Quality Improvement Over Speech Signal with Versatile Algorithm Based Kalman Filter

Misbah Uddin Mahammad[1], Subodh Panda[2], Srinivas Akula[3], B.N.Srinivasarao[4]
Assistant professor, University College of Engineering, Osmania University, Hydrabad[1], Professor[2] , Associate Professor[3,4], Pragati Engineering College Surampalem, Kakinada, AP[2,3,4]

Abstract.

The voice signal improvement has remarkable contribution over noise affected corrupted speech signal. Restructuring of a Kalman algorithm utilized for the noisily speaking signal is primary concept on process of speech improvement technique. Parameters of AR (auto-regressive) mode need to be calculated along with traditional Kalman filter for voicing signal improvement, deal with complex matrix operations which practically is a non-versatile. The proposed voicing improvement algorithm in this research work basically avoid the matrix calculations and correspondingly decreases the calculation time through regular update of the primary estimation for state direction x(n) . A multiplication element to the versatile filtrate is associated, for change the evaluation of surrounding unwanted signal with perception information. The outcomes are compared between traditional Kalman filtering and Fast Versatile algorithm based kalman filter (FVKF) in three unique stages. Overall simulated results justify the effectiveness of the Fast versatile algorithm based Kalman filter (FVKF) over voice signal improvement.

Keywords: Voice Improvement, Auto Regressive model, Fast Versatile algorithm, Noise estimation ,Traditional Kalman Filter.

1 Introduction

Earlier research on this related provide a guideline on satisfactory improvement of voice quality using Kalman separating calculation. [1]. Where it basically need to assess the parameters of Auto Regressive model followed by the noisy elimination utilizing Kalman calculation. This followed estimations of Linear Prediction Coding (LPC) coefficient and inverse matrix but it significantly increments over the computational complexity of the KF calculations. A better filtering efficiency can be achived with further technique of suppressed noisy signal on decaying the nature of the voicing signal subject to exactness of estimating the parameters of Auto Regressive model [2]. The simple KF design in the absence of computing Linear Prediction Coding multiplying element for Auto Regressive type and Non adaptive design having more number of excess information with matrix reverse operations has non-versatile behavior.

On improvement over lagging of traditional KF for voicing signal enhancement, we propose a Fast versatile algorithm on KF designing through this research work. This algorithm fixed the principal estimation for state direction X(n), it removes more number of matrix activities and decreases the time complication nature of the calculation. Since this is very problematic to find atmosphere noisy signal exactly, the KF calculation is further requirement. in association with versatile algorithm on evaluating the surrounding noise. the overlooking element which has been referenced by [4] and [5] to correct the estimation of surrounding noise by the perception information consequently has great role on designing proposed model on getting out the real noise.

2 Designing Of Proposed Model

2.1 Kalman Basic Modelling

Conventional Kalman Filter provide a recurrent solving approach to the discontinuous-data linear filtrate problem to evaluate the present value of the variables it measures regardless of precision, with utilization of

(1) Information on the framework and estimation gadget elements,
(2) actual description of the framework noises, estimation mistakes,
(3) accessible data about beginning states of the variables.

This principle utilizes a total description of the probability for its estimation errors towards ideal filtering gains deciding factor. On evaluating its performance as an element of the "structure parameters" of the following systems has to be adopted

Different sensors are utilized;

- Current Place and directions to be assessed for individual sensors.
- Permissible range of noisy qualities for the sensing devices;
- First -special methods for removing noise sensing communication
- Testing grades for various sensor types and degree of type re-arrangements for lessening usage prerequisites.

The following can be use on Kalman Filter designing as required with possible accessibility and figure (1) expressing all its operation behaviors

- Those which predict the conditions or update the time
- Those which update conditions or update the observed information

To Update Time:

Corresponding State prediction and error co variance process can be express through this relationship given on following equations.

$$x_{(n|n-1)} = A_{(n|n-1)} \cdot x_{n-1} \quad \ldots\ldots\ldots\ldots \quad (1)$$

$$R_{\{e,n|n-1\}} = A_{\{,n|n-1\}} \cdot R_{e,n-1} \cdot A^{T}_{n,n-1} + u . R_w . u \ldots\ldots\ldots\ldots\ldots (2)$$

To Update Observation:

The specified equations for the state correction further express as as follows .

$$R_{e,n} = R_{\{,e,n|n-1\}} - R_{\{,e,n|n-1\}} \cdot C^T \cdot F_n^{-1} \cdot C . R_{\{,e,n|n-1\}} \ldots\ldots (3)$$

$$x_n = x_{(n|n-1)} + R_{\{,e,n|n-1\}} \cdot C^T \cdot F_n^{-1} \cdot (y_n - C . x_{(n|n-1)} \ldots\ldots (4)$$

$$F_n = C . R_{\{,e,n|n-1\}} \cdot C^T + R_v \ldots\ldots\ldots (5)$$

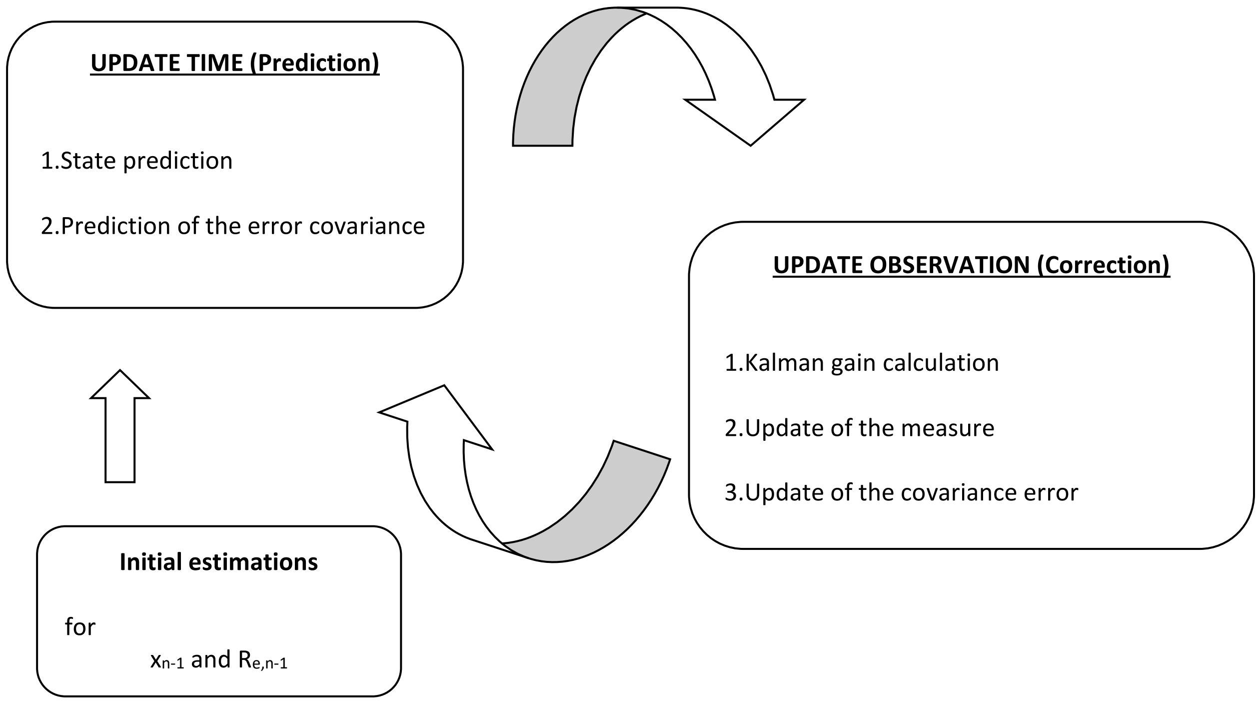

Fig. 1. A pictorial representation of Kalman Filter designing

2.2 Fast Versatile Algorithm with Kalman Filter

It observed that noise changes in environmental conditions drastically. Versatile algorithm with kalman filter (FVKF) is utilized on estimation of noise for stably updated. This in practice assured an accurate expression of the atmosphere noise, so it can adjust to any alternates in surrounding noise and purposeful on voicing signal enhancement.

The designing structure of this modeling system. containing two individual operation sequence, that can be express as follows

1. Update the Change of the surrounding environmental noise ‘ $R_{v(n)}$’
2. Update the brink point ‘ U’

Modernizing The Change of Surrounding Atmosphere Unwanted Signal

Change of the surrounding atmosphere unwanted signal

$$Rv(n) = (1 - L) * Rv(n) + L * Ru(n) \text{----6}$$

Here to improve the new observations under present estimations we use L, Where L is the factor for Loss it restricts the filtering memory length and. 'L' is characterized as:

$$L = {(1-b)}/{1 - b^{t+1}} \quad \text{------7}$$

Where b value is constant in between 0.95 and 0.99.

Updating Of The Threshold

$$limit\ U = (1 - L) * U + L * Ru(n) \text{ ------8}$$

Here L is used again to reduce the error, where error is large with increase of noise. Due to the updated brink point, U is not at all barred by the constrained

$$R_u(n) <= U$$

Two SNRs between the current speech frames and whole speech signal has been calculated.
The estimation of noise signal is attenuated to avoid the corrupted voicing signals [7].
The variance of noisy signal is guessing to be δ_v^2 and Signal to Noise Ratio of the noisy data SNR_{in}, is given by

$$SNR_{in} = 10\ log_{10}\{[\frac{1}{n+1}\sum_{k=0}^{n} L^2(k)]/\delta_v^2\}dB \text{ -----9}$$

Here k being the entire total samples present in the voicing data

3 Simulated Results & Analysis

On evaluating filtering efficiency, pure voicing data, noisy data and noisy signal with speech signal has been compared from the result of traditional KF and FVKF under the state of SNR_{in} 10[dB] . Fig. 2 show here the improved strategy hat gives an incredible with same outcomes as the Traditional technique.

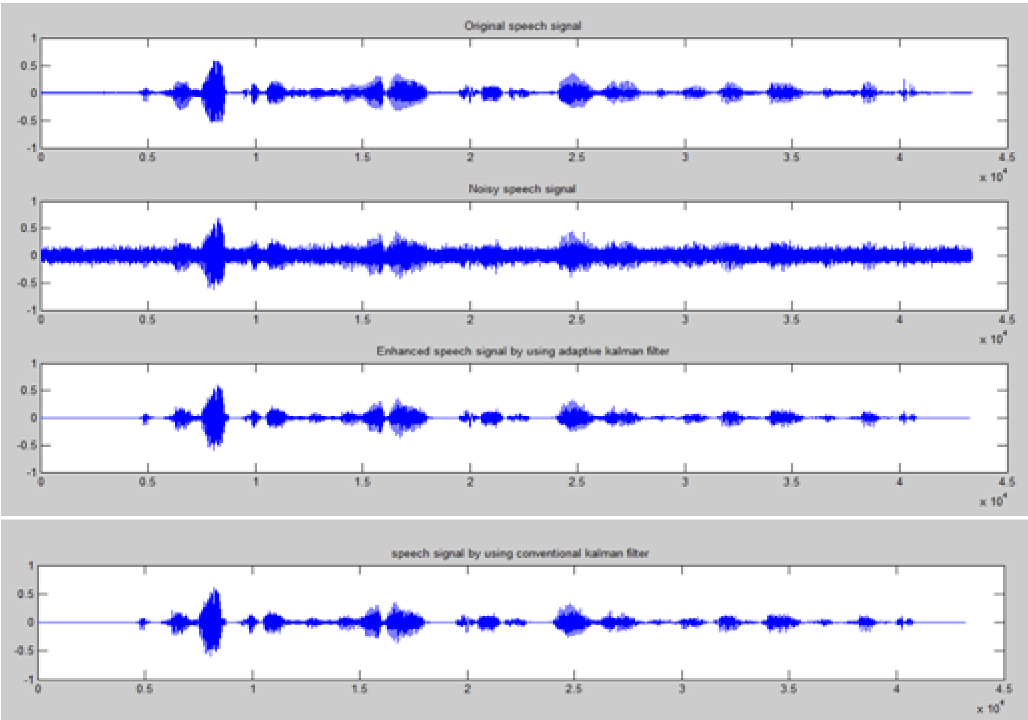

Fig.2. Representation Of Signal Wave

The process of Comparison of ouput Signal to Noise Ratio can be through the relationship as bellow

$$SNR_{Out} = 10\, log_{10}\left[\sum_{k=0}^{n} \mathrm{L}^2(\mathrm{k})\right] / \sum_{k=0}^{n}\{\mathrm{L}(\mathrm{k}) - \mathrm{L}(\mathrm{k})\}^2] \quad \mathrm{dB} \quad \text{---10}$$

Let D is transition Matrix dimension, Dt implies the transition matrix for traditional technique, observation process consider 20, 30, 40, and 50 as four different level .Assume Df value equal to 1 since It implies the transition matrix for FVKF technique whose value is 1.

SNR_{in} (dB) ------→			0dB				5dB				10dB			
Transition matrix dimension(Dt) ---→			20	30	40	50	20	30	40	50	20	30	40	50
SNR_{Out} (dB)	Female	Tradition kalman	2.63	3.02	3.25	3.44	7.09	7.55	7.89	8.03	11.02	11.83	12.02	12.27
		FVKF	3.67				8.31				13.99			
	Male	Tradition kalman	2.25	2.75	3.14	3.63	5.65	6.42	6.9	7.07	9.2	10.58	10.79	11
		FVKF	3.84				7.092				11.26			

Table 1. The Noisy Voice Signal, Snr_{out} With Gaussian Noise

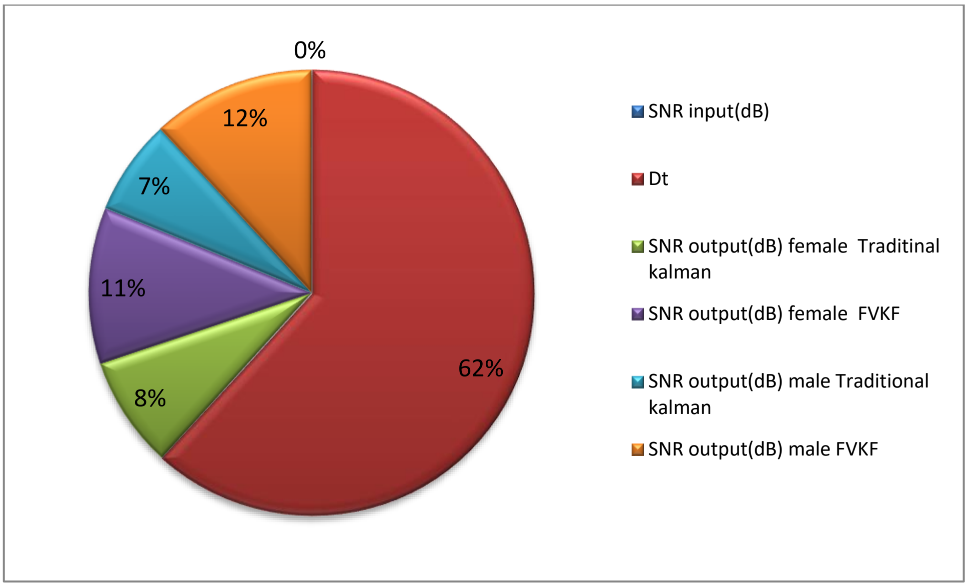

Fig.3. Snr_{out} Percentage Improvement

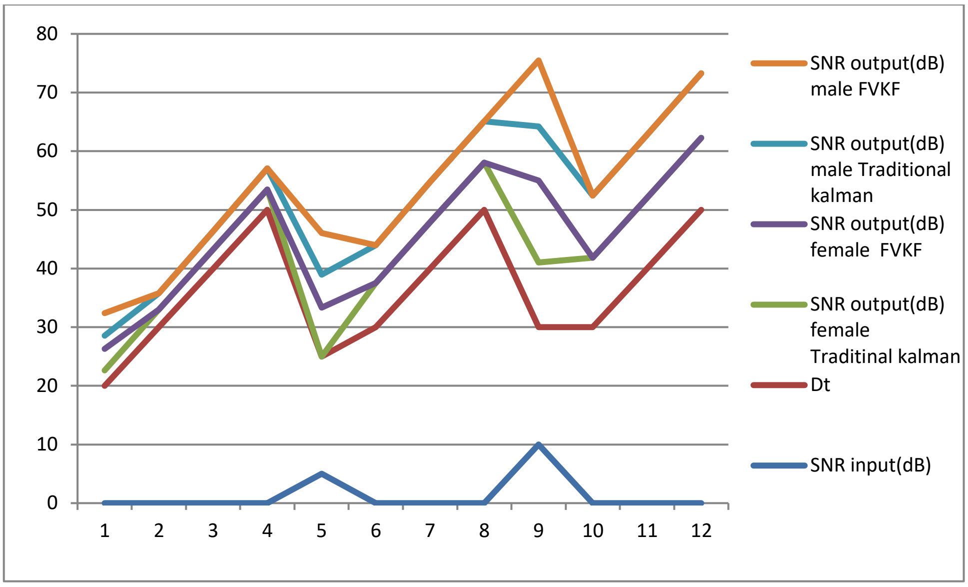

Fig.4. SNR_{Out} Comparisons

Running Time Comparision

Compare results in three levels of SNR_{in} that is 0 dB,5dB and 10dB with the duration of lady voicing signal duration is 10 seconds , male voicing is 10 seconds ,with four levels of Dt such as 20,30,40 and 50

Fig.5.shows Running Time Percentage Improvement between traditional kalman and FVKF and Fig. 6. Shows the Comparision of Running Time

SNR input(dB) ---→			0dB				5dB				10dB			
Transition matrix dimension(Dt) ----→			20	30	40	50	20	30	40	50	20	30	40	50
RUNNING TIME (Sec)	female (10s)	Traditional kalman	10.233	32.75	39.56	56.8	10.26	33.02	39.6	57.01	10.2	33.02	39.62	57.02
		FVKF	**2.3464**				**2.362**				**2.4088**			
	Male (10s)	Traditional kalman	9.750083	30.98	37.64	53.1	9.633	31.23	37.3	53.677	9.63	31.2	37.58	52.832
		FVKF	**2.195**				**2.231**				**2.234**			

Table 2. Running Time

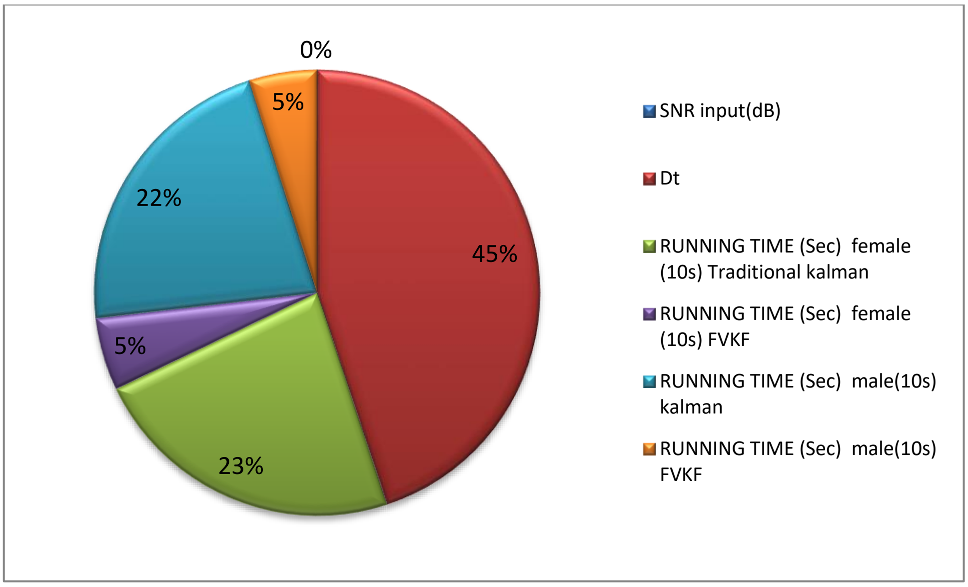

Fig.5.Running Time Percentage Improvement

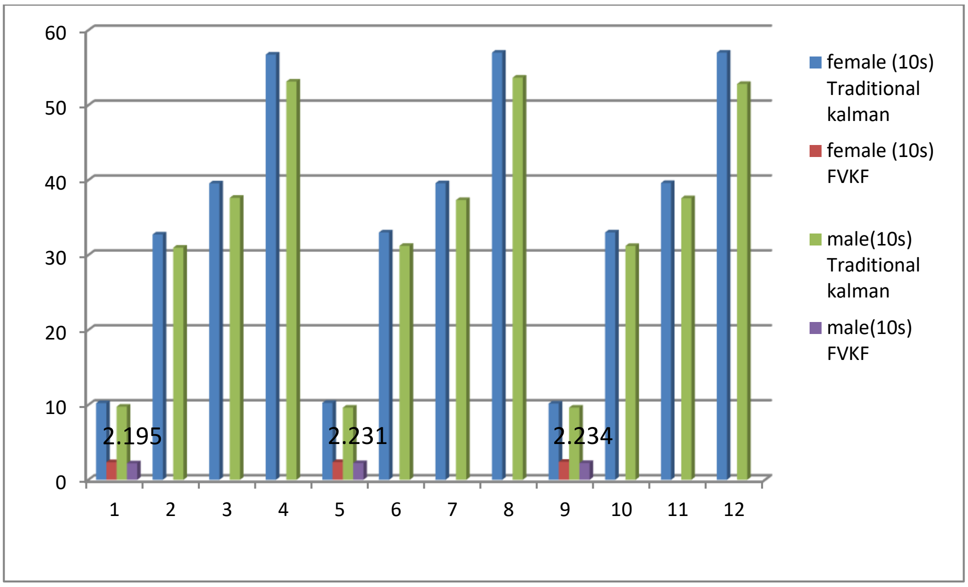

Fig. 6. Comparision of Running Time

4 Conclusion

This research work outcome expressing embedding of Versatile algorithm with Kalman Filtering for increasing the quality of speech signal removing the Matrix calculation and calculate a coefficient factor. At the same time the effectiveness of proposed algorithm based on the numerical results and subjective evaluation is successfully justified. the novality of this approach is Even though kalman is invented earlier, but improving the performance of this filter is ever green. We can get better results with kalman prediction and estimation of the signals. Here we proposed improvement in kalman operational speed and reduce the number of matrix calculations

Particularly, two-step multiplications is the basic reason to get less running time with higher SNR_{out} while the Voicing datas are decreased by the noisy signal. It is also analyzed that this FVKF concept applied is less complex and can knowing the better noisy elimination in spite of the decrease of the computations complication with-out effecting the nature of the voicing data too. Furthermore, it can be improving over the Fast versatile technique in this paper to give a greater accuracy over calculation of surrounding noisy data.

References

1. Nikolaos Dionelis and Mike Brookes,"Phase-Aware Single-Channel Speech Enhancement With Modulation-Domain Kalman Filtering", IEEE/ACM Transactions On Audio, Speech, And Language Processing, Vol. 26, No. 5, (2018)

2. Martin Krawczyk-Becker, *Student Member, IEEE*, and Timo Gerkmann, *Senior Member, IEEE*, "Fundamental Frequency Informed Speech Enhancement in a Flexible Statistical Framework IEEE/ACM Transactions On Audio, Speech, And Language Processing, (2016)

3. Andrew C. Harvey, "Forecasting, structural time series models and the Kalman filter", Cambridge university press ,online publication ,Cambridge (2014)

4. J.D. Hamilton, "Time Series Analysis", Princeton University Press, 1994. Jacob Benesty, Shoji Makino, Jindong Chen (Eds.), "Speech enhancement", Springer, (2005)

5. L.R. Rabiner, R.W. Schafer, "Digital Processing of Speech Signals", Prentice Hall, (1978).

6. Wan-xin, HUANG Chun-mei, LIU Peiwei, MA Ming-long."Application of adaptive Kalman filter technique in initial alignment of inertial navigation system". Journal of Chinese Inertial Technology, Volume.18, No.1, pp.44-47, February.(2010)

7. Srilakshmi garaga,O.Sudhakar" Speech enhancement by a Kalman filter based smoother in white and colored noise" IJSER journal volume3,issn 2229-5518,September (2012)

8. Y. Hu, P. Loizou, "Subjective comparison of speech enhancement algorithms" in Proc. IEEE Acoustics, Speech, and Signal Proc. ICASSP- 2006, Touluse, , Vol. I, Pages 153-156. France, May (2006)

9. Nari Toshiniro Furukawa ,Hideaki Matsue and Shigeo Tsujii. "Kalman Filter for Robust Noise Suppression in White and Colored Noise". IEEE International Symposium on Circuits and Systems,1172-1175 (2008)

10. Deller, J.R., J.H.L. Hansen, J.G. Proakis, "Discrete – Time Processing of Speech Signals", IEEE Press, (2000).

11. GAO Yu, ZHANG Jian-qiu. "Kalman Filter with Wavelet-Based Unknown Measurement Noise Estimation and Its Application for Information Fusion". Acta Electronica Sinica, Vol.35, No.1, pp.108-111, Jan.(2007).

12. WU Chun-ling, HAN Chong-zhao. "Square-Root Quadrature Filter".Acta Electronica Sinica, Vol.37, No.5, pp.987-992, May.(2009)

13. S. So and K. K. Paliwal, "Modulation-domain Kalman filtering for single-channel speech enhancement," *Speech Commun.*, vol. 53, no. 6, pp. 818–829, Jul. (2011)