

Broad-Band Noise Reduction using Least Mean Square (LMS)-Adaptive Line Enhancer (ALE) on Doppler Blood Flow Sound Signal

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ABSTRAK

Haemorrhoidal artery ligation (HAL) adalah metode operasi wasir yang memiliki keunggulan rasa sakit minimal dan waktu pemulihan cepat. Prosedur HAL bergantung pada bunyi Doppler untuk menentukan letak arteri dalam liang anus. Derau yang tercampur dalam sinyal membuat dokter kesulitan mendengarkan bunyi Doppler yang menandakan letak arteri wasir. Adaptive line enhancer (ALE) dengan algoritma least mean square (LMS) digunakan dalam penelitian ini untuk mengurangi aras derau yang terkandung dalam sinyal bunyi Doppler aliran darah. Hasil simulasi menunjukkan parameter mean square error (MSE), signal to noise ratio (SNR), dan waktu eksekusi untuk berbagai orde tapis. Nilai maksimum power spectral density (PSD) sinyal yang masuk ke tapis -86 dB/Hz, kemudian turun menjadi -101 dB/Hz setelah lewat tapis. SNR tertinggi 40,03 dB didapat ketika orde tapis 16. Waktu eksekusi hampir tidak berubah meskipun orde filter dinaikkan.

Kata kunci: pengurangan derau, bunyi Doppler, ALE, LMS, HAL

ABSTRACT

Hemorrhoidal artery ligation (HAL) is one of surgical methods in treating hemorrhoids disease known for less pain and fast recovery. HAL procedure relies on the Doppler audible sound to locate arteries within rectal column. Noise interfering the signal of interest makes arteries detection becomes difficult task for the surgeon since signal and noise spectral overlaps. Adaptive line enhancer (ALE) with least mean square (LMS) algorithm is used in this study to reduce noise level contained in Doppler blood flow sound signal. The simulation results show the mean square error (MSE), signal to noise ratio (SNR), and execution time for different filter order. The maximum power spectral density (PSD) value of the noisy signal and filtered signal are -86 dB/Hz and -101 dB/Hz, respectively. The highest SNR of 40.03 dB is obtained when filter tap order is set to 16. However signal processing time remains almost unchanged as the filter tap order increased.

Keywords: noise reduction, Doppler sound, ALE, LMS, HAL

1. INTRODUCTION

In 1995, Japanese surgeon Morinaga proposed a novel non-invasive surgical treatment for hemorrhoids disease called HAL (**Morinaga, et al., 1995**). Several studies reported successful results using HAL in treating patients suffered with majority grade II or III hemorrhoids. These studies also demonstrated that the technique is relatively painless and can reduce length of hospital admission (**Hoyuela, et al., 2016**) (**Mott, et al., 2018**). The main purpose of HAL procedure is to significantly lessen arterial blood flow into hemorrhoidal tissues. This subsequently results in shrinkage of the tissues and cessation of hemorrhoidal bleeding. The method requires accurate identification of the superior rectal arteries supplying hemorrhoids using a Doppler transducer attached on side wall of specially designed proctoscope.

Obviously, hemorrhoidal arterial detection plays important role in overall success of HAL procedure because it provides information to the surgeon about the orientation of each hemorrhoidal artery on rectal circumference. Hemorrhoidal arteries, which are usually located at the odd-numbered clock positions (**Giamundo, et al., 2016**), are explored by carefully rotating or tilting the proctoscope. However, some arteries are not located at the usual odd hours position. The surgeon performing HAL procedure only relies on the Doppler sound produced by Doppler processor to locate hemorrhoidal artery within rectal column. Doppler sound is a specific sound pattern characterized by middle – low frequency spectrum with pulsing amplitude in accordance with the pulsatile blood movement inside the blood vessel. Generally, this sound is clearly audible to the surgeon when the Doppler transducer is right on top of the hemorrhoidal artery. However, there is variety in Doppler sound intensity due to several factors, such as the position of the artery, the distance from the Doppler probe, and the direction of blood flow relative to ultrasound waves emitted by the transducer (**Ratto, et al., 2014**). Since audible sound produced by Doppler processor carries essential information, so it is important to have a clean signal.

Most of the signals extracted from human body are periodic or have cyclostationary components. These biomedical signals are very weak and often buried by background noise or combined with either periodic or non-periodic signals (**Jiao, et al., 2012**). In healthcare settings, noise possibly caused by electromagnetic interference (EMI) between medical equipment and radio waves emission from mobile phones, wireless local area networks (WLANs), radio station transmitters, or even other medical devices (**Ishida, et al., 2016**). Medical equipments that are used for measuring or detecting body parameters are more susceptible to interference because they can measure very low – level signal and have probing wires act as antenna. Equipment for performing HAL procedure falls into this category. To make matters worse, HAL equipment is normally operated in surgery room, the location in hospital where has highest electric field strength readings, over 30 V/m (**Iadanza, et al., 2019**).

Regular filters are difficult to remove background without losing essential information due to overlapping spectrum of biomedical signals and noise. Adaptive filters can reduce noise while preserving biomedical signals even though they have similar spectrum. Among numerous constructions, adaptive line enhancer (ALE) has been widely used for separation of a narrow – band signal from broad – band noise (Gaussian noise). ALE is effective denoising tool in the situation where signal and noise cannot be extracted individually because ALE does not require direct access to noise source. An adaptive filter automatically tunes its linear filter coefficients according to an adaptive algorithm. Among others, least mean square (LMS) algorithm is the most popular mostly because of its computational simplicity. ALE with LMS has various

applications in biomedical signal processing area, yet its performance in reducing background noise within Doppler blood flow sound signal has not been investigated thus far.

There are various studies dealing with usage of adaptive filters in biomedical signals extraction. N. Razzaq et al. applied adaptive filter based on recursive state model to eliminate power line interference (PLI) as well as its harmonics in high – resolution electrocardiogram (HRECG) **(Razzaq, et al., 2016)**. Different methods comprising of discrete wavelet transform (DWT), Normalized LMS (NLMS), finite impulse response (FIR), and infinite impulse response (IIR) were applied by S. Saxena et al. to denoise the ECG signal which was corrupted by the PLI **(Saxena, et al., 2019)**. In the research conducted by E. Fotiadou et al. time-sequenced adaptive filtering algorithm was utilized to enhance quality of abdominally extracted fetal ECG ($fECG$) signal that contaminated by interferences. Major contribution coming from the maternal ECG ($mECG$), maternal and fetal muscle noise, and multiple layers of dielectric biological tissues through which the electrical signal must pass **(Fotiadou, et al., 2018)**. An approach using ALE with LMS has been reported in the literature to address breath and cardiac sounds separation. This issue needs to be addressed since cardiac sounds can cause interference during lung sound analysis **(Krishnaswamy, et al., 2015)**. Under certain clinical situations, medical practitioners may perform respiratory sounds auscultation while patient being transported to the hospital. However, the sound of sirens of the ambulance often create huge disturbance to the respiratory sound. A study presented the application adaptive filter based on LMS algorithm combined with low pass Butterworth filter to reduce both the harmonics and fundamental frequency of the sirens **(Lu, et al., 2015)**. Another research proposed NLMS adaptive filter with modified step size to eliminate baseline wandering from ECG signals with minimal distortion to the original signal **(Rahman, et al., 2017)**.

This study presents a technique to improve arterial Doppler sound quality that is buried in background noise by utilizing ALE with LMS using real time recorded sound signal. The results show the MSE, SNR, and execution time for different filter order.

2. MATERIAL AND METHODS

2.1 Adaptive filtering systems and adaptive algorithms

Conventional filters are effective when characteristics of the signal, noise, and transmission channel are known and there is no spectral overlap between noise and signal. However, when they vary over time and spectral overlap occurs, adaptive filters should be used. Adaptive filters adapt to these changes to suppress or eliminate noise in the most favorable way.

There are two similar adaptive filtering systems for noise cancellation with slightly different architectures that is adaptive noise canceller (ANC) and ALE. The early version of ANC uses different sensors to receive signal and noise separately whereas ALE only needs single sensor to receive the signal covered with noise.

Accomplishing the finest performance of an adaptive filter demands incorporation of the best adaptive algorithm with simple computational process and rapid convergence rate. The LMS and recursive least square (RLS) are the most common algorithms used in adaptive filters. LMS is one of the most attractive algorithms in adaptive filtering applications due to its simplicity, stability, ease of implementation, robustness, and low computational cost **(Dixit, et al., 2017)**. LMS iteratively makes continuous corrections to the weight vector towards the negative of the gradient vector which finally reaches the minimum mean square error. It is not required to calculate correlation function or to perform matrix inversions. Its iterative algorithm

makes LMS suitable for highly time – varying signal environment. No memory involved in LMS algorithm, older values play no role in the considered total error. However, convergence takes very long time if the adaptive filter is fed with high spectral dynamic range signal (**Haykin, et al., 2014**), for example in non-stationary situations and non-white background noise. Another disadvantage is the fact that it has a fixed step size parameter for every iteration so it is required to understand the statistics of the input signal before starting the adaptive filter operation. RLS algorithm, on the other hand exhibits faster convergence, particularly for highly correlated input signals and tends to produce smaller error than the LMS algorithm. The RLS algorithm works by minimizing the total squared error between the desired signal and the output signal of a unknown system. However, RLS has stability problems and has more complex computational process compared to LMS – based algorithms (**Potdar, et al., 2015**).

The LMS principle is based on the steepest descent method to adjust its filter coefficients, reducing mean square prediction of a transversal filter. The summary of the LMS algorithm can be expressed using Equation (1) and Equation (2).

$$y(n) = x^T(n)w(n) \quad (1)$$

$$e(n) = d(n) - x^T(n)w(n) \quad (2)$$

In equations above, $x(n)$ is the input signal, $w(n)$ is the weight vector, $y(n)$ is the output signal of adaptive transversal filter, and $e(n)$ is the error signal. The equations here use the present estimate of the weight vector of the adaptive transversal filter. The weight vector update recursion of the conventional LMS can be expressed by Equation (3).

$$w(n + 1) = w(n) + \mu e(n)x(n) \quad (3)$$

Where μ is step size variable to control the convergence behaviour within a proper range. A too low value of μ causes the algorithm runs very slow towards convergence, whereas a too high value of μ causes the algorithm to diverge, thus makes adaptive filter difficult to achieve optimum error performance. Therefore, when implementing LMS as an adaptive filter, suitable value of step size is crucial for the filter to work properly.

ALE is actually a derivation of LMS algorithm which is introduced to solve the aforementioned problems. It is a self-tuning adaptive filtering system which is able to separate the periodic and stochastic components in a signal (**Widrow, et al., 1976**). The ALE detects very low-amplitude sine waves in noise and can be applied in noisy environment setting. Figure 1 shows structure of ALE.

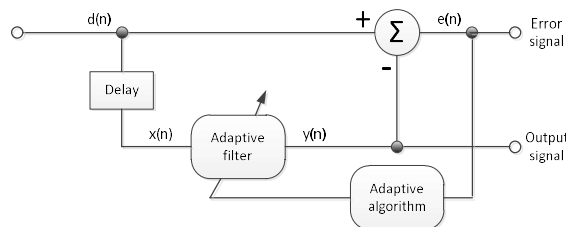


Figure 1. Structure of Adaptive Line Enhancer

ALE's single sensor simultaneously receives a signal and a noise uncorrelated with the signal $s(n) + \hat{x}(n)$ as input $d(n)$ to the enhancer. Delayed version of $d(n)$, denoted by $x(n)$ produced by delay $z^{-\Delta}$ to de-correlate the noise while the target signal remains correlated. The $x(n)$ is then filtered by the adaptive filter to produce an output signal $y(n)$ which is ideally an estimate

of the noise-free input signal. $y(n)$ is subtracted from $d(n)$ and appears as error signal, or $e(n) = d(n) - y(n)$.

ALE is indeed simple and easy to control since it only uses a single sensor. However, to achieve the best performance in its computational process, convergence rate of adaptive algorithm and complexity of adaptive filter structure must be well balanced.

2.2 Doppler blood flow detection

The Doppler Effect is the frequency shift of the signal which is caused by a moving object. Obtaining a Doppler signal from a blood is possible since it contains scattering particles in form of red blood cells. The size of a red blood cell is about $2 \times 7 \mu\text{m}$, which means that a Rayleigh scattering will occur. If the blood flow is moving closer to the transducer, the detected frequency will be higher, conversely, if the blood flow is moving away from the transducer, the detected frequency will be lower. Figure 2 illustrates an ultrasound beam is emitted toward moving red blood cells as the reflecting objects. The beam collides with the object and bounces back to the receiver with a Doppler-shifted frequency.

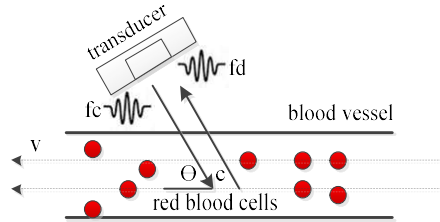


Figure 2. Doppler ultrasound measures the movement of the scatterers based on the frequency shift between transmitted and received beam

The shifted Doppler signal f_D of an ultrasound signal with the original frequency f_0 is expressed in Equation (4).

$$f_D = \left(\frac{2v \cos \theta}{c} \right) f_0 \quad (4)$$

where v is the velocity of the moving objects and c is the propagation speed of sound in the medium which is between 1500 to 1600 m/s in soft tissue and commonly set to 1540 m/s (**Atkinson, et al., 1975**). In practical use, the frequency f_0 is in the range of 2 to 10 MHz, which means the signal has wavelength between 0.15 and 0.77 mm. As Doppler ultrasound beam becomes more aligned to the flow direction, Doppler frequency increases. The peak systolic flow velocity in the haemorrhoidal arteries is normally 20 to 750 mm/s (**Fish, et al., 1991**), causing a Doppler shift in the range of 0.2 to 7.5 kHz from the original frequency.

A pulsed Doppler system must be used in order to get exact location along the beam the blood flow data are collected. In pulsed mode, the transducer transmits several cycles of the ultrasound signal, then after a specified time a gate is opened to allow transducer acts as receiver.

2.3 LMS-ALE Implementation

The proposed adaptive filtering system is shown in Figure 3. The blood flow sound signal was acquired with an 8 MHz transducer at wrist to target blood flowing through radial artery. This

vessel was chosen rather than the rectal artery because of the practical reason. After all, blood flowing through both radial artery and rectal artery will produce similar Doppler sound. The frequency-shifted signal received by the transducer was then processed by the processor to produce Doppler signal with the frequency within audible spectrum. The signal amplification was performed by a bluetooth-equipped portable amplifier. A particular pulsed sound will be emitted by the speaker when the transducer is situated on the right spot on the wrist, as indication that there is a blood stream beneath.

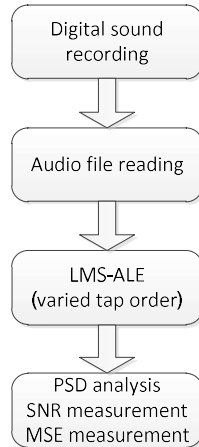


Figure 3. System design for Doppler blood flow sound signal adaptive filtering using LMS – ALE

The arterial sound was recorded using digital sound recorder for 10 seconds with sampling rate of 44.1 kHz. Signal processing and simulations was fully performed in Matlab environment. Figure 4 shows the real time recorded sound signal. It contains the Doppler blood flow sound signal with interference. The interference sources are mostly combination of the internal switched-mode power supply and bluetooth transmission channel which generate background noise as illustrated in Figure 5.

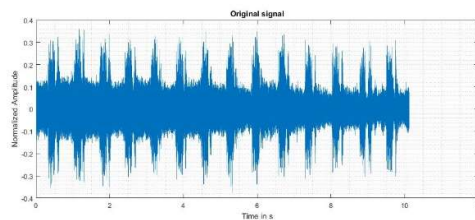


Figure 4. Doppler blood flow sound signal with noise

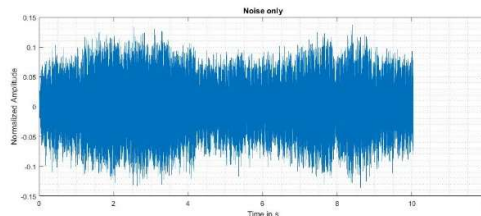


Figure 5. Noise contained in the signal

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A brief observation in frequency domain reveals that spectrum of the noise overlaps signal's spectrum but with lower amplitude, as illustrated in Figure 6. When there is spectral overlap between signal and noise, not conventional filters but adaptive filters are useful. Adaptive filter in form of ALE with LMS algorithm is used in this study to reduce noise level contained in Doppler blood flow sound signal.

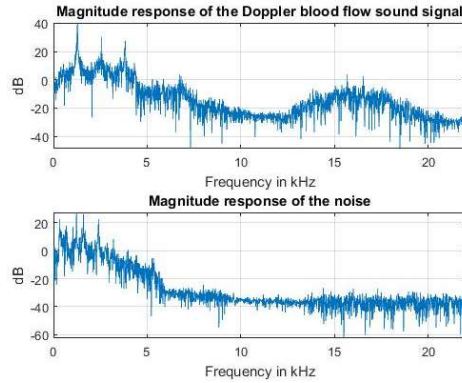


Figure 6. Doppler blood flow sound and interference signal in frequency domain

Recorded Doppler blood flow sound signal in .wav format was fed into ALE's single input. This signal, together with the signal's delayed version generated by delay unit inside the structure, were used to produce an output closely resembling input signal with reduced noise.

Different metrics used to measure the performance of adaptive filter are PSD, SNR and MSE. Filter tap order was varied to find the optimum value of SNR parameter.

4. RESULTS AND DISCUSSION

Figure 7 shows the noisy signal as well as the output of the LMS – ALE of the IIR filter of order 16 designed to remove interferences. Eventhough the noise is not completely removed, still there is some improvement at the output. In order to assess signal improvement quantitavely, PSD is analyzed both for noisy and filtered signal.

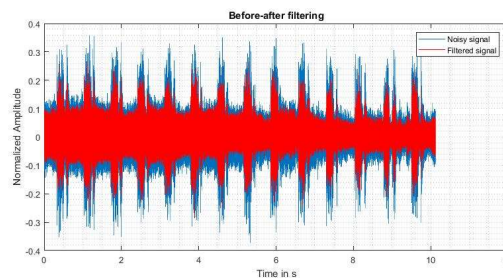


Figure 7. Doppler blood flow sound signal before and after enhancement process with LMS – ALE

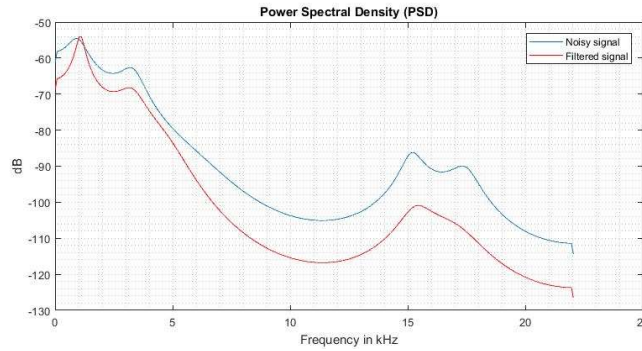


Figure 8. PSD of noisy and filtered signal

From Figure 8 it is seen that the maximum PSD value of the noisy signal is -86 dB/Hz. However, after the noisy signal was passed through LMS – ALE the maximum PSD value corresponding to the spectrum of the noise was then reduced to -101 dB/Hz. This means 17,5% in noise reduction. The figure also shows that space between two curves is wider at higher frequencies, indicating that the signal attenuation is more intensive outside the spectrum of interest. SNR is also one of the major parameters to judge the quality of the signal.

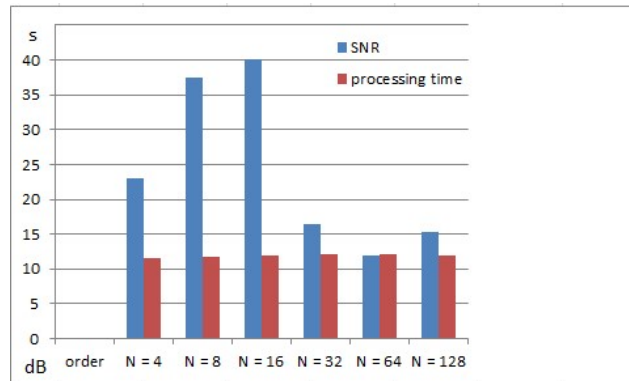


Figure 9. SNR rate at different filter orders

Figure 9 shows the SNR value measured at the output of the adaptive filter for different tap orders. As it can be seen the highest SNR of 40.03 dB was obtained when filter tap order was set to 16. Higher filter tap orders exhibited worse performance. Also according to the figure, signal processing time did not change significantly as the filter tap order increased.

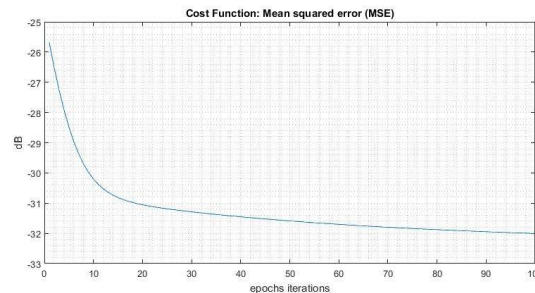


Figure. 10. MSE convergence with step size 10^{-3}

In this experiment, the optimization criterion is mean square error (MSE). It is the mean square of the error signal between the output of the adaptive filter and the desired signal. LMS incorporates an iterative procedure that updates coefficients which leads MSE converges to its minimal value. The convergence rate of the LMS algorithm depends heavily on the step-size parameter. The convergence characteristic of the LMS – ALE used in this experiment is shown in Figure 10. As it can be seen, the convergence starts after 10 iterations.

4. CONCLUSIONS

Adaptive filter in form of ALE with LMS algorithm is simulated in this study using MATLAB to reduce noise contained in Doppler blood flow sound signal. Based on the simulation, it is observed that the noise is not completely removed from the original signal. However, there is still signal enhancement after filtering process, indicated by reduced maximum PSD value of noise spectrum from -86 dB/Hz to -101 dB/Hz. It is also discovered that increasing filter tap order is not necessarily improving filter performance as well as processing time. The best SNR of 40.03 dB can be obtained when filter tap order is set to 16. Further increase in this parameter degrades filter performance and has no effect in processing time. MSE curve reveals that the error has converged into a minimum value, indicating the adaptive filter has been already in its optimal state. However, there is still plenty room for improvement on the performance of the ALE. Possible ways include altering filter structure and modifying adaptation algorithm.

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REFERENCES

- Atkinson, P. (1975). A Fundamental Interpretation of Ultrasonic Doppler Velocimeter. *Ultrasound in Medicine and Biology*, 2, 107–111.
- Dixit, S., & Nagaria, D. (2017). LMS Adaptive Filters for Noise Cancellation: A Review. *International Journal of Electrical and Computer Engineering (IJECE)*, 7(5), 2520 - 2529.
- Fish, P.J. (1991). Non-Stationary Broadening in Pulsed Doppler Spectrum Measurements. *Ultrasound in Medicine and Biology*, 17, 147–155.
- Fotiadou, E., van Laar, J.O.E.H., and Oei S.G. et al. (2018). Enhancement of Low-Quality Fetal Electrocardiogram based on Time-Sequenced Adaptive Filtering. *Medical & Biology Engineering & Computing*, 56, 2313–2323.
- Giamundo, P. (2016). Advantages and Limits of Hemorrhoidal Dearterialization in the Treatment of Symptomatic Hemorrhoids. *World Journal of Gastrointestinal Surgery*, 8(1), 1–4.

- Haykin, S.O. (2014). *Adaptive Filter Theory 5th ed*, New York: Pearson.
- Hoyuela, C., Carvajal, F., and Juvanny, M. et al. (2016). HAL-RAR (Doppler-Guided Haemorrhoid Artery Ligation with Recto-Anal Repair) is a Safe and Effective Procedure for Haemorrhoids. Results of a Prospective Study after Two-Years Follow-Up. *International Journal of Surgery*, 28, 39-44.
- Iadanza, E. (2019). *Clinical Engineering Handbook*. London: Elsevier Science.
- Ishida, K., Fujioka, T., & Endo, T. et al. (2016). Evaluation of Electromagnetic Fields in a Hospital for Safe Use of Electronic Medical Equipment. *Journal of Medical Systems*, 40(46), 1 - 11.
- Krishnaswamy, S. and Muniraj, N. (2015). Application of Adaptive Line Enhancer with LMS to Separate Heart Sound Signal from Lung Sound Signal at Real Time. *Research Journal of Applied Sciences, Engineering and Technology*, 9(6), 448–453.
- Lu, B. Y., Hsueh, C.M.L., and Hung H.W. et al. (2015). Auscultation on the Ambulance: A Case Study of Adaptive Filter Application for the Safety. *Journal of Electrical Engineering*, 3, 203–211.
- Morinaga, K., Hasuda, K., and Ikeda, T. (1995). A Novel Therapy for Internal Haemorrhoids: Ligation of the Haemorrhoidal Artery with a Newly Devised Instrument (Moricorn) in conjunction with a Doppler Flowmeter. *American Journal of Gastroenterology*, 9, 610–613.
- Mott, T., Latimer, K., and Edwards, C. (2018). Hemorrhoids: Diagnosis and Treatment Options. *American Family Physician*, 97(3), 172–179.
- Potdar, R.M., Meshram, M., & Kumar, R. et al. (2015). Review of Adaptive Filter Algorithm and Process for Echo Cancellation. *International Journal of Advanced Research in Electronics and Communication Engineering (IJARECE)*, 4(6), 1758 - 1761.
- Rahman, M.S., Kim, Y., and Choi C. et al. (2017). A Comprehensive Adaptive Filter to Eliminate Baseline Wander from ECG Signals. *2017 International Conference on Information and Communication Technology Convergence (ICTC)*, (pp. 43-48).
- Ratto, C. (2014). THD Doppler Procedure for Hemorrhoids: the Surgical Technique. *Techniques in Coloproctology*, 18, 291–298.
- Razzaq, N., Sheikh S.A., and Salman, M. et al. (2016). An Intelligent Adaptive Filter for Elimination of Power Line Interference from High Resolution Electrocardiogram. *IEEE Access*, 4, 1676-1688.

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Doppler Blood Flow Sound Signal

- Sanei, S., Lee, T.K.M, & Abolghasemi, V. (2012). A New Adaptive Line Enhancer Based on Singular Spectrum Analysis. *IEEE Transactions on Biomedical Engineering*, 59(2), 428 - 434.
- Saxena, S., Jais, R., and Hota, M. K. (2019). Removal of Powerline Interference from ECG Signal using FIR, IIR, DWT and NLMS Adaptive Filter. *2019 International Conference on Communication and Signal Processing (ICCSP)*, (pp. 0012-0016).
- Widrow, B. Glover Jr., J.R., and McCool, J.M. et al. (1976). Adaptive Noise Cancelling: Principles and Applications. *Proceedings of the IEEE*, 63(12), 1692 – 1716.