Adaptive Cancellation of Harmonic Interferences in Transcranial Doppler Signal

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ABSTRACT

This paper presents a method of improving the Transcranial Doppler (TCD) signal by removing harmonic interferences. Such interferences, originating from medical equipment using the high power HF signals are common in a clinical environment, especially in the neighborhood of the operating theater. The Adaptive Interference Canceler based on the NLMS FIR filter has been used. The reference signal was obtained by delaying of the original TCD signal.
The presented method allows significant improvement of a seriously disturbed TCD signal.

Keywords: Transcranial Doppler, medical diagnostics, ultrasound, adaptive interference cancellation, adaptive filters, LMS algorithm

1. INTRODUCTION

The Transcranial Doppler Sonograph (TCD) is a device allowing non-invasive measurement of blood flow velocity in the intracranial vessels. It is proven to be very useful tool particularly in the neurosurgical diagnostics.[1]

One of the factors deteriorating the quality of the TCD signal are interferences originating from the high power HF medical equipment (eg. diathermy knives). Usually these interferences, are harmonic signals with relatively stable frequency, and are visible as horizontal lines on the spectrogram of the TCD signal. The position of these lines depends on the base frequency of the interfering signal and on its shape (because of aliasing, the consecutive harmonics may be irregularly distributed over the frequency range). An example spectrogram of the highly disturbed TCD signal is shown in the Fig. 1.

Such interferences are easy to recognize by a human operating the TCD and even though annoying, they can be considered when "manually" analyzing the signal's spectrum. However the contemporary TCD devices usually perform automatic analysis of the TCD signal spectrum to estimate (among others) the intermittent maximum frequency f_{max} (also known as "spectrum envelope"- proportional to the intermittent maximum blood flow velocity) and the intermittent mean frequency f_{mean} (proportional to the intermittent mean blood flow velocity). Time trends of these quantities are then used to calculate other clinically significant parameters, eg. the "Pulsatility Index" (PI)[1] and the "Resistive Index" (RI) .

Unfortunately, the described above interferences corrupt calculated f_{max} and f_{mean} trends rendering them unusable. An example of the f_{max} trend (estimated with the neural network based algorithm[2]) from the real disturbed TCD signal is shown in the Fig. 2.

These interferences may be limited by the proper design and shielding of the TCD device. However the efficient DSP algorithms for their elimination can be useful for measurements in particularly difficult conditions.

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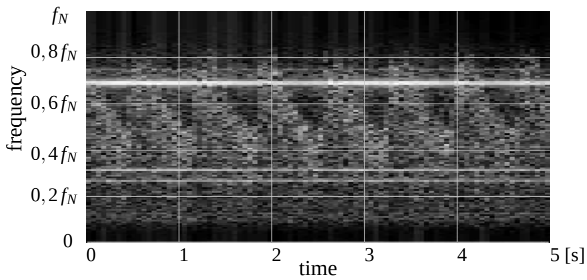

Figure 1. An example spectrogram of the real TCD signal with harmonic interferences

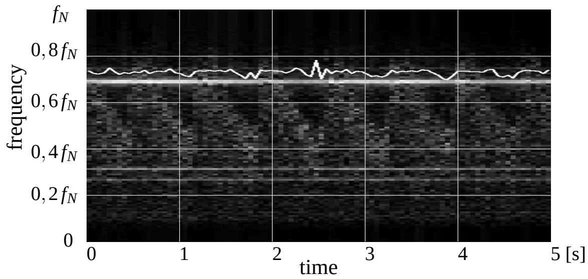

Figure 2. Results of f_{max} trend calculation in the real disturbed TCD signal (white line - f_{max}, background - the spectrogram)

2. MATERIAL

In this study three types of the TCD signal have been used:

- The real disturbed TCD signals recorded digitally
- The real TCD signals recorded digitally with artificially added interferences
- The simulated TCD signals with known parameters (generated with the model described in Ref. 3 and in Ref. 4) with added interferences

The proposed method has been tested with all the above types of the signal.

3. CHARACTERISTICS OF THE INTERFERENCES AND OF THE TCD SIGNAL

The TCD device uses the short pulses of ultrasound wave to insonate the cerebral vessel. The ultrasound wave is scattered on the moving blood cells, and the frequency of the scattered wave is changed due to the Doppler effect.

The returning ultrasound signal is received during the short time window (to obtain only signal returning from the particular depth). The received signal is then (after some additional processing) subjected to the quadrature detection providing two components - I (in-phase) and Q (quadrature) which can be considered as a real and imaginary part of the complex TCD signal.

The power density spectrum of the TCD signal corresponds to the blood flow velocity profile in the insonated vessel. Strictly, for the power density spectra estimated from the short, quasistationary blocks of the TCD signal the following relationship is true: The power of signal in a particular frequency range is the random variable with the χ^2 distribution and the expected value proportional to the amount of blood cells moving with radial velocities for which the Doppler shift encloses in that frequency range.

The TCD signal is a stochastic band limited noise, and its autocorrelation function quickly tends to zero for higher delays (the autocorrelation function of the sample TCD signal is shown in the Fig. 3). The more detailed analysis of the TCD signal can be found in Ref. 5, Ref. 6 and Ref. 4.

On the contrary the harmonic interference is a periodical deterministic signal with periodical autocorrelation function which reaches nonzero values even for high delays (Fig. 4). This difference in autocorrelation function properties between the TCD signal and the harmonic interference allows us to build the Adaptive Interference Canceler.

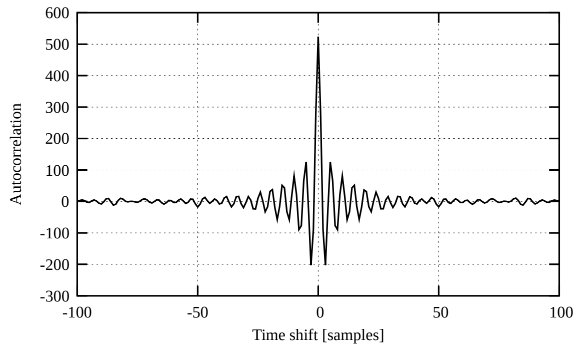

Figure 3. Autocorrelation function of sample TCD signal

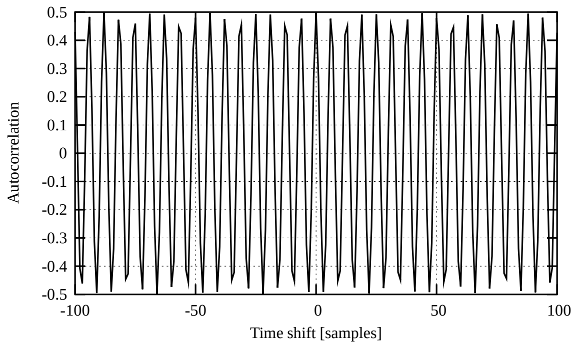

Figure 4. Autocorrelation function of sample harmonic interference

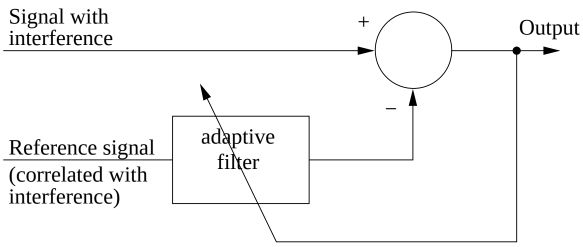

Figure 5. The general block diagram of Adaptive Interference Canceler

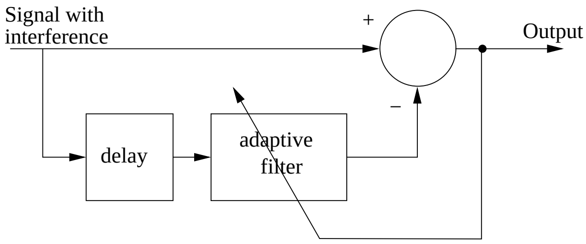

Figure 6. The block diagram of the Adaptive Interference Canceler for TCD signal

4. ADAPTIVE INTERFERENCE CANCELER

The Adaptive Interference Canceler (AIC) shown in the Fig. 5 removes from the noisy input signal the components, which are correlated with the reference signal (Ref. 7, Part I, Fig. 1.5d).

Because the autocorrelation function of the TCD signal tends to zero for higher delays, the sufficiently delayed TCD signal is not correlated with the non-delayed one.

On the other hand, the delayed interference signal is still correlated with the non-delayed signal, because its autocorrelation function is periodical (in fact it also tends to zero due to frequency instability, but for much higher delays).

This fact makes possible to build the Adaptive Interference Canceler, basing on the Adaptive Predictor (Ref. 7, Part I, Fig. 1.5a) structure shown in the Fig. 6.

The delayed TCD signal is a reference for the linear adaptive predictor. The output of the system is the error signal of the predictor. The error signal contains only those components of the disturbed signal, which are uncorrelated with the delayed signal. Therefore on the output we should obtain the clean TCD signal.

In the ideal case the adaptive filter should pass only the harmonic interferences, adjusting their phases and amplitudes to assure their cancellation when adding the delayed and filtered signal to the non-delayed one. The IIR filter with amount of poles equal to the amount of canceled harmonic signals should be best suited for that purpose. However, to avoid high complexity and potential instability problems associated with the adaptive IIR filters, much simpler FIR NLMS adaptive filter with complex coefficients has been used. Such a solution still provides reasonable results and may be efficiently implemented in simple DSP processor or in the FPGA.

The FIR filter is not able to pass the interference signals alone, the desired (but delayed) TCD component passes the delayed branch as well, and may affect the time/frequency properties of the output TCD signal.

Particularly, using too high delay causes an “echo” effect, shown in the Fig. 8.

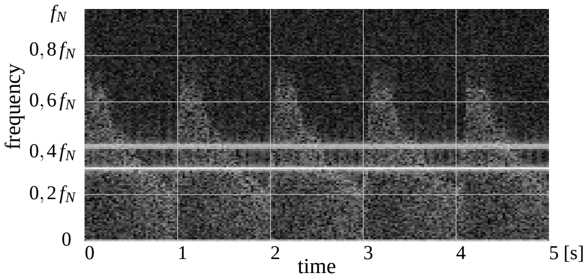

Figure 7. The simulated disturbed TCD signal (interference power 5dB above the signal power)

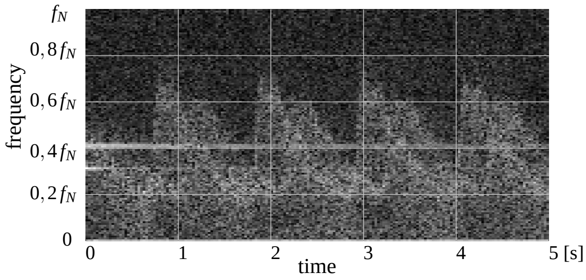

Figure 8. Results of too high delay - the undesired echo effect in the output signal

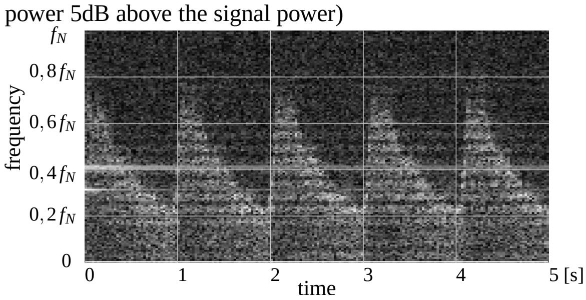

Figure 9. Results of too small delay - "frequency dropout" effect in the output signal

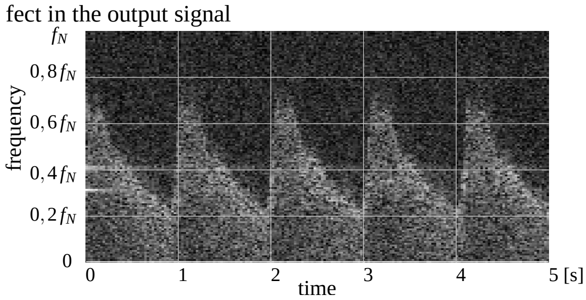

Figure 10. Signal with canceled interference, with the proper delay

On the other hand, using too small delay causes removing of some desired components from the output TCD signal. The visible sign of that problem are the dark horizontal stripes on the spectrogram (Fig. 9). The source of this effect is the cancellation of desired components with frequencies, for which the autocorrelation function at this delay significantly differs from zero.

The experiments have shown, that the optimum delay is the length of the quasistationary signal blocks used to perform FFT and to calculate the intermittent signal parameters. In this study this delay was equal to 256 samples at 10 kHz sampling frequency. At this delay both the "dropout effect" end the "echo" are still invisible (Fig. 10).

4.1. The adaptation algorithm

The simplest NLMS algorithm, described in Ref. 8 chapter 5.4.1 has been used. The algorithm was implemented to work with complex coefficients and complex signal. The adaptation coefficient was chosen as $\mu = 0.01$ to assure fast adaptation and good stability.

4.2. Selection of the order of the filter

The other problem is a selection of the appropriate order of the LMS filter. If the IIR filter was used - one pole for each canceled harmonic interference component (for the complex coefficients and signal) should be used.

In the case of FIR filter it is difficult to find such a simple rule. The numerical experiments have shown, that for single harmonic interference order of 1 is sufficient (which is obvious, because multiplying by a single complex coefficient may compensate the phase shift caused by delaying of the reference signal).

For two interferences the filter of order 5 was required, and for three harmonic interferences the order of 9 was necessary. So it seems, that the required order of the adaptive filter is equal to $4N-3$, where N is the number of canceled harmonic signals.

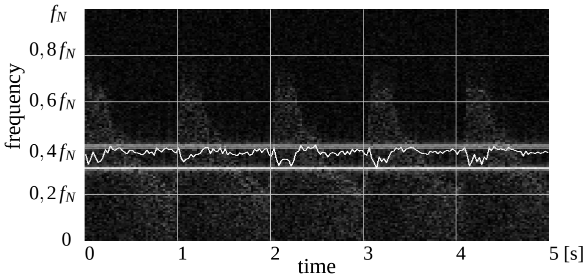

Figure 11. The simulated disturbed TCD signal and results of f_{max} estimation (NN based algorithm[2])

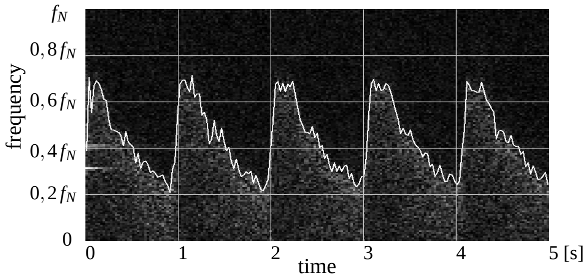

Figure 12. The simulated disturbed TCD signal after interference cancelation and results of f_{max} estimation (NN based algorithm[2])

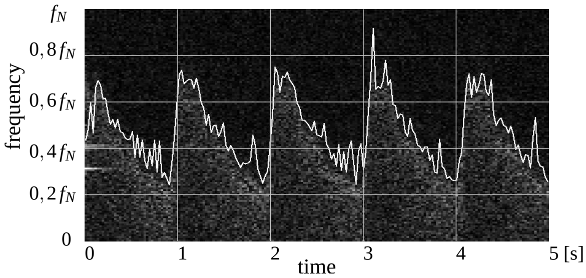

Figure 13. The simulated disturbed TCD signal after interference cancelation and results of f_{max} calculation ("geometrical" algorithm [9])

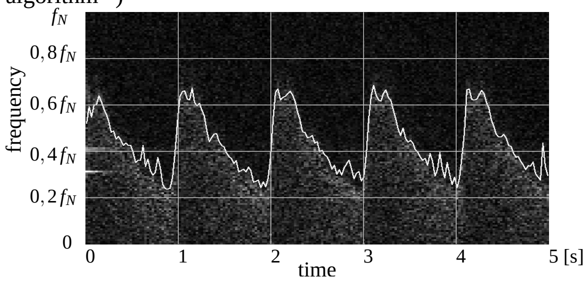

Figure 14. The simulated disturbed TCD signal after interference cancelation and results of f_{max} calculation ("percentile" algorithm, α=0.8)

5. RESULTS

Tests performed using both the recorded and simulated TCD signals have proven, that the described method is able to improve the disturbed signal and make it acceptable for further analysis. In the simulations it was possible to cancel interferences with power even 20 dB above the TCD signal power.

The figures 11 to 18 show two examples. First (Fig. 11 to 14) uses the simulated TCD signal, artificially disturbed with the interference consisting of two sinusoids with frequencies 0.32 f_N (Nyquist's frequency) and 0.42 f_N, and the total power 6 dB above the power of pure TCD signal.

The second example (Fig. 15 to 18) uses the real disturbed TCD signal digitally recorded from the outputs of TCD device.

As we can see it is impossible to estimate the f_{max} trend from the disturbed signals (Fig. 11 and 15).

However after processing with the presented Adaptive Interference Canceler, the interferences are suppressed sufficiently to allow successful estimation of the maximum frequency trend (Fig. 12 to 14 and 16 to 18).

However different algorithms for f_{max} estimation show different sensitivity to the residual harmonic interferences left after the cancellation. Therefore the thorough selection of analysis algorithms is required. In that study the best results were obtained using the "neural network based" algorithm described in Ref. 2.

6. CONCLUSIONS

The presented Adaptive Interference Canceler (AIC) may be used to eliminate the harmonic interferences from the Transcranial Doppler signals.

The proposed AIC is able to cancel interferences even of power higher, than the power of TCD signal. With this method it is possible to successfully analyze signals, which otherwise would be useless.

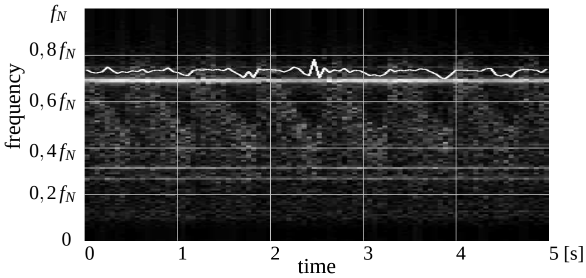

Figure 15. The real disturbed TCD signal and results of f_{max} calculation (NN based algorithm[2])

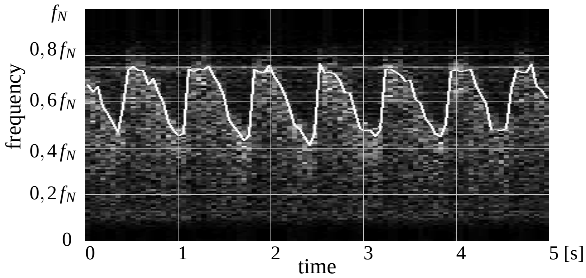

Figure 16. The real disturbed TCD signal after interference cancelation and results of f_{max} calculation (NN based algorithm[2])

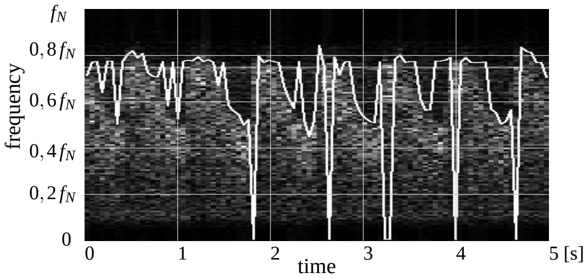

Figure 17. The real disturbed TCD signal after interference cancelation and results of f_{max} calculation ("geometrical" algorithm[9])

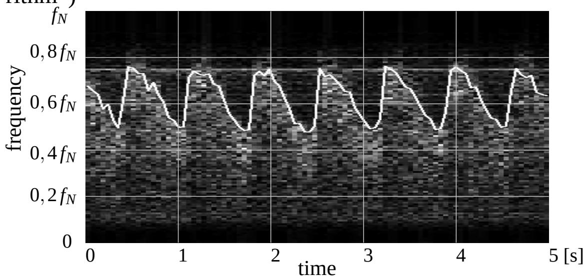

Figure 18. The real disturbed TCD signal after interference cancelation and results of f_{max} calculation ("percentile" algorithm, α=0.8)

The AIC is based on the adaptive NLMS FIR filter, which makes implementation easy even in simple fixed point DSP processor or in the FPGA based DSP system.

The required order of adaptive filter depends on the number of harmonic components (N) in the interfering signal. The empirical rule is that the order should be equal to $4N - 3$.

In further research the use of adaptive IIR filters should be investigated, as those filters may be better suited for cancelation of narrow-band interferences in the presented AIC structure.

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REFERENCES

1. P. J. Kirkpatrick and K.-H. Chan, "Transcranial doppler," in *Head Injury*, P. Reilly and R. Bullock, eds., ch. 13, pp. 243–259, Chapman & Hall, 1997.
2. W. M. Zabołotny, P. Wołłk-Łaniewski, E. Łuczywek, and E. Fersten, "Application of neural networks in estimation of maximum frequency in transcranial doppler signal," in *Proceedings of the MBB 98*, pp. 3–67 – 3–70, 1998.
3. W. M. Zabołotny, P. Wołłk-Łaniewski, E. Łuczywek, and E. Fersten, "Estimation of maximum frequency in the nonstationary transcranial doppler signal - neural network based approach," in *Proceedings of the XXI-st National Conference Circuit Theory and Electronic Networks*, pp. 529–534, 1998.
4. W. Zabolotny, *Methods of Estimation of Maximum Frequency in the Transcranial Doppler Signal (PhD thesis, in Polish)*, Warsaw University of Technology, Faculty of Electronics and Information Technology, Warszawa, 1999.

5. L. Y. L. Mo and R. S. C. Cobbold, “A stochastic model of the backscattered doppler ultrasound from blood,” *IEEE Transactions on Biomedical Engineering* **BME-33**, pp. 20–27, Jan. 1986.
6. B. A. J. Angelsen, “A theoretical study of the scattering of ultrasound from blood,” *IEEE Transactions on Biomedical Engineering* **BME-27**, pp. 61–67, Feb. 1980.
7. B. Widrow and S. Stearns, *Adaptive Signal Processing*, Prentice-Hall, Inc., New Jersey, 1985.
8. F. Gustafsson, *Adaptive Filtering and Change Detection*, John Wiley & Sons, 2000.
9. K. Marasek and A. Nowicki, “Comparison of the performance of three maximum doppler frequency estimators coupled with different spectral estimation methods,” *Ultrasound in Med. & Biol.* **20**(7), pp. 629–638, 1994.