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Adaptive Noise Cancellation Using Kalman Filter for Non-Stationary Signals

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Abstract. The present paper states with the Adaptive Noise Cancellation (ANC) of speech signal corrupted with additive Gaussian white noise. A new method is proposed based on adaptive Kalman filtering. The probabilistic approach of kalman filters over a packet-delaying network given the postpone distribution and decide the minimum required buffer length and algorithm has been used for estimation of unknown state variables within the system. Any work has been achieved for structures with fractional-order dynamics. The new techniques based totally at the kalman filter proposed within the beyond, function in steps: first the noise variance and the parameters of the signal version is estimated and secondly the speech signal expected. The strategies provided inside the paper gives an exclusive method consequently it does not require estimation of the noise variance. The noise variance estimation is always will become part of the kalman advantage calculation. The results of the application of Kalman filter on a non-stationary acoustic signal indicated that SNR of the ~ 1.17 dB and MSE`0.032 can be achievable using Kalman filters and Kalman filter can be efficiently used for noise cancellation in place of other adoptive filters.

Key Word: Adaptive filters, Adaptive Noise Cancellation (ANC), Gaussian noise, Kalman filter, Signal to Noise Ratio (SNR),

1. Introduction

The overall performance of any speech sign processing gadget can be degraded within the presence of noise (either additive or convolution). Numerous techniques for extracting the clatter from a speech waveform were studied. Most of these techniques are based upon the concept of adaptive filtering that seeks benefits from quasi-periodic speech signals which acts as reference for adaptive filter. The designed clear out has been applied on non -stationary speech alerts to have a look at its noise suppression capability measured because the distinction among enter signal snr and filter output sign SNR and the Mean Square Error (MSE). The efficiency of every filter out for suppression of noise in SNR(dB) is decided and in comparison to

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assess the relative performance of numerous adoptive filters consisting of kalman filter out which has precise applications

The techniques even enhanced the output of linear prediction analysis. It makes the noise separation technique have a huge frequency variety and aren't without problems separated from noise the use of filtering strategies. Application of Kalman filter is the reduction of noise in communication channel. When it is non-stationary, recursive filter allows estimation of the useful signal in noise from time series [1],

A noise itself is a facts bearing sign that conveys facts concerning the sources of the noise and the surroundings wherein it propagates. Cosmic radiation presents data on formation and form of the universe and historical past speech conversations in a crowded venue can represent interference [2]. If the speech signal is considered as an output of a system adaptive kalman filter is used to estimate the speech sign. [3-4]. SNR optimization benefit procedure. So, this new approach appears very appealing in comparison to the earlier strategies [5]. For linear but time variation structures the kalman clear out is based on a kingdom area method of a non-stop or discrete time device. The system ought to be best considered in discrete time. The kalman filter will gives the estimate of the system given fast of outputs. It will additionally minimize the output errors of the clear out [6]. Many approaches the use of kalman filtering had the typically perform in steps: first, the noise and riding method variances and parameters of speech model are predicted; then, the speech sign is anticipated with the aid of kalman filtering. In fact, these strategies fluctuate best by the selection of the set of rules used to estimate model parameters and the selection of the models followed for the speech sign and the additive noise [7]. The signal it will be presence of noise degraded within (either additive convolution).it creatively due to the acoustic mismatch between the speech functions used to teach and take a look at this gadget and the capability of the acoustic fashions to describe the corrupted speech. [8-9]. the factors had been organized in a matrix technique so that the favored signal from the noisy can be separated. The evaluation will outcomes in designed kalman filters showing that it can correctly dispose of the noises [10]. In the present paper the kalman clear out-which counts as one of the notable filters- has been surveyed whose factors is being calculated to design an efficient filter [11-13]. Initially a pattern signal is randomly decided on which may be similar to an auto regressive sign. Then random Gaussian noise is implemented on auto regressive signal; and therefore the noisy sign is suppressed. A Mat lab software simulation has been carried out and the results are presented.

2. Kalman Filter

The kalman filter has played a critical function in systems concept and has observed extensive programs in lots of fields consisting of signal processing. The probabilistic technique of kalman filters over a packet-delaying community given the postpone distribution and decide the minimal required buffer length. The algorithm has been used for estimation of unknown country variables inside the device and work has been executed for structures with fractional-order dynamics

The work is primarily based on monitoring best a limited wide variety of strongest interference's assumptions of synchronous interferes operation with overlapping however extraordinary schooling alerts. Kalman filtering is used for interfering users channel estimation and following calculation of interference correlation matrix. Such in-time correlation matrix estimate exploited in MMSE primarily based developed algorithms may be utilized in subsequent technology

$$x(n+1) = \Phi(n+1,n).\mathrm{X(n)} + \mathrm{V_1(n)} \tag{1}$$

$$K(n) = \mathrm{K(n,n-1)} - \Phi(n+1,n).G(n).c(n).K(n,n-1)$$

$$K(n+1,n) = \Phi(n+1,n).K(n).\Phi^{\mathrm{H}}(n+1,n) + Q_1(n) \tag{2}$$

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The Kalman filter gives the solution to the following problem. Given the state space model in equation (3) or figure 1 where **Q1** (n) , **Q2** (n) , **C**(n) , $\Phi(n+1,n)$ and **y**(n) are known quantities, find the best estimate $\hat{X}(\frac{n}{Y^n})$ to the state vector **x**(n) , in the expected least errors squares sense.

$$Y(n) = \mathrm{C(n).X(n) + V_2(n)} \tag{3}$$

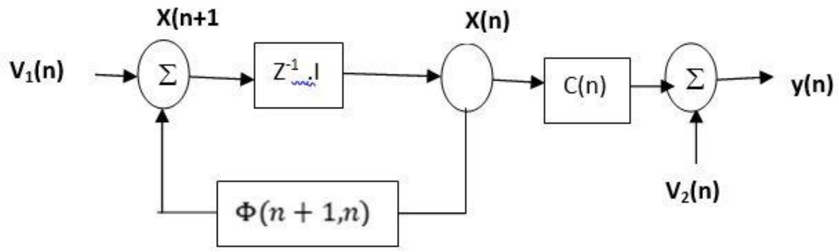

Figure 1. Space state model used for the system in the Kalman filter.

$$E[V_1(n).V_1^H(n)] = Q_1(n), E[V_2(n), E[V_2(n).V_2^H(n)] = Q_2(n) \tag{4}$$

$$G(n) = \Phi(n+1,n).K(n,n-1).C^H(n)[(n).K(n,n-1).C^H(n) + Q_2(n)]^{-1} \tag{5}$$

$$\alpha(n) = y(n) - C(n).\hat{X}(\frac{n}{Y_{n-1}}) \tag{6}$$

$$\hat{x}\left(\frac{n+1}{y_n}\right) = \Phi(n+1,n).\hat{x}\left(\frac{n}{y_{n-1}}\right) + G(n).\alpha(n) \tag{7}$$

$$\hat{x}\left(\frac{n}{y_n}\right) = \Phi(n,n+1).\hat{x}\left(\frac{n+1}{y_n}\right)$$

$$K(n) = \mathrm{K(n,n-1)} - \Phi(n+1,n).G(n).c(n).K(n,n-1)$$

$$K(n+1,n) = \Phi(n+1,n).K(n).\Phi^{\mathrm{H}}(n+1,n) + Q_1(n) \tag{8}$$

There has been much interest in fast convergence algorithms, but is fast convergence really needed in ANC. Fast convergence means that the time the algorithm takes from its initialization to the point it reaches an optimal value is short. These algorithms are of great interest in telecommunications system where the goal is to reduce the size of the training sequence and the Corresponding overhead.

3. Kalman Filter Automatic Noise Cancellation (KFANC)

Acoustic noise cancellation ANC is best suited to remove ambient noise. The conventional algorithms based on ANC had superior performance at low frequency bands and with increase in frequency and bandwidth the performance disorients. Generally, the noises that affect the system have wider frequency range and only a small amount of energy is not present near low frequency region have relatively high frequency components. When ANC is cascaded with different methods. The frequency dependent noise cancellation will discard any effect on speech signal. However, the ANC systems fall back in performance when noise frequency level goes high. The noise in real world has higher broadband and high frequency component.

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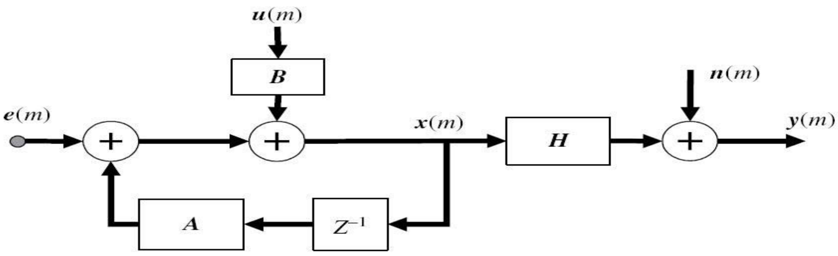

Figure 2. The Block diagram of Kalman filter

The use of this method makes it feasible to evaluate a sign transmitted through manner of a distorted channel, even as gaining a few noises at the identical time (2-five). Formula of this clear out, in order that it could dispose of the noise and distortion, is based totally on country space. In this regard, x(m) vector is taken to expose the favored vector and y(m) vector indicates the output noisy vector. Equations 1 and pair of display the vectors of the country area and output filter.

$$x(m) = A\, x(m-1) + B\, u(m) + e(m) \tag{9}$$

$$y(m) = H\, x(m) + n(m) \tag{10}$$

U (m) is the P-dimensional control input;

E (m) is the P-dimensional equation,

Figure.2 is the block diagram Shows Kalman filter Vectors is showed in this figure Y (m) and X (m).

4. Kalman Filter Algorithm

Step 1: Read the input signal (speech signal) using wave read i.e. input signal

Step 2: Parameters of input signal are Fs = 8000, number of bits encoded is 16bits (39500 samples = 39500/8000 = x seconds)

Step 3: Generate noise signal of length of the input signal using Gaussian white noise i.e. noise

Step4: Calculate input SNR in DB

Step 5: Add input signal with noise signal to generate a estimated signal i.e. input_signal+noise = Estimated signal

Step 6:

(i) Calculate AR parameters for both input signal and noise signal of 16 co-efficient using Burg's algorithm

(ii) Give this AR parameter to unidirectional non-stationary Kalman filter, output of this function is output signal

Step7: Now calculate output SNR in DB

Step8: Calculate mean square error (MSE)

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5. Result and Methods

The input non stationary acoustic input signal chosen for the present analysis is shown in figure 3.

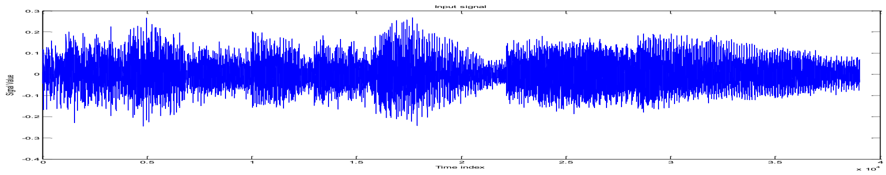

Figure 3. Input non stationary acoustic signal

It can be seen from the figure 3 that the signal is also associated with minimum amount of noise.
Further the acoustic input signal is combined with the white Gaussian noise and the estimated input signal is shown in figure 4.

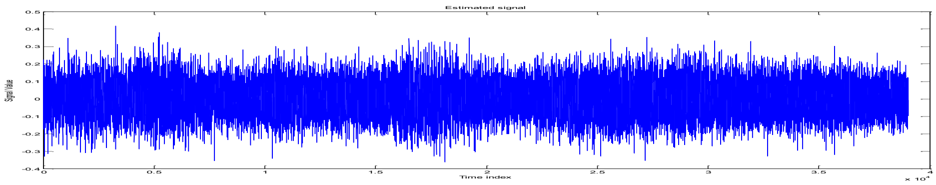

Figure 4. Estimated signal (input signal + Generate noise = ES) signal

It can be seen from the figure 4 that the signal is almost merged within noise The estimated input signal generated with by adding noise to the signal (ES=Signal + Noise). The SNR for the estimated input signal is determined is ~87dB. The estimated input signal is passed through Kalman filter for the cancellation of noise.
The output signal derived after the Kalman filter using the algorithms mentioned above is shown in figure 5.

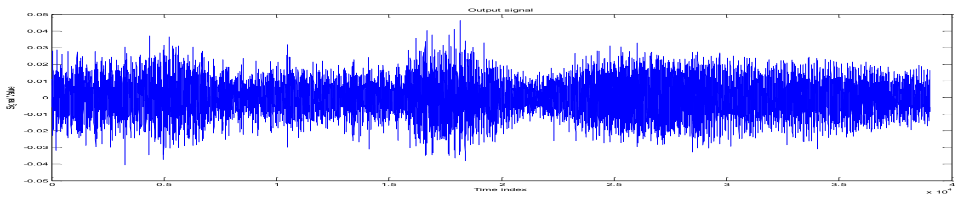

Figure 5. Output signal for Kalman filter

It is evident from the figure 5 that the noise cancellation in the Kalman filter output Signal is very substantial. The SNR Determined form the Kalman filter signal output ~ 69 dB. It is indicated that the noise reduction by the Kalman filter is very significant and SNR ~ 1.16 dB

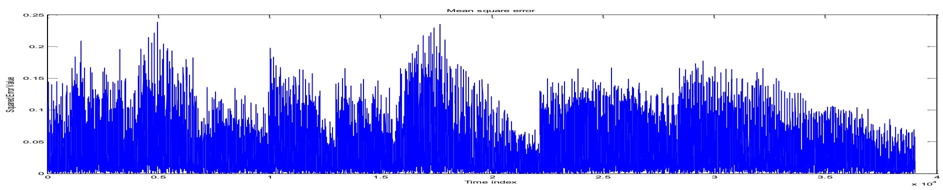

Figure 6. MSE of output Kalman filter signal

The mean square error distribution derived for the acoustic signal using Kalman filter is shown in figure 6. The MSE for the acoustic signal is found to be ~ 0.032 suggesting that the filtering process in very efficient.

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The noise suppression for real time signal using the Reference of few paper of Kalman filters in found to be SNR ~0.2 dB and MSE~0.0030 for the non-stationary speech signal. The new method of the adoptive Kalman filter techniques are effective in noise suppression can be used only for specific applications in signal processing is show in table 1.

Table 1. Results of Non-Stationary Kalman filter

signal	SNR before(dB)	SNR after(dB)	Noise Suppression (dB)	MSE
Real time signal	-87.4679	86.2998	1.1681	0.0032

The result of the Kalman filtering of the non-stationary acoustic signal is summarized in table 1. It is clearly seen from the table that the noise cancellation ~1.17 dB using the Kalman filter for the non-stationary acoustic signal input. The results indicate that the Kalman filter techniques are very efficient in noise cancellation and can be effectively used in acoustic signal processing.

The noise suppression for real time signal using the Kalman filters in found to be SNR ~0.2 dB and MSE~0.0030 for the non-stationary speech signal. The adoptive Kalman filter techniques are effective in noise suppression can be used only for specific applications in signal processing.

6. Conclusions

From the results presented above the following conclusions can be drawn

1. The Kalman filter is found to be very efficient can effectively replace the adoptive filters for noise cancellation in non- stationary signals.

2. The noise cancellation is found to be very significant~ 1.17 dB and MSE ~ 0.032 are achievable for the acoustic non stationary signals.

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