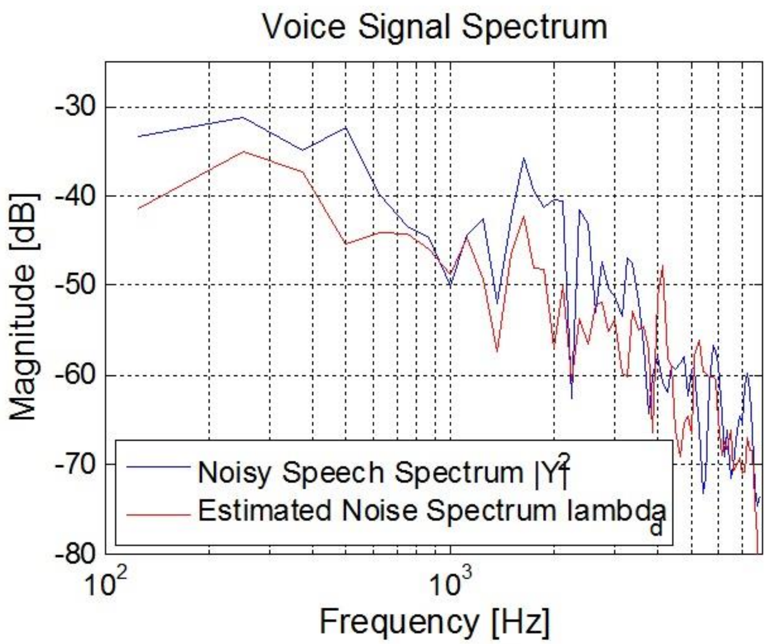

Figure 2. An example of the noisy speech spectrum (*upper blue line*) and the estimated noise spectrum (*lower red line*)

Figure 2 shows an example of the noisy speech spectrum and the estimated noise spectrum. If the noisy speech spectrum is higher than the estimated noise spectrum at a particular frequency band, we assume that the noisy speech spectrum should be amplified at that particular frequency band. However, if the noisy speech spectrum is lower than the estimated noise spectrum at another particular frequency band, the noisy speech spectrum should be suppressed at that particular frequency band. The estimated spectral gain, $G\left(k, \ell\right)$, is calculated as following procedures [7]:

$$\gamma\left(k, \ell\right) \triangleq \min\left[\frac{\left|Y\left(k, \ell\right)\right|^2}{\hat{\lambda}_d\left(k, \ell\right)}, 1000\right]. \quad (15)$$

$$\hat{\xi}\left(k,\ell\right)=\alpha\chi\left(k,\ell-1\right)+(1-\alpha)max[\gamma\left(k,\ell\right)-1,0]. \tag{16}$$

$$\hat{\xi}'\left(k,\ell\right)=\hat{\xi}\left(k,\ell\right)/\left(1+\hat{\xi}\left(k,\ell\right)\right). \tag{17}$$

$$\upsilon\left(k,\ell\right)\triangleq\gamma\left(k,\ell\right)\hat{\xi}'\left(k,\ell\right). \tag{18}$$

$$G\left(k,\ell\right)=\hat{\xi}'\left(k,\ell\right)\exp\left(\frac{1}{2}\int_{\upsilon\left(k,\ell\right)}^{\infty}\frac{e^{-t}}{t}dt\right). \tag{19}$$

$$\chi\left(k,\ell\right)=G^2\left(k,\ell\right)\gamma\left(k,\ell\right). \tag{20}$$

where α is 0.9899

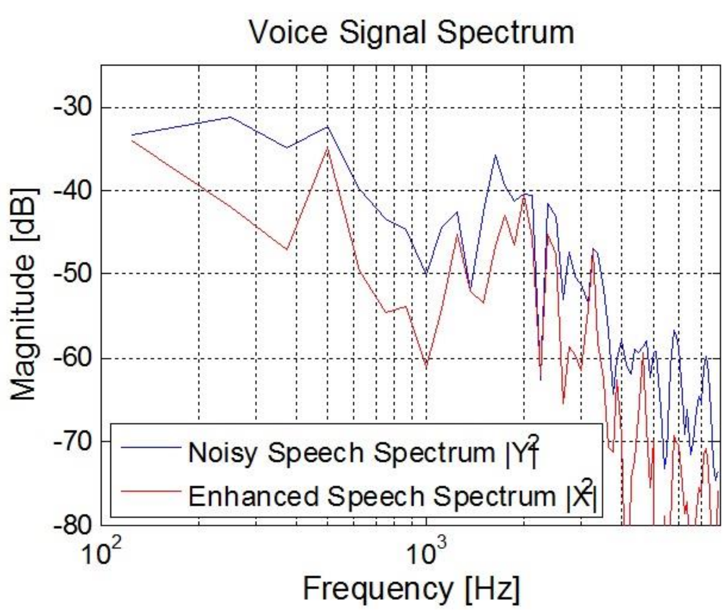

Figure 3. An example of the noisy speech spectrum (*upper blue line*) and the enhanced speech spectrum (*lower red line*)

Figure 3 shows an example of the noisy speech spectrum (upper blue line) and the enhanced speech spectrum $\hat{X}\left(\mathrm{k}, \ell\right)$ after multiplying $Y\left(\mathrm{k}, \ell\right)$ with $G\left(\mathrm{k}, \ell\right)$, that is, $\hat{X}\left(\mathrm{k}, \ell\right) = G\left(\mathrm{k}, \ell\right) Y\left(\mathrm{k}, \ell\right)$.

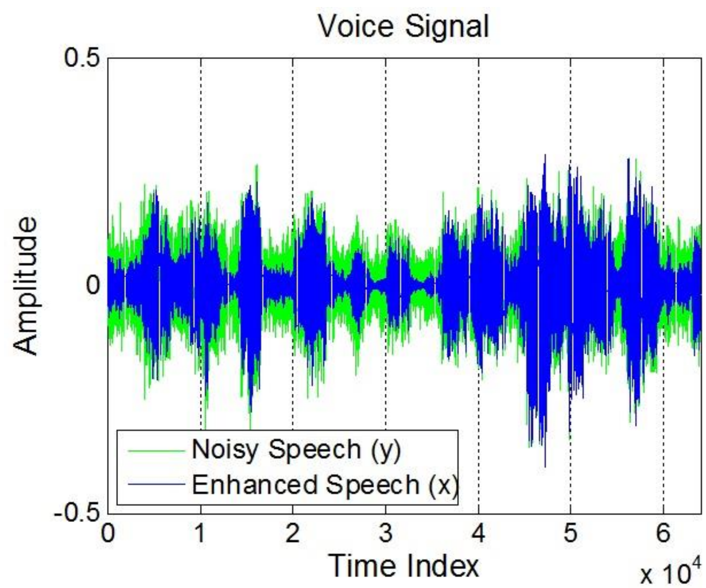

Figure 4. An example of the noisy speech data (*green line*) and the enhanced speech data (*blue line*)

3. Results

The sampling frequency was 16 kHz and we applied 4 seconds speech data mixed with environmental noise captured at a noisy café. L of equation 4 is 62. The number of a single temporal frame is 128 and 64 data are sequentially overlapped. Figure 4 shows an example of the noisy speech data and the enhanced speech data as a result of the present noise reduction method.

After the present noise reduction method was verified by matlab codes, the codes were translated to an assembler codes and were implemented into a digital hearing aid with an Ezairo 5920 hybrid. We applied a vacuum cleaner as a noise source. The noise is almost like a white noise. A few random words (“me” or “hello”) were spoken into the microphone, with the vacuum cleaner running in the background. The voice was picked up by the hearing aid microphone, processed by the Ezairo 5920 hybrid, and funneled out by the hearing aid receiver. A standard 2cc acoustic coupler transmitted the receiver output sound to a standard measuring amplifier and was the output displayed on a digital oscilloscope.

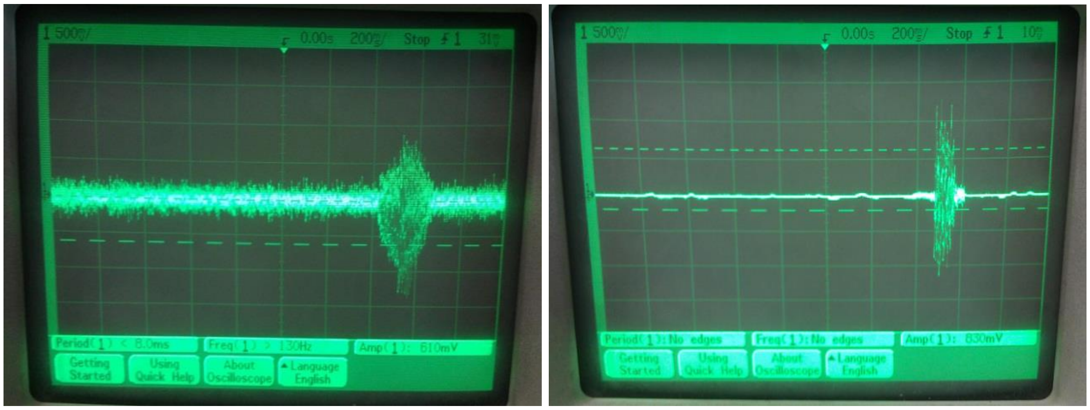

Figure 5. An example of the noisy speech data before noise reduction (*left*) and the enhanced speech data after noise reduction (*right*); One syllable "Me"

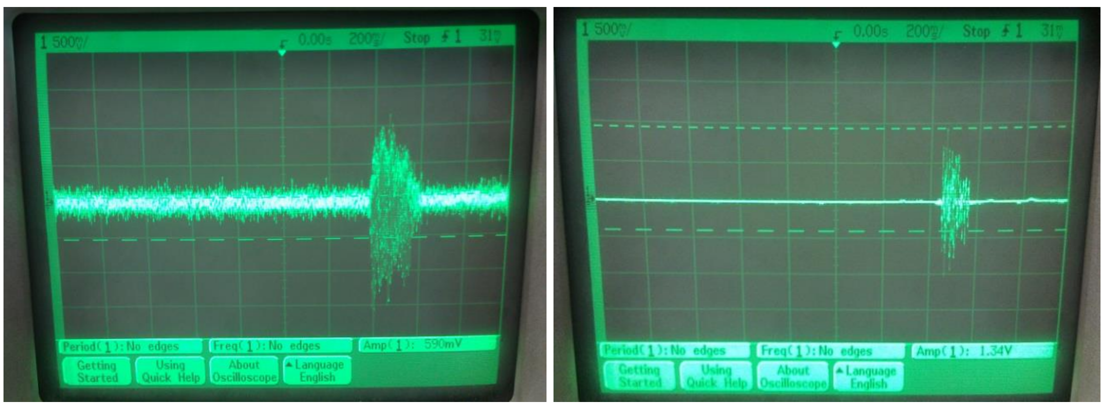

Figure 6. An example of the noisy speech data before noise reduction (*left*) and the enhanced speech data after noise reduction (*right*); Two syllables "Hello"

Figure 5 and figure 6 show examples of the noisy speech data before noise reduction (left) and the enhanced speech data after noise reduction (right). From the experimental results, it is clear that our noise reduction algorithm distinctly reduces unwanted environmental noise.

4. Conclusions

We will continue applying the noise reduction algorithm to different environmental noises while optimal parameters of related equations are researched. It should be noted that if the noise spectrum, $\hat{\lambda}_d(k, \ell)$, is estimated to be too steep, some musical noise is generated, on the other hand, if it is estimated to be too low, the input noise speech is distorted. Therefore optimal parameters should be used for a varying noisy atmosphere. Also noise reduction algorithm should be updated for better performance.

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