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Modified Adaptive Filtering Algorithm for Noise Cancellation in Speech Signals

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Introduction

Speech is a very basic way for humans to convey information to one another with a bandwidth of only 4 kHz; speech can convey information with the emotion of a human voice. The speech signal has certain properties: It is a one-dimensional signal, with time as its independent variable, it is random in nature, it is non-stationary, i.e. the frequency spectrum is not constant in time. Although human beings have an audible frequency range of 20Hz to 20 kHz, the human speech has significant frequency components only up to 4 kHz.

The most common problem in speech processing is the effect of interference noise in speech signals. Interference noise masks the speech signal and reduces its intelligibility. Interference noise can come from acoustical sources such as ventilation equipment, traffic, crowds and commonly, reverberation and echoes. It can also arise electronically from thermal noise, tape hiss or distortion products. If the sound system has unusually large peaks in its frequency response, the speech signal can even end up masking itself.

One relationship between the strength of the speech signal and the masking sound is called the signal-to-noise ratio, expressed in decibels. Ideally, the S/N ratio is greater than 0dB, indicating that the speech is louder than the noise. Just how much louder the speech needs to be in order to be understood varies with, among other things, the type and spectral content of the masking noise.

The most uniformly effective mask is broadband noise. Although, narrow-band noise is less effective at masking speech than broadband noise, the degree of masking varies with frequency.

High-frequency noise masks only the consonants, and its effectiveness as a mask decreases as the noise gets louder. But low-frequency noise is a much more effective mask when the noise is louder than the speech signal, and at high sound pressure levels it masks both vowels and consonants.

In general, noise [2] that affects the speech signals can be modeled using any one of the following:

1. White noise,
2. Colored noise,
3. Impulsive noise.

White noise

White noise is a sound or signal consisting of all audible frequencies with equal intensity. At each frequency, the phase of the noise spectrum is totally uncertain: It can be any value between 0 and 2π, and its value at any frequency is unrelated to the phase at any other frequency. When noise signals arising from two different sources add, the resultant noise signal has a power equal to the sum of the component powers. Because of the broad-band spectrum, white noise has strong masking capabilities.

Colored noise

Any noise that is not white can be termed as colored noise. Colored noise has frequency spectrum that is limited within a range unlike white noise which extends over the entire spectrum.

There are different types of colored noise (brown noise, pink noise, orange noise etc.) depending upon the gradation in the Power Spectral Density (PSD) of the noise. Colored noise can be generated by passing white noise through a filter with required frequency response.

Impulsive noise

Impulsive noise refers to sudden bursts of noise with relatively high amplitude. This type of noise causes click sounds in the signal of interest.

Impulsive noise is generally modeled as contaminated Gaussian noise, as indicated in equation (1).

$$\eta_0(n) = \eta_g(n) + b(n)\ \eta_w(n), \qquad (1)$$

where η_g (n) and η_w (n) are zero-mean Gaussian noises and b (n) is a Bernoulli random variable. The ratio of the variances of the two Gaussian noises decides the impulsive character of the noise generated.

Adaptive Filtering

Adaptive filtering can be considered as a process in which the parameters used for the processing of signals changes according to some criterion. Usually the criterion is the estimated mean squared error or the correlation [1]. The adaptive filters are time-varying since their parameters are continually changing in order to meet a performance requirement. In this sense, an adaptive filter can be interpreted as a filter that performs the approximation step on-line. Usually the definition of the performance criterion requires the existence of a reference signal that is usually hidden in the approximation step of fixed-filter design.

The general set up of adaptive filtering environment [2,6] is shown in Fig. 1, where k is the iteration number, x(k) denotes the input signal, y(k) is the adaptive filter output, and d(k) defines the desired signal. The error signal e(k) is calculated as d(k)-y(k). The error is then used to form a performance function or objective function that is required by the adaptation algorithm in order to determine the appropriate updating of the filter coefficients. The minimization of the objective function implies that the adaptive filter output signal is matching the desired signal in some sense.

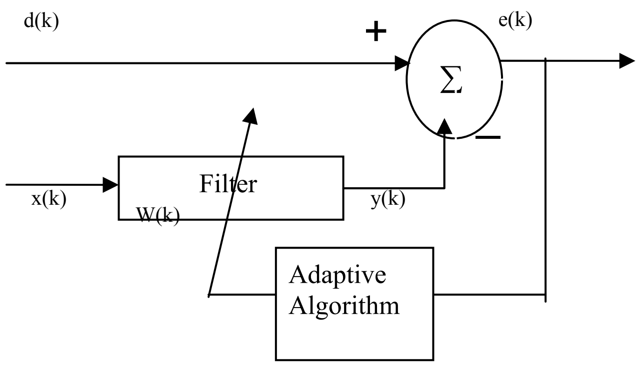

Fig. 1. General setup of adaptive filter

Adaptive Noise Canceller

Adaptive filter is widely used as noise canceller. In an adaptive noise canceller (Fig. 2) [3] two input signals, d(k) and x(k), are applied simultaneously to the adaptive filter. The signal d(k) is the contaminated signal containing both the desired signal, s(k), and the noise n(k), assumed uncorrelated with each other. The signal, x(k), is a measure of the contaminating signal which is correlated in sole way with n(k), x(k) is processed by the digital filter to produce an estimate y(k), of n(k). An estimate of the desired signal, e(k) is then obtained by subtracting the digital filter output, y(k), from the contaminated signal.

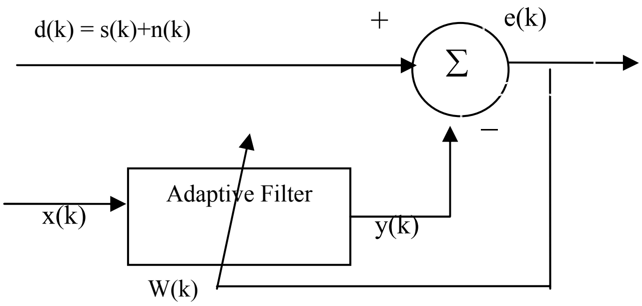

Fig. 2. Adaptive filter as a noise canceller

The filter output y(k) and the error e(k)[2] are given by equations (2) and (3).

$$y(k) = X^T(k)W(k), \qquad (2)$$

$$e(k) = d(k) - y(k). \qquad (3)$$

Adaptive AFA algorithm

In many applications of noise cancellation the changes in signal characteristics could be quite fast. This requires the utilization of adaptive algorithms, which converge rapidly. From this point of view the best choice is the recursive least squares (RLS) algorithm. Unfortunately this algorithm has high computational complexity and stability problems. The adaptive filtering with averaging (AFA) [4] is used for noise cancellation in above conditions. The algorithm [4] described above is summarized in equations (4)–(7):

$$y(k) = W^T(k-1)X(k), \qquad (4)$$

$$e(k) = d(k) - y(k), \qquad (5)$$

$$\overline{xe(k)} = \sum_{m=1}^{N} x(m-i)e(m). \qquad (6)$$

For filter of N-th order

$$W_i(k+1) = \overline{W_i(k)} + \frac{1}{k^\gamma}\overline{xe_i(k)}, \qquad (7)$$

where $0 < i < N$ and $1/2 < \gamma < 1$.

The averaging here does not create additional burden since the terms $\overline{W(k)}$ and $\overline{X(k)e(k)}$ can be recursively computed from their past values. Second, the algorithm does not use the covariance matrix, so there is no need of covariance estimate. This implies low computational complexity and escape from stability issues compared to the RLS [5] algorithm.

Proposed Modification in AFA algorithm

We propose a modification in the existing AFA algorithm to improve the performance in terms of signal to noise ratio. By introducing a variance factor in the AFA algorithm, the performance of the algorithm can be improved. The modified update equation is shown in equation (9):

$$W(k+1)=\overline{W(k)}+\frac{(\frac{1}{k^g}\overline{X(k)e(k)})}{2}+\frac{\mathrm{var}(\frac{1}{k^g}\overline{X(k)e(k)})}{2}. \quad (9)$$

The proposed modification was simulated and tested for noise cancellation in speech signal corrupted by white noise. The simulation results are shown in the following section.

Simulation and results

The parameters of clean speech sample considered for testing of the algorithms were: duration 2 seconds, PCM 22.050 kHz, 8 bit mono sample recorded under laboratory conditions.

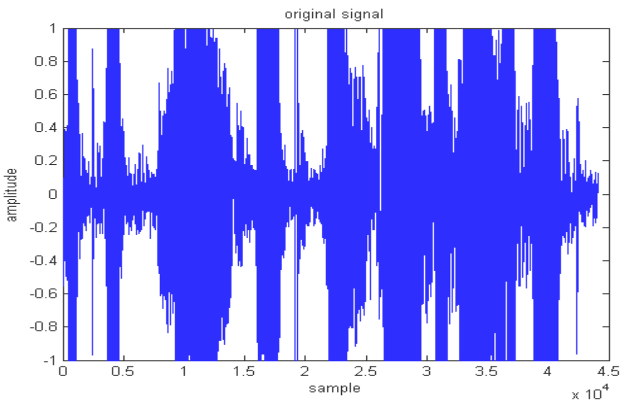

Fig. 3. Original speech signal

The filter order was fixed at 12 for all cases of noise cancellation in speech .The recorded sentence “A quick brown fox jumps over the lazy dog” was used as the clean speech. This sentence is conventionally used as a benchmark for speech processing. The above sentence contains all the alphabets of the English language. Hence the variability of effect of noise on speech with frequency of the signal is accounted. The original speech signal is shown in Fig. 3.

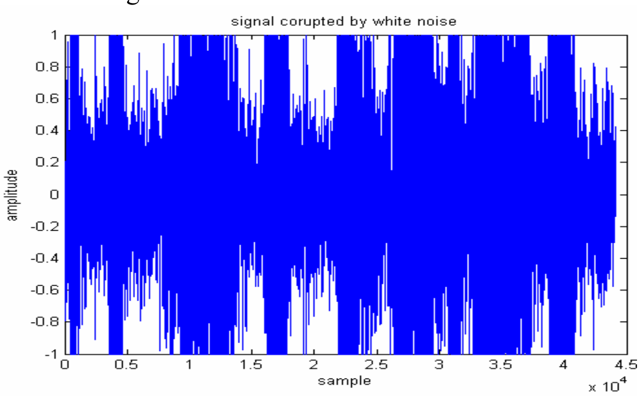

Fig. 4. Speech signal corrupted by white noise

White Gaussian noise was generated and added to the original speech signal. The SNR of the signal corrupted with noise was 7.776 dB. A linear combination of the generated noise and the original signal is used as the primary input for the filter. Fig. 4 shows the original speech corrupted by white noise

The denoised speech signal using the original AFA algorithm is shown in Fig. 5. The output Signal to noise ratio of the signal denoised with original AFA algorithm was 28.5101 dB.

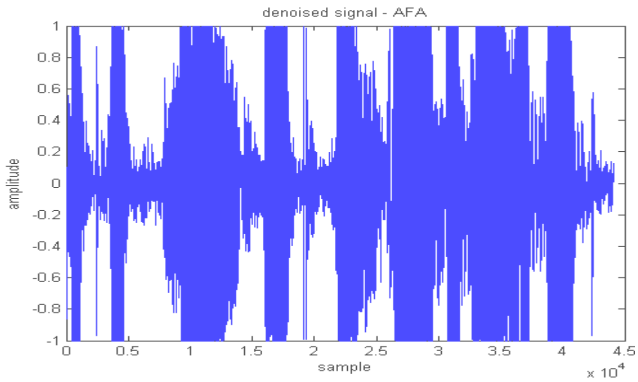

Fig. 5. Denoised signal using original AFA algorithm

Fig. 6 shows the signal denoised with proposed modification included in the AFA algorithm. The figure shows the output of modified AFA algorithm. The output signal to noise ratio of the signal was 30.4568 dB as compared to 28.5101 dB of the original AFA algorithm.

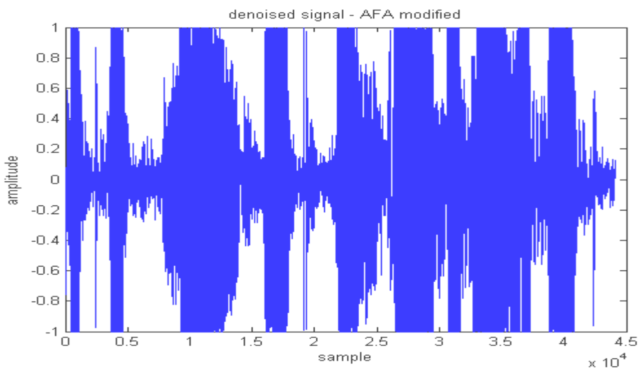

Fig. 6. Denoised signal using modified AFA algorithm

Table 1 shows the output signal to noise ratio of the denoised speech signal using the original and modified AFA algorithms.

Table 1. Output SNR of denoised signal

Algorithm	Output SNR in dB
AFA	28.5101
Modified AFA	30.4568

Conclusions

The proposed modification in the existing adaptive filtering with averaging algorithm was simulated and tested for noise cancellation in speech signals. When tested with white noise, the modified AFA algorithm showed an improved performance of around 2 dB compared to the original AFA algorithm. For future work we planned to test this modified algorithm for colored and impulsive noise. The study and comparison of adaptive algorithms can be extended to the use of multi-dimensional adaptive filtering techniques; for applications like noise cancellation in images. Further, the modified algorithms proposed can be optimized to have lower complexity.

References

1. **Widrow Bernard, Samuel Stearns D.** Adaptive Signal Processing. – Pearson Education, Delhi, 2004.
2. **Monson Hayes H.** Statistical Digital Signal Processing and Modelling. – John Wiley & Sons Inc, Kundli, 2002.
3. **Emmanuel Ifeachor C., Jervis Barrie W.** Digital Signal Processing – A practical approach. – Pearson Education, Delhi, 2004.
4. **Georgi Iliev, Nikola Kasabov.** Adaptive filtering with averaging in noise cancellation for voice and speech recognition. – Department of Information Science, University of Otago, 2001.
5. **Diniz P.** Adaptive Filtering Algorithms and Implementation Issues. – Kluwer Academic Publishers, USA, 2002.
6. **Xiao Hu, Ai-qun Hu, Li Zhao.** A Robust Adaptive Speech Enhancement System // IEEE Int. Conference on Neural Networks and Signal processing. – Nanjing, China, December 14-17, 2003.
7. **Ikeda S., Sugiyama A.** An adaptive noise canceller with low signal distortion for speech codecs // IEEE Trans. Signal Processing. – Vol. 47, Mar. 1999. – P. 665–674.
8. **Julie E. Greenberg.** Modified LMS Algorithms for Speech Processing with an Adaptive Noise Canceller // IEEE Trans. On Speech and Audio Processing. – Vol. 6, No. 4, July 1998.

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Adaptive filtering techniques are one of the important techniques used for noise cancellation in speech and biomedical signals. The Least Mean Squares (LMS) algorithm is one of the widely used algorithms in many adaptive signal processing environments. The adaptive filtering algorithm with averaging (AFA) algorithm is an improvement over the widely used Least Mean Squares (LMS) algorithm and has an improved performance. In this paper, we propose a modification in the AFA algorithm with improved performance for speech signal processing. The proposed modification was implemented in Matlab and was tested for noise cancellation in speech signals. The simulation results showed that modification has improved performance in terms of signal-to-noise ratio compared to the original adaptive filtering algorithm. Ill. 6, bibl. 8 (in English; summaries in English, Russian and Lithuanian).

В. Р. Вияйкумар, П. Т. Ванати, П. Канагасабапати. Модифицированный адаптивный алгоритм фильтрования для удаления шума в сигналах речи // Электроника и электротехника. – Каунас: Технология, 2007. – № 2(74). – С. 17–20.

Методы адаптивной фильтрации являются одними из самых популярных при устранении шумов из речи и биомедицинских сигналов. Алгоритм наименьших квадратов широко используется во многих сферах обработки адаптивного сигнала. Алгоритм фильтрации адаптивного сигнала с применением усреднения (AFA) является более преимущественным по сравнению с методом наименьших квадратов. Предложена модификация (AFA) алгоритма, позволяющая более эффективно обработать речевые сигналы. Модификация внедрена применяя программу Matlab и тестирована устраняя шумы из речевых сигналов. Результаты моделирования показали, что при модификации алгоритма его эффективность по сравнению с оригинальным адаптивным алгоритмом фильтрации в соотношении сигнал-шум, улучшилась. Ил. 6 , библ. 8 (на английском языке; рефераты на английском, русском и литовском яз.).

V. R. Vijaykumar, P. T. Vanathi, P. Kanagasabapathy. Modifikuotas adaptyvus filtravimo algoritmas triukšmams iš kalbos signalų pašalinti // Elektronika ir elektrotechnika. – Kaunas: Technologija, 2007. – Nr. 2(74). – P. 17–20.

Adaptyvaus filtravimo metodai yra vieni iš dažniausiai taikomų triukšmui iš kalbos ir biomedicininių signalų šalinti. Mažiausių kvadratų algoritmas yra plačiai naudojamas daugelyje adaptyvaus signalo apdorojimo sričių. Adaptyvaus signalo filtravimo, naudojant vidurkinimą, algoritmas (AFA) yra pranašesnis, palyginti su mažiausių kvadratų metodu. Siūloma AFA algoritmo modifikacija, leidžianti efektyviau apdoroti kalbos signalą. Pasiūlytoji modifikacija įgyvendinta naudojant Matlab programą ir testuota šalinant triukšmą iš kalbos signalų. Modeliavimo rezultatai parodė, jog modifikuoto algoritmo našumas, lyginant su originaliu adaptyviu filtravimo algoritmu pagal signalo ir triukšmo santykį, pagerėjo. Il. 6, bibl. 8 (anglų kalba; santraukos anglų, rusų ir lietuvių k.).