ISSN 2347 - 3983
Volume 8. No. 8, August 2020
International Journal of Emerging Trends in Engineering Research
Available Online at http://www.warse.org/IJETER/static/pdf/file/ijeter90882020.pdf
https://doi.org/10.30534/ijeter/2020/90882020

PLI Cancellation from ECG signal Using Modified Sign-Regressor Normalized Least Mean Square Sub band Adaptive Algorithm

Madduri Venkateswarlu[1], S. Nagakishore Bhavanam[2]
[1]Research Scholar, Dept. of ECE,,University College of Engineering & Technology,
Acharya Nagarjuna University, Guntur, Andhra Pradesh,India
[1] maddurii.venkateswarlu@gmail.com
[2]Assistant Coordinator, IQAC, Dept. of ECE, University College of Engineering & Technology,
Acharya Nagarjuna University, Guntur. Andhra Pradesh,India
[2] kishore.ece.anu@gmail.com

ABSTRACT

The Electrocardiogram (ECG) record is a procedural electrical activity of the heart which is non invasive recording and is acquired by surface electrodes at designated locations on the skin of patient's body during acquisition, different artifacts/noises such as power-line interference (PLI), baseline wander (BW), electrode motion artifacts (EM) and muscle artifacts (MA), obscure the ECG. The artifacts need to be removed prior to diagnosis by the medical experts. In this paper work an ECG denoising design structure using hybrid subband adaptive filter (HSAF) is constructed to solve structured problems in conventional sub band adaptive filter (SAF). This paper investigates the new detailed adaptive noise canceller (ANC) system for ECG signals with robustness based on subband decomposition structured HSAF using proposed modified sign-regressor normalized least mean square (MSRNLMS) adaptive algorithm. Uniform and non-uniform subband decomposition structured SAF algorithms are applied on ECG records obtained from MIT-BIH data base and the performance is compared in terms of parameters SNR, MSE, RMSE and distortion.

Key words: ECG, PLI, BW, MA,EM, filter bank, HSAF, ANC

1.BACKGROUND

The signal processing is one of the important methods to attenuate unwanted signal present in the original signal. The signal which is used for medical source or biological is known as biomedical signal. The biomedical signs are taken from atomic level, cell level, fundamental or organ level. Models for the biomedical signs are Electrocardiogram (ECG) which is the electrical action from the heart, Electroencephalogram (EEG) which is the electrical action from the cerebrum, Electromyogram (EMG) which is the electrical action from the muscle, etc. The biomedical signs are the amazing sources to investigation the conditions or determination and assessing the appropriate treatment.

Electrocardiogram (ECG) is a wave which is created from the heart. The electrical movement of heart is recorded as a wave structure which is called as Electrocardiogram signal. The ECG signal is utilized to finding of the heart. The ECG signal is taken by putting the cathodes on the outside of the skin which is the likely contrast between the terminals. The electric field around the heart can be estimated by utilizing terminals by setting them on the skin of the patient. The force of the voltage recognized relies upon the direction of cathodes regarding that of dipole closes [1]. The sufficiency of the sign is corresponding to the mass of tissues associated with making that dipole at some random time. Anodes on the outside of the skin to recognize the voltage of this electric field is called electrocardiogram (ECG). The ECG signal is portrayal of the atrial depolarization/repolarization and ventricular depolarization/repolarization during the heart beat.

The basic noises present in the ECG signal are Power Line Interference noise, Baseline Wander noise, motion artifacts, Electromyographic noise. The Power Line Interface clamor is a result of capacitor and inductance coupling. The capacitance commotion is because of vitality sharing between two frameworks. The neighboring framework having coupling capacitors are answerable for this clamor. Inductance commotion is a direct result of current stream in the wire. When there is a current it will bring about the creation of attractive field. The attractive fields of various frameworks bring about common inductance. This common inductance is liable for inductance clamor. Contrasted with inductance commotion the capacitor clamor is high recurrence clamor. A typical static (fixed coefficient) filter is designed in advance with statistical information of both desirable signal and the un desirable signal *i.e* noise, but if the statistics of the noise are not known earlier, or revision after some time, the coefficients of the filter can't be determined ahead of time. In these things, adaptive algorithms are required with a specific end goal to persistently refresh the filter coefficient [2].

Adaptive filters have two sections like computerized channel and adaptive calculation. Advanced filters are utilized to play out the separating activity and the adaptive calculation is utilized to refresh the loads or coefficients of the channel. Adaptive filter applies assortment of calculations, for example, LMS and NLMS calculations and so forth. The filter tap length (M) and the learning rate boundary (μ) impact the presentation of the calculation. The principle disadvantage of utilizing FIR filter in versatile filtering is high computational multifaceted nature and more slow intermingling [3]. Figure 1 defines the basic drawback and noise cancelling answer. As appeared in the figure u(n) is reference input, y(n) is yield, d(n) is wanted sign, e(n) is error signal. Error signal is contrast of wanted sign and the yield signal. Versatile calculations are utilized to refresh the channel coefficients dependent on the mean square estimation of the error. The yield of the versatile channel is given in Equation 1.

$$e(n)=d(n)-y(n) \tag{1}$$

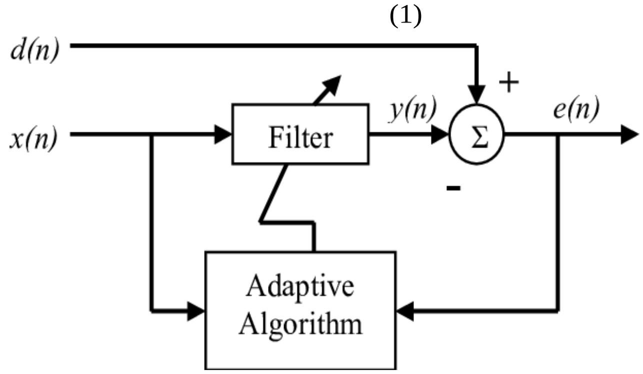


Figure 1: Block diagram of Adaptive Noise Canceller

Conventional filters, for example, versatile filters [4], sign based standardized versatile filters [5] were recommended in the writing to limit noises (artifacts). Different ECG denoising techniques join new factor step size NLMS [6,7]. Promising execution are obtained by non direct versatile calculations [8]. Channel bank (FB) breaks down a computerized signal into various recurrence groups. Based on time recurrence goal, FB's can be sorted into two kinds i.e., uniform filter bank (UFB) and non uniform filter bank (NUFB). UFB gives fixed and uniform time recurrence goal. While non uniform filter bank gives non-uniform and variable time recurrence goal prompting better execution and decreased number juggling multifaceted nature in certain applications like sound investigation, ECG signal upgrade [9,10,19,20]. Various techniques for decaying signals into sub groups have gotten conspicuous and were proposed in the writing [11]. Sub band versatile filter (SAF) structures have been proposed to conquer the issues of versatile filters. Thusly, versatile filtering utilizing sub band coding transforms into a superb decision for such countless versatile frameworks.

As of late extraordinary commotion crossing out techniques are proposed utilizing SAFs incorporate variable advance size sign SAF [12], Variable individual advance size SAF [13] and new standardized SAF [14]. This proposed strategy utilizes FBs to pulverize the info signal into various recurrence groups, each filling in as an unprejudiced contribution to SAF. In customary SAF's each bound uses an individual versatile sub filter in its own adaption that diminishes the union rate.

To take care of these organized issues multiband structured SAF (MSAF) are created in which the full band adaptive filter's tap weight vectors are refreshed by a solitary adaptive calculation utilizing sub band signal [9,15]. The point of paper is to create non uniform multi band organized sub band versatile filter which can refine the presentation of the conventional ANC, to dissect the use of MSAF to clamor dropping issue in ECG.

2. DESIGN METHODOLOGY

The adaptive filters are notable for self-learning advanced filter by changing the channel coefficients. adaptive filters are utilized in numerous applications however for high request sifting it requires more assembly rate. To conquer this issue another method is accessible called as sub-band adaptive filters. The essential thought of sub-band adaptive filters is shown up from sub-band coding. The examination/combination channels are utilized for sub-band coding is known as the Quadrature Mirror Filter (QMF) bank. In sub-band versatile sifting input signal and wanted sign are isolated into sub-groups and for each sub-band separate calculations are performed. The union pace of sub-band versatile channel is constrained by associating and band edge impacts. For this issue multiband weight control instrument has been created in which the full-band versatile channel is refreshed by input sub-band signals standardized by their particular changes. In view of the multiband structure, a recursive weight-control component called the multiband-strctured sub-band adaptive filter (MSAF).

The concept of multiband structured-sub band adaptive filter (MSAF) is presented in this section. In the intended method, the primary input signal given to the upper FB in Figure 2 consists of noise contaminated ECG record denoted as ***d(n)***. The Secondary (reference) signal is given to SAF (lower FB with adaptive filter) is noise signal $n_2(n)$ denoted as ***u(n)***. The full band input signal ***u(n)***, primary input signal or desired response ***d(n)*** and filter output signal ***y(n)*** are decomposed into ***N*** sub bands by means of analysis filters $H_i(z)$; for $i=0,1,2...(N-1)$.In this Figure.3 $H_0,H_1,H_2....H_{N-1}$ and $F_0,F_1,F_2....F_{N-1}$ are analysis & synthesis filters of ***N*** channel perfect reconstruction (PR) filter bank respectively. These sub band signals are decimated to a lower rate using same factor and are processed by individual sub band adaptive sub filters ***W (z)***.[10]

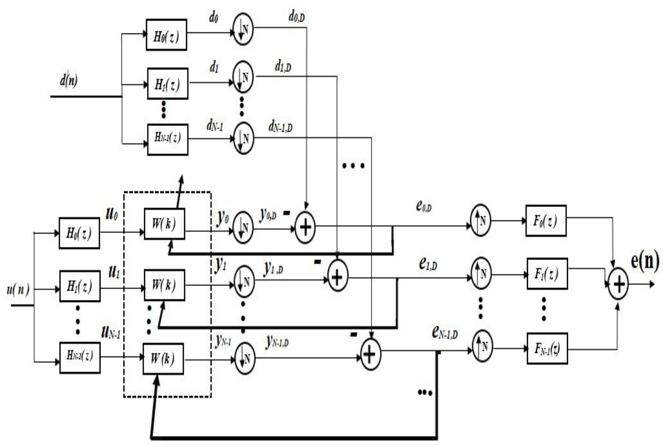


Figure 2: Multiband structured sub band adaptive filter (from Reference [10])

The two kinds of tree-structured filter banks are regular and irregular. In a regular sub band tree-structure, sub bands in each level are decomposed by the same filter banks where as in irregular sub band tree-structure, sub bands in each level are decomposed by different filter banks or only some sub bands are decomposed. The distinct advantage of the equivalent form of the tree-structured FB is the possibility of realizing analysis and synthesis filters for more important tasks than simple band separation [9]. Proposed tree-structured NFVBD with decimation factors (16,16,8,4,2) is illustrated is shown in Figure 3.

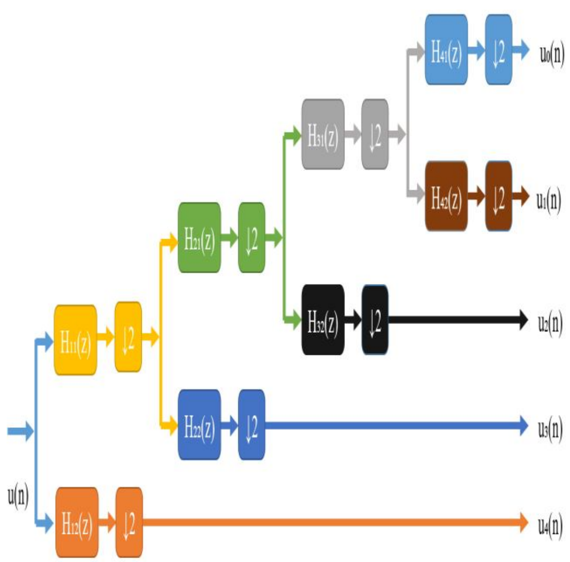


Figure 3: Proposed block diagram of tree structured NFVBD

The generalized structure of N-channel NFVBD based on tree structured approach having decimation $N_0, N_1, N_2 --- N_{N-1}$ for each band then the decimation factors must fulfill the accompanying condition [9]

$$\sum_{k=0}^{N-1} \frac{1}{N_k} = 1 \quad (2)$$

In N-channel NUFB the PR is possible if

$$\sum_{k=0}^{N-1} \left|H_k(e^{jw})\right|^2 = 1 \quad \text{for } 0 \le w \le \frac{\pi}{N} \quad (3)$$

Where $H_k(e^{jw})$ is frequency response of k^{th} filter in equivalent NUFB parallel form

For tree structured NUFB design having decimation factors (16,16,8,4,2) the PR condition can be achieved by using following equation

$$\left|H_0(e^{jw})\right|^2 + \left|H_1(e^{jw})\right|^2 + \left|H_2(e^{jw})\right|^2 + \left|H_3(e^{jw})\right|^2 + \left|H_4(e^{jw})\right|^2 = 1 \quad \text{for} \quad 0 \le w \le \frac{\pi}{5} \quad (4)$$

Here $H_0(z), H_1(z), H_2(z), H_3(z), H_4(z)$ analysis filters with the following relations

$$H_0(z) = H_{11}(z)H_{21}(z^2)H_{31}(z^4)H_{41}(z^8) \quad (5)$$
$$H_1(z) = H_{11}(z)H_{21}(z^2)H_{31}(z^4)H_{42}(z^8) \quad (6)$$
$$H_2(z) = H_{11}(z)H_{21}(z^2)H_{32}(z^4) \quad (7)$$
$$H_3(z) = H_{11}(z)H_{22}(z^2) \quad (8)$$
$$H_4(z) = H_{12}(z) \quad (9)$$

Where H_{11}, H_{21}, H_{31} and H_{41} are low pass filters in the first ,second, third and fourth stages respectively and H_{12}, H_{22}, H_{32} and H_{42} are high pass filters in the first ,second, third and fourth stages respectively. To evaluate performance of MSAF's various Uniform structures such as three band decomposition **(TBD)** with decimation factors (3,3,3), four band decomposition **(FBD)** with decimation factors (4,4,4,4) and five band decomposition **(FVBD)** with decimation factors (5,5,5,5,5) and non uniform structures with decimation factors (4,4,2) *i.e* three band decomposition (**NTBD**), non uniform structures with decimation factors (8,8,4,2 *i.e* four band decomposition (**NFBD**), non uniform structures with decimation factors decimation factors (16,16,8,4,2) 2 *i.e* five band decomposition (**NFVBD**) using proposed adaptive algorithms are implemented.

3. PROPOSED ALGORITHM FORMULATION

This basic principle is used to design a constraint optimization criterion for deriving the adaptive algorithm. In the MSAF shown in Figure.2 with the sub band signals sets $\{U(k), d_D(k)\}$,a criterion that ensures convergence to the optimum solution after a adequate number of iterations is to have the updated tap weight vector $w(k+1)$ in all N sub bands at each iteration $\boldsymbol{k}$ as follows:

Least Mean Square (LMS) Algorithm

$$w(k+1) = w(k) + \mu * u_i(k) * e_{i,D}(k) \quad (10)$$

Where μ is learning rate parameter and it should be selected in the stability bound *i.e* $0 < \mu < \frac{2}{N * P_u}$, here P_u is average power of the $u(n)$ calculated as $P_u = u^T(n)u(n)$.

Normalized Least Mean Square (NLMS) Algorithm

$$w(k+1) = w(k) + \mu \frac{u_i(k)}{\|u_i(k)\|^2 + \alpha} e_{i,D}(k) \quad (11)$$

Where α is a small positive constant used to avoid possible division by zero.

The computational complexity of the LMS algorithm is mainly due to multiplications performed in the coefficient updating and in the calculation of the adaptive-filter output. A first step to simplify the LMS algorithm is to apply quantization to the error signal, generating the quantized-error algorithm which updates the filter coefficients according to

$$w(k+1) = w(k) + \mu * u_i(k) * Q[e_{i,D}(k)] \quad (12)$$

Where Q[·] represents a quantization operation.

Walsh-Hadamard transformation codes are the most common fixed length orthogonal codes used in CDMA applications. A set of Walsh codes of length ‘Q’ consists of the Q rows of an Q by Q Walsh matrix [18]. The matrix is defined as follows

$$w1 = 1$$

$$\boldsymbol{w_{2Q}} = (w_Q \;\; w_Q;\; w_Q \; -w_Q); \quad (13)$$

The Walsh matrix has the property that every row is orthogonal to every other row,

$$\boldsymbol{w_2} = (0\;0; 0\;1)$$

$$\boldsymbol{w_4} = (0\;0\;0\;0;\; 0\;1\;0\;1;\; 0\;0\;1\;1;\; 0\;0\;1\;0)$$

he filter coefficients or weight vectors of the filter are changed for the next iteration for Modified Sign Regressor Normalized Least Mean Square (MSRNLMS) depending on ***w(k).*** Where ***w(k)*** is the impulse response of the individual subband adaptive filter transfer function is ***W(z)*** i.e the frequency response of low pass FIR filter with filter length 32,cut off frequency 100 Hz and sampling frequency 360 Hz. Here ***w(k)*** initial FIR filter numerator polynomial coefficients which are generated by using Walsh-Hadamard transformations matrix elements.These algorithms update the filter coefficients of adaptive filter using the following equation.

Modified Sign Regressor Normalized Least Mean Square (SRNLMS) Algorithm

$$w(k+1) = w(k) + \mu \frac{\boldsymbol{sgn(u_i(k))}}{\|u_i(k)\|^2 + \alpha} e_{i,D}(k) \quad (14)$$

where ***sgn*** is the well known signum, function i.e

$$sgn(x(n)) = \begin{cases} 1 & for\; x(n) > 0 \\ 0 & for\; x(n) = 0 \\ -1 & for\; x(n) < 0 \end{cases} \quad (15)$$

Sign based algorithms that make use of signum function (polarity) of either error or input signal or both have been derived from the Mean square error LMS algorithm.

4. SIMULATION RESULTS AND DISCUSSION

In this simulation the benchmark Massachusetts Institute of Technology-Beth Israel Hospital (MIT-BIH) arrhythmia database [16] ‘.mat’ file recordings (100m, 105m, 108m, 203m and 228m) were utilized to test the execution of various SAF algorithms for ECG denoising. Each record of 10 seconds duration is chosen. Digitization of recordings is done at a rate of 360 samples per second per channel with 11-bit resolution over a 10 mV range. The simulations were done on a sub set of 3600 samples of 10 seconds duration ECG recordings. In order to test the filtering capability in non-stationary environment a synthetic PLI (Low amplitude synthetic PLI generated with frequency of 60Hz) noise with 3600 samples are considered, which are obtained from MIT-BIH normal sinus rhythm database (NSTDB) [17]. Both the signals should be re sampled at same sampling rate for ease of processing. The proposed SAF-ANC are compared quantitatively by quality assessment parameters mean square error (MSE), root mean square error (RMSE),signal-to-noise ratio before filtering (SNRBF), SNR after filtering (SNRAF) and distortion are outlined in Table.1.

Table 1: Quality assessment parameters

PARAMETERS	FORMULA
Mean Square Error	$\frac{1}{N}\sum_{n=0}^{N-1}[s(n)-e(n)]^2$
Root M.S.E	$\sqrt{\frac{1}{N}\sum_{n=0}^{N-1}[s(n)-e(n)]^2}$
SNRBF	$10\log_{10}(\frac{\sum_{n=0}^{N-1}[s(n)]^2}{\sum_{n=0}^{N-1}[d(n)-s(n)]^2})$
SNRAF	$10\log_{10}(\frac{\sum_{n=0}^{N-1}[s(n)]^2}{\sum_{n=0}^{N-1}[e(n)-s(n)]^2})$

Table 2, Table 3, Table 4 and Table 5 shows the performance of the various artifact cancellation using proposed algorithm is assessed by SNR, MSE, RMSE and distortion improvement values for entire dataset.

A) Power-line Interference (PLI) Cancellation

Table 2: SNR (in dB) obtained using MSRNLMS-SAF algorithm for PLI noise cancellation

Type of FB	Rec.No. 100m	Rec.No. 105m	Rec.No. 108m	Rec.No. 203m	Rec.No. 228m	Avg.SNR
Before Filtering	0.1971	0.6946	-0.0232	2.9497	-1.0706	**0.54952**
TBD	13.0304	13.2868	12.8584	14.7069	12.4	**13.2565**
FBD	15.4276	15.5955	15.3032	16.6857	15.1097	**15.62434**
FVBD	16.3768	16.4494	16.2061	17.2153	16.0008	**16.44968**
NTBD	17.0768	17.3157	16.9559	18.4295	16.6901	**17.2936**
NFBD	17.5926	18.4099	17.4528	19.9567	17.0615	**18.0947**
NFVBD	**22.2727**	**21.7628**	**22.3112**	**23.3244**	**21.3654**	**22.2073**

In Table 2 the average SNR values of proposed ANC are shown, for PLI Noise cancellation using MSRNLMS adaptive algorithm, where input SNR before filtering is 0.54952 dB. As can be seen from the Table 2 the average SNR values of 5 records are **13.2565 dB**, **15.62434 dB**, **16.44968 dB**, **17.2936 dB**, **18.0947 dB** and **22.2073 dB** using TBD,FBD,FVBD,NTBD,NFBD and NFVBD structured SAF’s-SRNLMS Algorithm. From the table the maximum average SNR value in dB obtained is 22.2073 for NFVBD structured SAF using SRNLMS algorithm.

Table 3: MSE Values obtained using MSRNLMS-SAF algorithm

Type of FB	Rec.No.1 00m	Rec.No.1 05m	Rec.No.1 08m	Rec.No.2 03m	Rec.No.2 28m	Avg. MSE
TBD	0.066	0.1905	0.0634	0.4641	0.0749	**0.17178**
FBD	0.0854	0.209	0.0812	0.4755	0.092	**0.18862**
FVBD	0.0945	0.2185	0.0905	0.4877	0.1012	**0.19848**
NTBD	0.1053	0.2741	0.099	0.6339	0.1133	**0.24512**
NFBD	0.1911	0.4998	0.1854	1.0678	0.1984	**0.4285**
NFVBD	**1.0996**	**1.2499**	**1.1624**	**2.4397**	**0.971**	**1.38452**

Table 4: **R**MSE Values obtained using MSRNLMS-SAF algorithm

Type of FB	Rec.No.1 00m	Rec.No.1 05m	Rec.No.1 08m	Rec.No.2 03m	Rec.No.2 28m	Avg.R MSE
TBD	0.2568	0.4365	0.2517	0.6812	0.2737	**0.37998**
FBD	0.2922	0.4572	0.2849	0.6895	0.3033	**0.40542**
FVBD	0.3074	0.4675	0.3008	0.6983	0.3181	**0.41842**
NTBD	0.3245	0.5236	0.3146	0.7962	0.3367	**0.45912**
NFBD	0.4371	0.707	0.4305	1.0334	0.4454	**0.61068**
NFVBD	**1.0486**	**1.118**	**1.0781**	**1.5619**	**0.9854**	**1.1584**

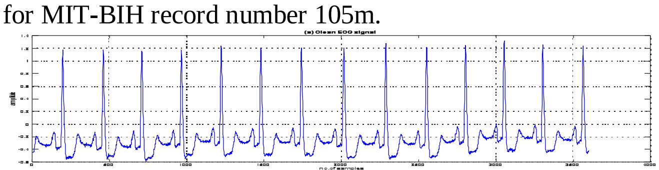
Table 3 and Table 4 compare the performance of proposed methods, in terms of M.S.E and R.M.S.E. In From the Table 3 and Table 4 the average M.S.E and R.M.S.E values obtained by using TBD-MSRNLMS adaptive algorithm is minimum.

Table 5: Distortion Values obtained using MSRNLMS-SAF algorithm

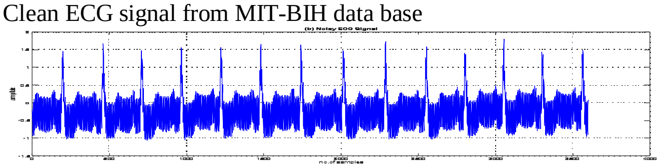
Type of FB	Rec.No. 100m	Rec.No. 105m	Rec.No. 108m	Rec.No. 203m	Rec.No. 228m	Avg.Dist ortion
TBD	-11.8065	-7.2008	-11.9813	-3.3342	-11.255	**-9.11556**
FBD	-10.6859	-6.7987	-10.9068	-3.2289	-10.3614	**-8.39634**
FVBD	-10.245	-6.6047	-10.4341	-3.1187	-9.9495	**-8.0704**
NTBD	-9.7753	-5.6206	-10.0449	-1.98	-9.4564	**-7.37544**
NFBD	-7.188	-3.0121	-7.3198	-0.285	-7.0248	**-4.96594**
NFVBD	**0.4122**	**0.9686**	**0.6535**	**3.8733**	**-0.1279**	**1.15594**

The average Distortion values are plotted in Table.5. It is also evident that the average Distortion for TBD structured SAF's was found to be minimum than other structured SAF's. From the TBD structured SAF which has minimum average Distortion -9.11556 dB.

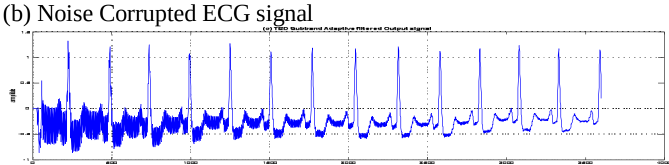
The PLI noise cancellation using SAF is shown in Figure.5, for MIT-BIH record number 105m.



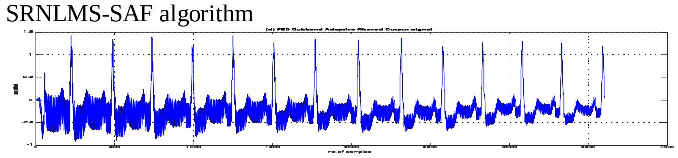
(a) Clean ECG signal from MIT-BIH data base



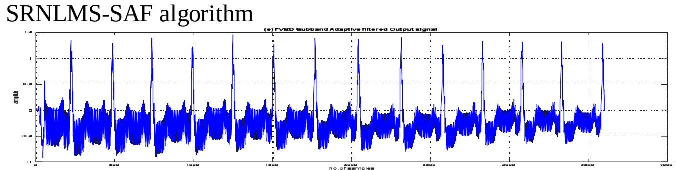
(b) Noise Corrupted ECG signal



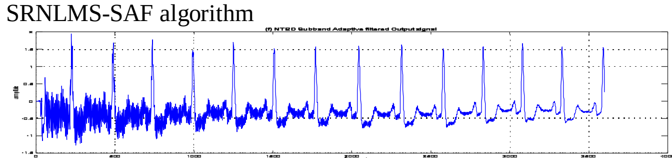
© Sub band adaptive filtered signal using TBD structured SRNLMS-SAF algorithm



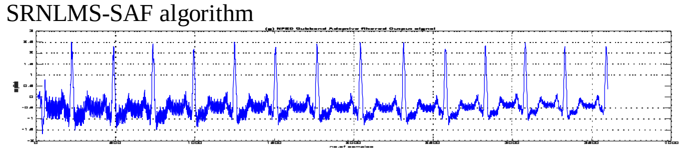
(d) Sub band adaptive filtered signal using FBD structured SRNLMS-SAF algorithm



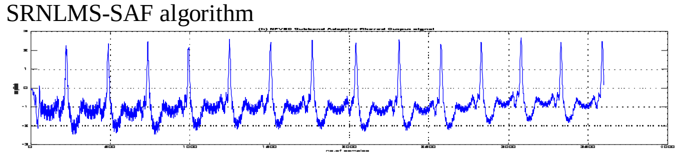
(e) Sub band adaptive filtered signal using FVBD structured SRNLMS-SAF algorithm



(f) Sub band adaptive filtered signal using NTBD structured SRNLMS-SAF algorithm



(g) Sub band adaptive filtered signal using NFBD structured SRNLMS-SAF algorithm



(h) Sub band adaptive filtered signal using NFVTBD structured SRNLMS-SAF algorithm

Figure 5 Adaptive filtering simulation results of PLI noise cancellation (a) ECG record number 105m from database (b) MIT-BIH record 105m with PLI noise (c,d,e,f,g,h) Sub band adaptive filtered signal using TBD,FBD,FVBD,NTBD,NFBD and NFVBD structured MSRNLMS-SAF algorithm.

5. CONCLUSION

This article presents the new Hybrid denoising of ECG signal with power dependent on subband deterioration organized MSAF's utilizing SRNLMS versatile calculation. The investigation of subband organized SAF framework is completed and reenactments are performed utilizing MATLAB. So as to examine the exhibition of the proposed plan an examination has been made between six distinctive ECG denoising plans i.e TBD,FBD,FVBD,NTBD,NFBD and NFVBD organized SAFs utilizing MSRNLMS versatile calculation. The proposed HSAF-ANC framework comprises of reference sign and essential sign as information boundaries for which versatile filtered evaluated signal, mistake sign and coefficients of the filter are acquired as yield boundaries. Better filtering execution results are gotten by NFVBD organized SAF.

Contributions: The main contributions of my work is to perform pre processing of ECG signal and to remove PLI noise from signal based on Subband decomposition using MSRNLMS adaptive algorithm.

Source of Funding: “This research received no external funding”

Acknowledgement: The authors would like to thank Administrative, University staff, Teaching and Non Teaching faculty members of University College of Engineering Technology Acharya Nagarjuna University, Guntur, India for cordially giving opportunity to do research work.

Conflicts of Interest
“The authors declare no conflict of interest.”

REFERENCES

1. Catalano, Joseph T. Guide to ECG analysis. Lippincott Williams Wilkins, 2002.
2. Haykin, Simon S. Adaptive filter theory. Pearson Education India, 2008.
3. Kuo, Sen M., and Dennis Morgan. Active noise control systems: algorithms and DSP implementations. John Wiley Sons, Inc., 1995.
4. Sankar, A. Bhavani, D. Kumar, and K. Seethalakshmi. ” Performance study of various adaptive filter algorithms for noise cancellation in respiratory signals.” Signal processing: An international journal (SPIJ) 4.5 (2010): 267.
5. Rahman, Muhammad Zia Ur, Rafi AhamedShaik, and DV Rama Koti Reddy. "Efficient sign based normalized adaptive filtering techniques for cancelation of artifacts in ECG signals: Application to wireless biotelemetry." Signal Processing 91.2 (2011): 225-239.
6. Huang, Hsu-Chang, and Junghsi Lee. “A new variable step-size NLMS algorithm and its performance analysis.” IEEE Transactions on Signal Processing 60.4 (2012): 2055-2060.
7. Gowri, Thumbur, P.RajeshKumar, and DV Rama Koti Reddy."An efficient variable step size Least Mean Square Adaptive Algorithm used to enhance the quality of electrocardiogram signal." Advances in Signal Processing and Intelligent Recognition Systems. Springer, Cham, 2014. 463-475.
https://doi.org/10.1007/978-3-319-04960-1_41
8. Bhati, Ravina, et al. "Noise Suppression From ECG Signal Using Adaptive Filter Technique." International Journal of Computer Science and Information Security 14.8 (2016): 135.
9. Vaidyanathan, Parishwad P. Multirate systems and filter banks. Pearson Education India, 1993.
10. Lee, Kong-Aik, Woon-Seng Gan, and Sen M. Kuo. Subband adaptive filtering: theory and implementation. John Wiley & Sons, 2009.
11. Shin, JaeWook, JinWooYoo, and PooGyeon Park. "Variable step-size sign subband adaptive filter." IEEE Signal processing letters 20.2 (2013): 173-176.
12. Seo, Ji-Hye, and PooGyeon Park. "Variable individual step-size subband adaptive filtering algorithm." Electronics Letters50.3 (2014): 177-178.
13. Kumar, A., G. K. Singh, and S. Anurag. "An optimized cosine-modulated non uniform filter bank design for subband coding of ECG signal." Journal of King Saud University-Engineering Sciences 27.2 (2015): 158-169.
14. Yu, Yi, Haiquan Zhao, and Badong Chen. "A new normalized subband adaptive filter algorithm with individual variable step sizes." Circuits, Systems, and Signal Processing 35.4 (2016): 1407-1418.
15. Abadi, Mohammad Shams Esfand, Mohammad Saeed Shafiee, and MehrdadZalaghi. "A low computational complexity normalized subband adaptive filter algorithm employing signed regressor of input signal." EURASIP Journal on Advances in Signal Processing 2018.1 (2018): 21
16. Mark, R., and G. Moody. “MIT-BIH arrhythmia database directory.” Cambridge: Massachusetts Institute of Technology (1988).
17. Goldberger, A. L., et al. ” The MIT-BIH normal sinus rhythm database.” Online
18. Wysocki, B. J., and Tadeusz A. Wysocki. "Modified Walsh□Hadamard sequences for DS CDMA wireless systems." International Journal of Adaptive Control and Signal Processing 16.8 (2002): 589-602.
19. Seok-Woo Jang, Sang-Hong Lee “ Classification of Epileptic Seizure EEG Based on Fully Connected Neural Network”, International Journal of Emerging Trends in Engineering Research, ISSN 2347 – 3983, Volume 8. No. 7, July 2020
https://doi.org/10.30534/ijeter/2020/21872020
20. Saravana Selvan , Umayal “RF Powered ECG Monitoring System for Silent Ischemia Patient”, International Journal of Emerging Trends in Engineering Research, ISSN 2347 – 3983, Volume 8. No. 7, July 2020