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Statistical Evaluation In Speech Processing Techniques

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ABSTRACT

The objective of this paper is to generate a reconstructed output speech signal from the input signal involving the application of a filter estimation technique. In this paper, filter is used to estimate the parameters represented in the state-space domain, in which the speech signal is modeled as such.

The results of this speech coding technique are demonstrated and obtained with the help of MATLAB. Accurate estimations by the filter on speech is simulated and presented. Comparison between the assigned values and the estimated values of the parameters is described. The reconstruction of speech and the input speech is also compared for error estimations.

Key words: filter, speech coding technique.

1. INTRODUCTION

Speech processing has been a growing and dynamic field for more than two decades and there is every indication that this growth will continue and even accelerate. During this growth there has been a close relationship between the development of new algorithms and theoretical results, new filtering techniques are also of consideration to the success of speech processing. One of the common adaptive filtering techniques that are applied to speech is the Wiener filter. This filter is capable of estimating errors however at only very slow computations. On the other hand, the Kalman filter suppresses this disadvantage.

According to **[1]**, Kalman filter is so popular in the field of radar tracking and navigating system is that it is an optimal estimator, which provides very accurate estimation of the position of either airborne objects or shipping vessels. Due to its accurate estimation characteristic, electrical engineers are picturing the Kalman filter as a design tool for speech, whereby it can estimate and resolve errors that are contained in speech after passing through a distorted channel. Due to this motivating fact, there are many ways a Kalman filter can be tuned to suit engineering

Applications such as network telephony and even satellite phone conferencing.

II. BACKGROUND

The term speech processing basically refers to the scientific discipline concerning the analysis and processing of speech signals in order to achieve the best benefit in various practical scenarios **[2]**. The field of speech processing is, at present, undergoing a rapid growth in terms of both performance and applications. This is stimulated by the advances being made in the field of microelectronics, computation and algorithm design **[3]**.

Sampling

The purpose of sampling is to transform an analog signal that is continuous in time to a sequence of samples discrete in time. The signals we use in the real world, such as our voices, are called "analog" signals. In order to process these signals in computers, most importantly it must be converted to "digital" form. While an analog signal is continuous in both time and amplitude, a digital signal is discrete in both time and amplitude. Since in this thesis, speech will be processed through a discrete Kalman filter, it is necessary for converting the speech signal from continuous time to discrete time, hence this process is described as sampling.

The value of the signal is measured at certain intervals in time. Each measurement is referred to as a sample. Once the continuous analog speech signal is sampled at a frequency **f**, the resulting discrete signal will have more frequency components than the analog signal. To be precise, the frequency components of the analog signal are repeated at the sample rate. Explicitly, in the discrete frequency response they are seen at their original position, and also centered around +/- **f**, and +/- 2 **f**, etc.

The signal still preserve the information, it is necessary to sample at a higher rate greater than twice the maximum frequency of the signal. This is known as the Nyquist rate. The Sampling Theorem states that a signal can be exactly reconstructed if it is sampled at a frequency **f**, where **f** > 2fm where fm is maximum frequency in the signal.

III. PROPOSED METHODOLOGY

From a statistical point of view, many signals such as speech exhibit large amounts of correlation. From the perspective of coding or filtering, this correlation can be put to good use [6]. The all pole, or autoregressive (AR), signal model is often used for speech. From [4], the AR signal model is introduced as:

$yk = a_1 y_{k-1} + a_2 y_{k-2} + \ldots\ldots + a_N y_{k-N} + w_k$ (3.1)

where,
k = Number of iterations;
y_k= current input speech signal sample;
yk=$(N-1)^{th}$ sample of speech signal;
a_N =N^{th} Kalman filter coefficient;
and w_k = excitation sequence (white noise).

In order to apply Kalman filtering to the speech expression shown above, it mustbe expressed in state space form as

$H_k = X\, H_{k-1} + W_k$ (3.2)

$y_k = g\, H_k$ (3.3)

X is the system matrix, H_k consists of the series of speech samples; W_k is the excitation vector and g, the output vector. The reason of (k-N+1)th iteration is due to the state vector, $H_{k,}$ consists of N samples, from the k^{th} iteration back to the$(k-N+1)^{th}$ iteration.

The above formulations are suitable for the Kalman filter. As mentioned in the previous chapter, the Kalman filter functions in a looping method. Referring to [5] as a guide in implementing Kalman filter to speech, we denote the following steps within the loop of the filter. Define matrix $H^T{}_{k-1}$ as the row vector:

$H^T k\text{-}1 = -[y_{k-1}\ \ y_{k-2} \ldots y_{k-N]}$ (3.4)

and zk = yk. Then (3.1) and (3.4) yield

$z_k = H^T{}_{k-1}\, X_k + w_k$ (3.5)

where X_k will always be updated according to the number of iterations, k. when the k = 0, the matrix H_{k-1} is unable to be determined. However, when the time z_k is detected, the value in matrix H_{k-1} is known. The above purpose is thus sufficient enough for defining the Kalman filter, which consists of:

$X_k = [\, I - K_k H^T{}_{k-1}]\, X_{k-1} + K_k\, z_k$ (3.6)

Where I= Identity matrixwith

$K_k = P_{k-1} H_{k-1}[\, H^T{}_{k-1} P_{k-1} H_{k-1} + R\,]^{-1}$ (3.7)

where K_k is the Kalman gain matrix, P_{k-1} is the a priori error covariance matrix,R is measurement noise covariance.and

$P_k = P_{k-1} - P_{k-1} H_{k-1}\, [H^T{}_{k-1}\, P_{k-1} H_{k-1} + R]^{-1}.H^T{}_{k-1} P_{k-1} + Q$

where P_k is the a posteriori error covariance matrix, and Q is identity matrix.

Thereafter the reconstructed speech signal, **Y**k after Kalman filtering will be formed in a manner similar to (3.1):

$\mathbf{Y}_k = \mathbf{a}_1 \mathbf{Y}_{k-1} + \mathbf{a}_2 \mathbf{Y}_{k-2} + \ldots\ldots + \mathbf{a}_N \mathbf{Y}_{k-N} + \mathbf{w}_k$ (3.9)

Since the value of **Yk** is the input at the beginning of the process, there will be no problem forming $\mathbf{H^T{}_{k-1}}$. In that case a question rises, how is $\mathbf{Y}_k$ formed? The parameters $\mathbf{w}_k$ and $\{a_i\}_{i=1}{}^N$are determined from application of the Kalman filter to the input speech signal $\mathbf{Y_k}$. That is in order to construct $\mathbf{Y}_k$, we will need matrix **X** that contains the Kalman coefficients and the white noise, $\mathbf{w}_k$ which both are obtained from the estimation of the input signal. This information is enough to determine$\mathbf{HH_{k-1}}$.

Where

$$\mathbf{HH_{k-1}} = \{ \begin{matrix} Yk - 1 \\ Yk - 2 \\ Yk - N + 1 \end{matrix} \} \qquad 3.10$$

Thus, forming the equation (3.10) mentioned above.

IV. RESULTS

This approach is to prove that Kalman filter functions properly in MATLAB 7. Reconstructed accurate sound output are shown in fig.4.1.The results shown below from Fig 4.2 to Fig 4.6, the Kalman filter generates a very close set of coefficients, which are similar to those, selected above in (4.2). From the following results, accurate estimation of the coefficients took several hundreds of iterations in order to stabilize to the required value. The reason for this is that Kalman filter works in a loop fashion. In order to obtain a more accurate estimate, it will need to go through more "predict" and "correct" procedures

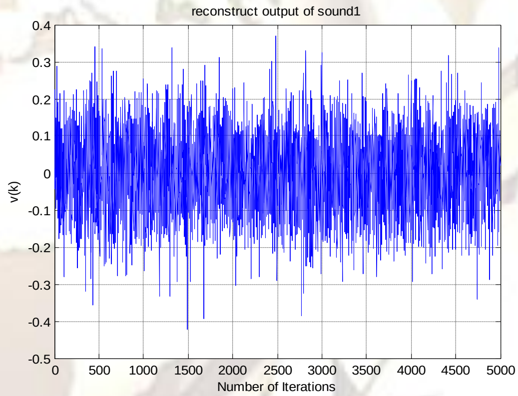

Fig. 4.1 Output of reconstructed signal

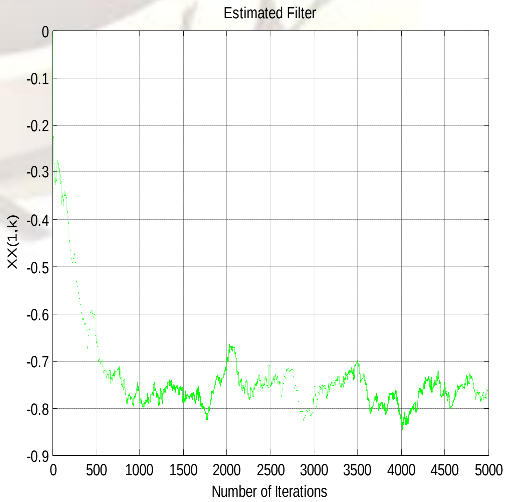

Fig. 4.2 First coefficient of filter

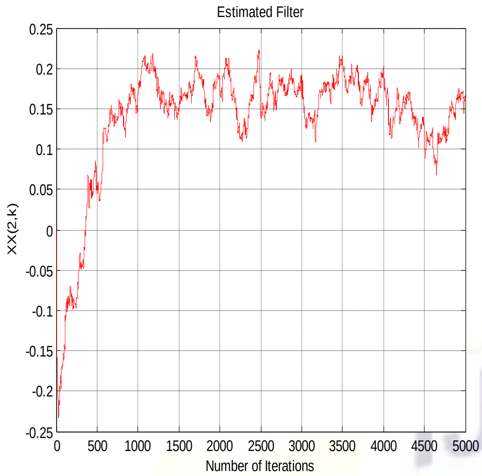

Fig. 4.3 Second coefficient of filter

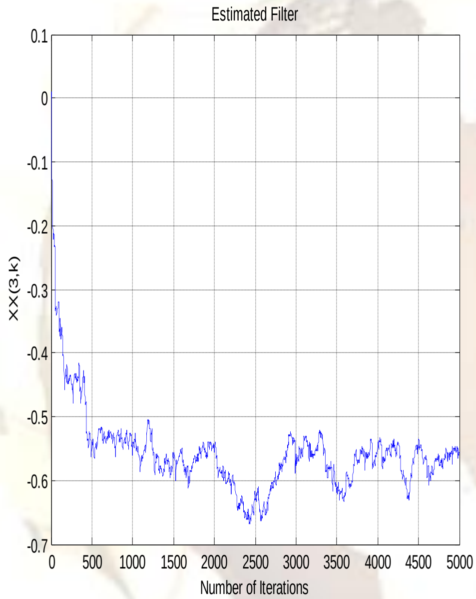

Fig. 4.4 Third coefficient of filter

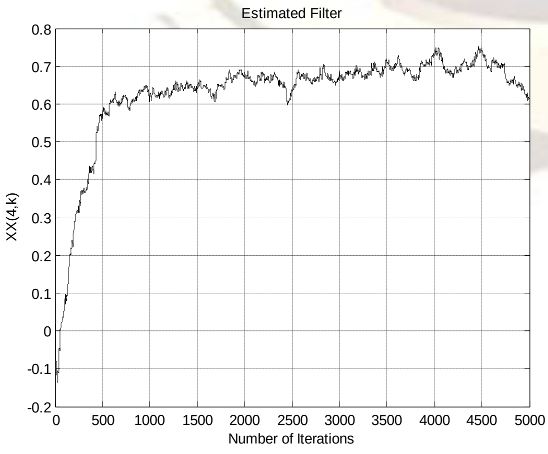

Fig. 4.5 Fourth coefficient of filter

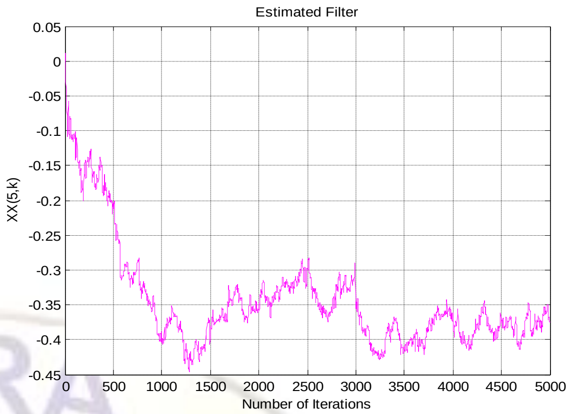

Fig. 4.6 Fifth coefficient of filter

V. CONCLUSION

In this paper, an implementation of employing Kalman filtering to speech processing had been developed. As has been previously mentioned, the purpose of this approach is to reconstruct an output speech signal by making use of the accurate estimating ability of the Kalman filter. Simulated results from the previous section had proven that the Kalman filter has the ability to estimate accurately.

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