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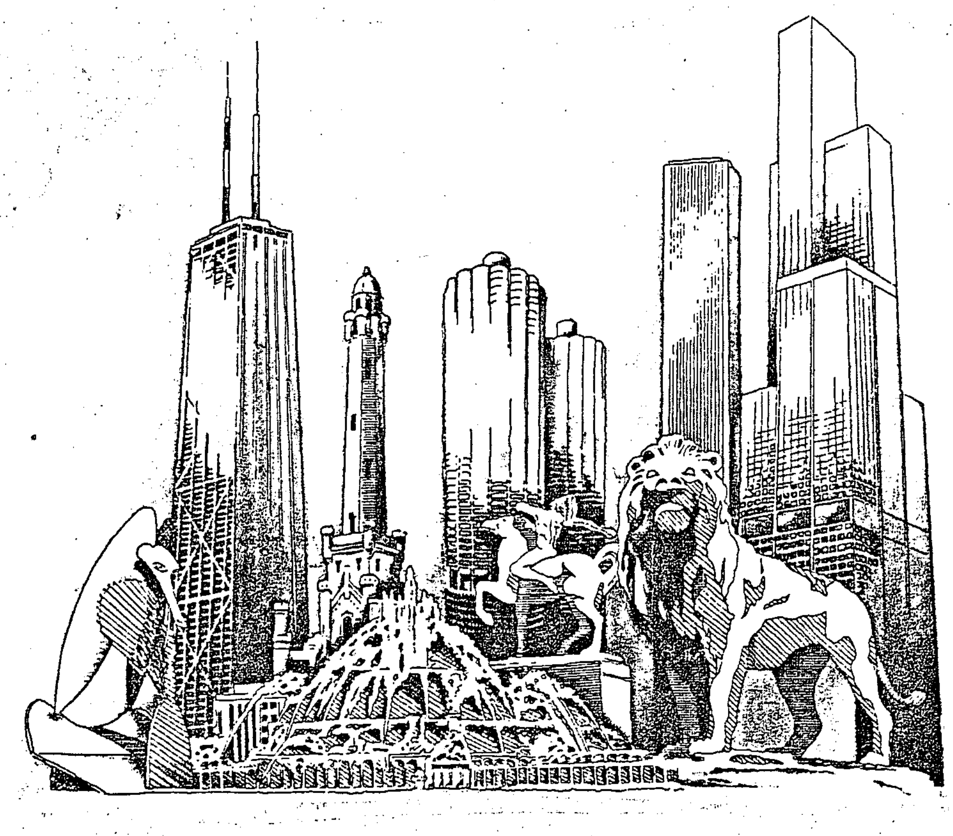

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Noise Reduction in Speech Using a Modified LMS Adaptive Predictive Filter

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Abstract

A modified version of a "leaky weight" LMS adaptive filter is applied to the problem of noise suppression with speech present. It is shown that noise may be suppressed while preserving speech components essential to intelligibility by appropriately choosing algorithm parameters.

Introduction

It is well known that speech intelligibility is greatly reduced in the presence of noise for the hearing impaired who use conventional hearing aids. This has given rise to numerous papers over the years which have utilized various signal processing techniques to separate speech and noise [1], [2], [3], [4], [5]. A two microphone noise canceler using the Widrow-Hoff LMS digital adaptive filter has been demonstrated to be very successful in significantly restoring intelligibility in the presence of both speech babble and broadband noise at SNR levels as low as -12 dB. before processing[6], [7]. However, a wearable two microphone hearing aid for noise suppression introduces additional problems such as microphone placement, hardware miniaturization or RF/hardwired link requirements, separation between microphone and transducer for feedback elimination, and imposes consideration of directional microphone properties. These problems are research and development topics by themselves and add additional cost, complexity and annoyance factors for the hearing impaired. There is thus strong motivation to perform single microphone cancellation.

In 1978 Sambur [1] proposed to apply the an LMS version of the digital adaptive filter using reference delays equal to one or two voice pitch periods necessitating a pitch period estimator. Sambur reasoned that in the speech component of the corrupted signal there would be strong correlation between the primary and delayed reference inputs. He reported improved SNR and speech quality but did not claim improved intelligibility. In fact, the time domain LMS adaptive filter concentrates its computational power first on those frequencies in the signal with the highest energy, (i.e. pitch and pitch harmonics where little intelligibility is carried) and last on frequencies with the least energy (i.e. high frequency sounds where most of the information in speech is carried). This results in an output which sounds like muffled speech.

The muffling effect appears to be present in most speech enhancement systems prompting the statements by Lim [5] and Schafer [8] that the various speech enhancement systems appear to improve the subjective speech quality but not speech intelligibility and that successful approaches must exploit more knowledge about the information bearing elements of speech.

Time Domain Filters for Noise Suppression

Time domain filters for noise suppression are inherently attractive because of their simplicity. Further, the computational savings of fast frequency-domain implementations only becomes evident at modest to large filter sizes, typically for filter lengths in excess of 32 weights. Sambur [1] proposed a time domain implementation of the adaptive filter shown in Fig. 1

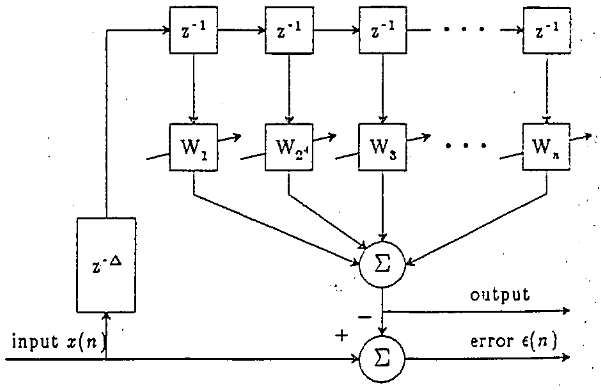

Figure 1. Time domain LMS adaptive filter

which employs the LMS Widrow-Hoff algorithm in Eq. 1 for the weight updates.

$$W_p(n+1) = W_p(n) + 2\mu\epsilon(n)x(n-p) \tag{1}$$

Implementation of this LMS adaptive filter results in three deficiencies:

(1) The speech spectrum is distorted, with the low frequency region enhanced due to the high energy content at the low frequencies.

(2) Unvoiced speech sounds are eliminated in the signal processing by delays which are in the neighborhood of a pitch period. This results in confusion between such words as net, nets, next, etc. leading to reduced intelligibility.

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(3) Reverberation is introduced because the LMS algorithm in Eq. 1 responds to minimize mean square error and will leave large values for W_p when $x(n-p)$ becomes small or is zero - as is the case during the silent portions of speech. The sound introduced is reminiscent of listening to a sea-shell and hearing that reverberant background.

There are several approaches to reducing these deleterious effects introduced by time domain adaptive processing. By forcing increased processor attention to high frequencies using the LMS adaptive filter, the spectral distortion may be made acceptable. Another problem is introduced due to the change from speech to silence between words. The time domain adaptive processor stops updating the filter weights when the input signal power decreases significantly (during speech silence) so the filter "remembers" the weights from the previous speech. When the next word begins an annoying echo or synthetic reverberation effect is produced. This effect was the primary motivation for the work reported in this paper and may be overcome using a modified form of the LMS algorithm..

While solutions to the problem of suppressing noise are available for either frequency domain or time domain versions of the adaptive filter, this paper deals only with the time domain versions. A general frequency domain adaptive filter is treated by Chabries, et. al [9].

PREWHITENING EFFECT

The general behavior of the adaptive LMS filter in processing both high frequency - low energy and low frequency - high energy portions of the speech spectrum is illustrated in Fig. 2. Here in Fig. 2b it is clear that the high frequency information bearing elements of the speech sample have been removed by the adaptive processing. In the time domain representation of the LMS adaptive filter a single zero pre-whitening filter was introduced prior to the input to the adaptive filter. The transfer function for this filter is given in Eq. 2

$$H(z) = 1 + a\,z^{-1} \tag{2}$$

where the value for a was chosen to place the corner frequency at 100Hz. The effects of prewhitening in preserving the high frequencies is shown in Fig. 3c.

PRESERVATION OF UNVOICED SPEECH

Previous attempts to enhance speech have given little attention to speech quality issues arising from the unvoiced portions of speech. In both the previous architecture, shown in Fig. 1, and in the proposed architecture, shown in Fig. 4, the amount of delay allowed for decorrelation in speech must be minimized. In the work reported, typical values for delay were on the order of 3 to 5 samples with a sample rate of 14kHz. This limited the amount of noise suppression, but preserved essential portions of the speech.

LEAKY WEIGHT ALGORITHM

From examination of Eq. 1, it is clear that the weights, W_p, will not be updated nor decay when the energy in the input signal, $x(n-p)$, is zero or very low. This lack of decay in the weights is responsible for the artificial reverberation effects in the processed speech using the conventional time-domain LMS adaptive filter algorithm. A solution to the decay problem was

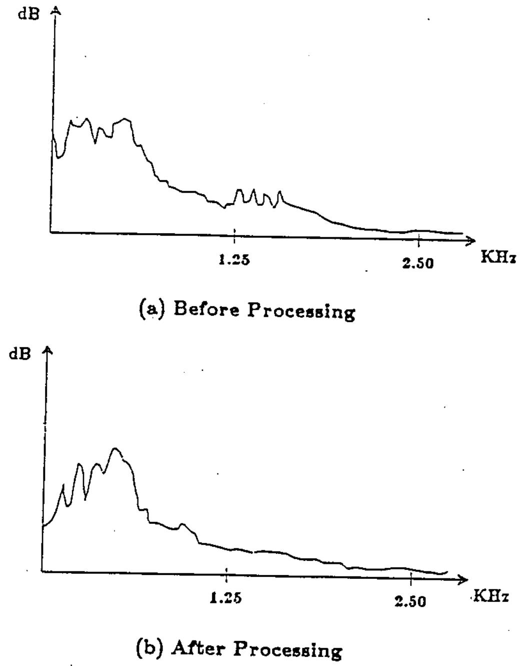

Figure 2. Noiseless speech before and after processing by an adaptive LMS filter.

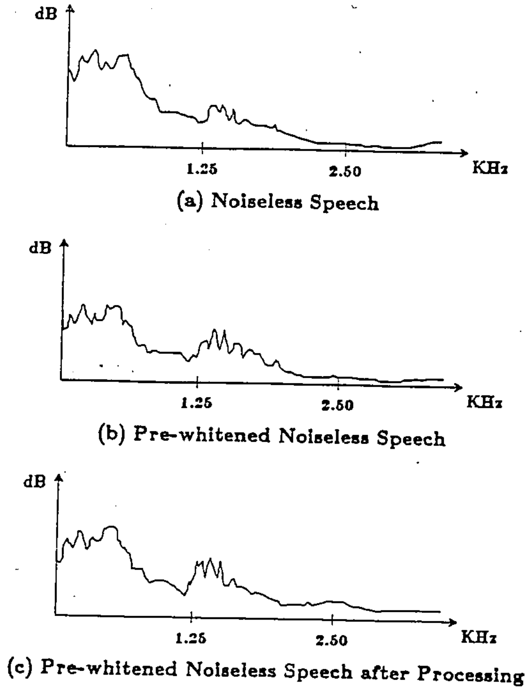

Figure 3. Pre-whitened noiseless speech before and after processing by an adaptive LMS filter.

proposed by Gitlin [10] for an adaptive equalizer application on telephone channels. This solution referred to as the "Leaky Weight Algorithm," or LW-LMS, has been implemented by the authors in the time-domain LMS adaptive filter version of the adaptive filter. This is shown in the modified weight update equation

$$W_p(n+1) = W_p(n)(1-\rho) + 2\mu\epsilon(n)x(n-p) \qquad (3)$$

Where $(1-\rho)$ is the leak factor. The leak rate is proportional to ρ and may be adjusted for listener preference. Once the leak time constant is determined the value of ρ can be determined as given in equation 4.

$$\rho = 1 - e^{\frac{-1}{\tau_L f_s}} \qquad (4)$$

The LW-LMS algorithm finally implemented is actually a modification of the version proposed by Gitlin. This new structure is shown in Fig. 4. This version uses interpolation rather than extrapolation by employing half the weights with output from the tapped delay line to predict *Delta* samples into the future and the other half to estimate *Delta* samples into the past. After subjective

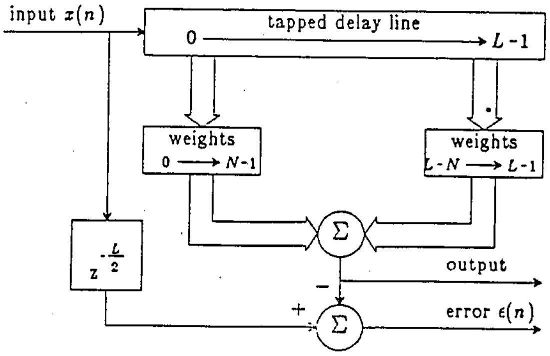

Figure 4. Split filter LW-LMS configuration.

listening experiments following processing, the values determined for μ and ρ were 1.5×10^{-3} and 3.33×10^{-3} respectively. The tapped delay line length was selected to be 32 and the time delay, *Delta*, was set to 15 for a sample frequency of 75kHz. This high sample frequency allowed a small μ to be chosen to minimize algorithm noise while preserving rapid adaptation - thus pointing out the inefficiency of the time-domain algorithms for this purpose.

CONCLUSIONS

The LW-LMS algorithm, coupled with a pre-whitening of the input signal, was found to be very effective in suppressing the noise component of a +6 dB SNR speech-noise signal. Gains of greater than 10 dB. noise suppression were obtained. A tape of these processing results was prepared for CID word list 22. Deleterious effects on the speech component do occur, but they can be controlled by adjustment of the filter parameters. While the filter, as described herein, is not efficient, it shows great potential for further refinement and hence widespread application to the problem of noise suppression in speech signals. In particular, the leak factor in the LW-LMS adaptive filter weight adjustment algorithm is successful in causing the filter output to converge to zero during the silent periods of speech or to converge to zero in the presence of noise without speech. This eliminates a great deal of filter noise that would otherwise be present. Also the pre-whitening of the input signal preserves the high frequency components of the speech spectrum that would otherwise be severely attenuated.

Further research is needed in order to find methods for improving the efficiency of this algorithm. In addition, it is believed that the filter's performance may be improved by making the leak factor inversely proportional to the power in the filter's output signal. This would reduce the amount of leakage during adaptation with speech present, and increase the leakage when no speech is present. Finally, it is felt that the speech quality would be improved if an optimal process were employed in the input signal pre-whitening. Such an optimal pre-whitener should only boost those portions of the input spectrum containing higher frequency voice components.

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