"An Experimentally Optimized Real-Time Adaptive Noise Canceler/Self-Tuning Filter Implemented with a TMS320C30 DSP"

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Abstract.

A Texas Instruments TMS320C30 floating point digital signal processor has been used to implement a noise canceler/self-tuning filter for extracting sinusoidal audio signals from broad-band noise. A very significant part of the work is the optimization of the tuner's performance through proper choice of relevant adaptive filter parameters as well as efficient TMS320C30 code design and implementation.

The paper describes results of work performed to assess: a) the improvement in Signal-to-Noise Ratio(SNR) from input to output versus input SNR; b) the speed of adaptation(or convergence of adaptive filter weights) and the steady state misadjustment noise, versus gain parameter, k, and; c)the effects on stability of convergence, by the choice of delay parameter, delta, and input frequency f. The following results were obtained.

1. Improvement in SNR between output and input of 13db. was measured.

2. Data clearly showed that steady state misadjustment error was minimized by choosing a smaller gain parameter, k. For example, a k value of 0.001 produced a fifty percent reduction in misadjustment noise over that for a k=0.01

3. Delay parameter, delta values of from 1 to 61 were found to yield highly stable filters.

Sinusoidal frequencies from 100 Hz to 3.6 Khz were used as input sinusoids. An 81-tap adaptive FIR was used. The heart of the equipment used for the experiment was the Texas Instruments TMS320C30 evaluation module(EVM). An LMS algorithm was implemented in real-time using the TMS320C30's assembly language. The input signal was constructed by summing a sinusoidal from a Wavetek 3260 function generator. Random noise noise source was provided by a Hewlett-Packard 3562A Dynamic Signal Analyzer at an op-amp summing junction.

Introduction.

The first major work on adaptive noise canceling and its applications was done by Widrow et al[1]. A particularly interesting adaptive noise canceler proposed by Widrow was the self-tuning filter or the adaptive line enhancer[ALE]. The ALE has been analyzed by researchers using various modified forms of the LMS[Least Mean Square] algorithm. The LMS algorithm represents a good compromise for convergence speed, steady state misadjustment error, stability and complexity that are required simultaneously for real-time implementations[2]. It therefore continues to attract a great deal of research interest[3,4]. However, theoretical analysis and simulation of ALE performance using LMS algorithms have identified many complex issues that are far from resolved. For example:

1. How to choose the adaptation step size k, for a stable LMS algorithm, rapid convergence, and small steady state misadjustment error.

2. What is the optimum value for the decorrelation parameter , delta, for ensuring best ALE performance in extraction of sinusoidal signals.

3. Most importantly, what kind of noise concelation performance can be realized when the algorithms are actually implemented on a modern high speed floating point digital signal processor such as the Texas Instruments' TMS320C30 digital signal processor.

Experimental Procedure.

A real-time adaptive self-tuning filter was implemented using the TMS320C30. It experimentally investigated how the performance of the adaptive filter(the fidelity of the tuning mechanism) was affected by the delay parameter, delta, in the path of the reference signal, and k, the adaptation step size which controls the stability and convergence rate. The role played by the number of taps of the filter was also investigated.

The sinusoidal portion of the "signal+ white noise", representing the input to the filter was set to different voice-band frequencies. The frequency response measurements of the corresponding adaptively tuned filter was made with a signal analyzer. The power spectra of the input was compared with that of the output for all tested frequencies. Convergence speed and steady state misadjustment error were measured.

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Results.

The real-time power spectral response of the filter output for input signal frequencies of 515Hz, 1.2KHz, and 2 KHz was measured. For each frequency, a narrow band tuned filter resulted and the output of the filter showed the sinusoid to be enhanced over the noise by about 12 db.

Dependence of Filter Performance on Tap Length, Delay, and Step(Gain) Size.

The filter stability and tuning performance was measured as a function of number of taps, delay parameter, and step size(or gain). The following conclusions were made:

1. The adaptive filter's effectiveness does not strongly depend on the filter tap length.

2. A broad range of delay values is allowed. However, larger values of delay led to more stable and better tuned characteristics. Delay values from 1 to 41 sample times were utilized.

3. The stability and tuning effectiveness were found to be most strongly dependent on the gain parameter(step size). A step size of 0.01 did not yield a stable filter for any of the combinations of tap length and delays studied. Values of gain of 0.005 and smaller, namely 0.001 and 0.005, produced stable filters.

Signal to Noise Ratio(SNR) and Misadjustment Noise.

The following results were obtained.

1. Above a threshold of input SNR of 22db., an improvement of SNR between output and input of 13db. was measured. Below this threshold, improvement in SNR falls rapidly with decreasing input SNR.

2. Data clearly showed that a misadjustment error was minimized by choosing smaller gain parameter (or step size). For example, a step value of 0.001 produced a fifty percent reduction in misadjustment noise over that for a step size of 0.01. On the other hand, a gain step of 0.001 resulted in a factor of 8 reduction in speed of adaptation over that for a step of 0.01..

References.

1. Widrow, et al., "Adaptive Noise Canceling: Principles and Applications", Proceedings of IEEE, Vol. 63, pp. 1692-1716, Dec. 1975.
2. Sugiyama, A., "An Interference-Robust Stochastic Gradient Algorithm with a Gradient-Adaptive Step Size", Proceedings of ICASSP 1993, pp.III-539-542.
3. Douglas, S.C., "Exact Expectation Analysis of the LMS Adaptive Filter for Correlated Gaussian Input Data", Proceedings of ICASSP 1993, pp. III 519-522.

4. Freire, N.L., and Douglas, S.C., “Adaptive Cancellation of Geomagnetic Background Noise Using a Sign-Error Normalized LMS Algorithm”, Proceedings of ICASSP 1993, pp. III 523-1428.