Ehco-cancellaton using General Kalman filter and comparison between GKF,APA and NLMS

Mr. Rahul R. Hattarki, Prof. P. D. Bahirgonde

***Abstract*—The Kalman filter is very important tool in signal processing which is widely used in many application. In this paper we used the GeneralKalman filter (GKF) in contest of echo cancellation, to reduce complexity of Kalman filter we used the General Kalman filter. At first stage the processing of speech signal by adding noise as well as echo on real time as well as on database, by using different algorithm such as GKF, NLMS and APA algorithm removed Echo form the original signal. Simulation result shows that the performance of General Kalman filter is having best results amongst above algorithm.**

***Keywords*—Echo cancellation;Room impulse response(RIR); General Kalman filter ;**

Tool : MATLAB

1. Introduction

In case of transmission of sound the most affection parameter is the echo. Echo is the parameter which give the same sound with repeated version if it mixed into the original sound through microphone it amplified in the P.A. system and created problem to hear properly. There are the two types of Echo, Acoustic Echo and hybrid Echo these types are based on where it observe.
The Kalman Filter is having optimum solution which is widely used signal processing tool for the filtering purpose [1]. In this paper the Kalman filter is used for the echo cancellation. Before the addition of echo preparation of noisy environment is more important to observe the performance of all filters.

2. Creation of noisy environment

The noisy environment is created by using pre-defined functions in MATLAB such as Gaussian or by Room impulse response (RIR). In RIR can be calculated by using Image model. It gives response of n number of virtual sources [2] [6]. This concept is broken into three fold a) Approximationof physical situation b) Finding of unit impulse response for each echo with proper time delay c) Then we will calculate the magnitude of each echo's unit impulse response.

First Author Name:*Mr. Rahul R. Hattarki*
Department of electronics and telecommunication engg.
N.K. Orchid Collage of engg. Technology Solapur
Solapur, India

Second Author Name:*Prof. P. D. Bahirgonde*
Department of electronics and telecommunication engg.
N.K. Orchid Collage of engg. Technology Solapur
Solapur, India

a) Approximation of Physical situation

As said the physical situation can be calculated by using Image model is called visualization process. As follow we shown the rectangular room within it the sound source is shown by green circle and the black star depicts that it is microphone where we have to calculate the room impulse response, between the black star and circle the path is taken by sound wave shown in fig. 1.[3]

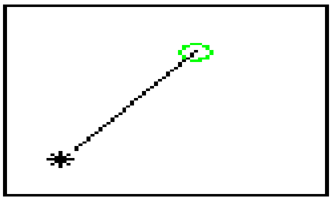

Fig. 1. Sound path in rectangular room

When we correlate this concept with the light such that how the reflection is done with the same way sound reflections are also observed then the path of sound can be observe as sum of incident path and the reflected path [4]as shown in fig.2

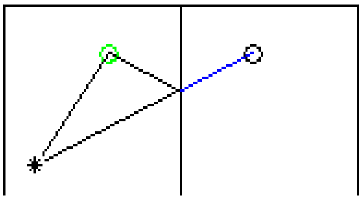

Fig.2. Reflected path of sound in rectangular room

Due to reflections the sound rays coming from the obstacle is behave as a virtual source for the microphone. So by multiple reflection it forms the multiple virtual sources as shown in fig.3

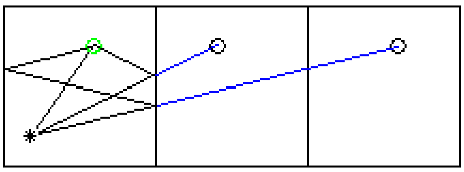

Fig.3. Reflection multiple virtual sources in rectangular room

Which is extended in [4] as two dimension and three dimension in our analysis by treating each virtual sources its response cause the noise (small echo as reverberation)
By considering above image model at the start we can take the initial approximate for the single dimension and with the same way we can transfer the same in tree dimension [2].
Now consider the individual virtual source and treat it as in the One dimension. This is depicted in fig. 4.

ISSN: 2278 – 1323 All Rights Reserved © 2015 IJARCET

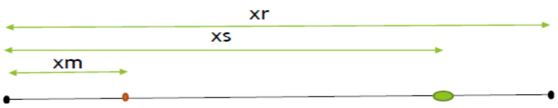

Fig.4.position model for one dimension

xs=co-ordinate of sound source, xr = length of room , xm = co-ordinate of microphone ,xs is considered as the x-ordinates of sound source and xris the length of room in single x-dimension. The position of I[th] virtual source and microphone can be calculated as follow[2]

$$Xi = (-1)^{i}\, xs + [i + \frac{1-(-1)^{i}}{2}]xr$$

$$Xi = (-1)^{i}\, xs + [i + \frac{1-(-1)^{i}}{2}]xr - xm$$

With the same way the relative positions along y-axis and z-axis can be calculated by converting all dimension with respect to y-axis and z-axis respectively

$$Yj = (-1)^{j} ys + [j + \frac{1-(-1)^{j}}{2}]yr - ym \text{ , and}$$

$$Zk = (-1)^{k} zs + [k + \frac{1-(-1)^{k}}{2}]zr - zm$$

Now we can find out the distance to each virtual source can be find out by Pythagorean Theorem as,

$$d_{ijk} = \sqrt{xi^2 + yj^2 + zk^2}$$

This equation will represent the distance in three dimensional Matrix. In MATLAB to convert these co-ordinates into the three dimensional matrix used the meshgrid function.

Let t is the time and d_{ijk} is the equation given by Pythagoras theorem the so difference in time for the given time is given by

$$u_{ijk} = t - \frac{d_{ijk}}{c}$$

Where c is velocity of sound in air when all parameters of environment is normal by default it is 343 m/s. In above equation the effective delay is given by the known formula distance upon velocity i.e. $\frac{d_{ijk}}{c}$

The unit impulse response can be calculated as,

$$a_{ijk} = 1 \quad \text{if } u_{ijk} = 0$$
$$= 0 \quad \text{otherwise}$$

Now each impulse response function has the magnitude of our echo at $u_{ijk} = 1$ there are two parameters affect to the magnitude of echo first is the distance travelled by the sound to get source to microphone,

$$b_{ijk} \propto \frac{1}{d_{ijk}}$$

and second one is the reflection due to wall if the all walls in the hall is having the same reflection coefficient, say r_w and raise the exponent n = |i|+|j|+|k|, which is represented the total number of reflection made by sound.

$$rijk = r_w^n = r^{|i|+|j|+|k|}$$

We now find the total magnitude of each echo as

$$eijk = bijk \,.\, rijk$$

and impulse response can be obtain as

$$h(t) = \sum_{i=-n}^{n} \sum_{j=-n}^{n} \sum_{k=-n}^{n} aijk.eijk$$

3. GKF

To cancel the echo, desired signal or microphone with discrete time index n [5](fig. 5).

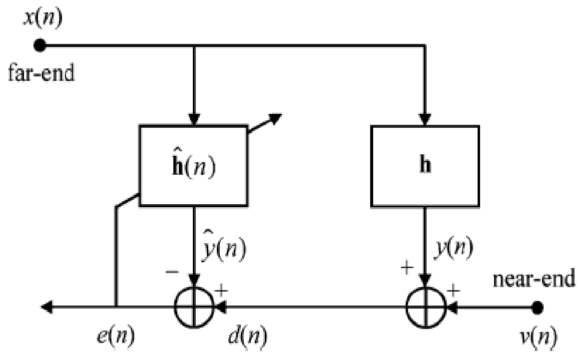

Fig5. the general configuration for echo cancellation

Then

$$d(n) = X^{T}(n)\, h + v(n)$$

$$= y(n) + v(n)$$

Where, v(n) is the zero mean white Gaussian noise. X(n) is the signal from loud speaker which contain the L most recent time samples of loud speaker.

$$\therefore X(n) = [x(n)\; x(n-1)\; x(n-2) \ldots x(n-L+1)]^{T}$$

Andh = $[h_0 \;\; h_1 \ldots h_{L-1}]$ Is the impulse response of the system from microphone to loud speaker, and $\sigma_v^2 = E[v^2(n)]$.The y(n) is the echoed signal that we have to cancel the echo using adaptive filters[1][5].

Here our objective is to identify the optimum recursive estimator h with an adaptive filter:

$$\hat{h}(n) = [\hat{h}_0(n)\; \hat{h}_1(n) \ldots \hat{h}_{L-1}(n)]^{T}$$

The linear sequential Bayesian approach, the effective estimator is h(n).

$$\hat{h}(n) = \hat{h}(n-1) + K(n)\,[d(n) - X^{T}(n)\, \hat{h}(n-1)]$$

$=\hat{h}(n-1) + K(n).e(n)$,

Where the K(n) is the Kalman gain Matrix and

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

$= d(n) - X^{T}(n).\hat{h}(n-1)$

Which is the priori error signal vector between the microphone signal vector and the state estimation error is,

$$\mu(n) = h(n) - \hat{h}(n)$$
$$R\,\mu(n) = E[\mu(n).\mu^{T}(n)]$$

And the priory misalignment is as

$$m(n) = h(n) - \hat{h}(n-1)$$

$$= \mu(n-1) + w(n),$$

For which the correlation matrix is

ISSN: 2278 – 1323 All Rights Reserved © 2015 IJARCET

$$Rm(n) = E[m(n).\ m^T(n)]$$

Then the equations can be summarizes into

$$Rm(n) = R\ \mu(n-1) + \sigma_w^2(n)\ I_l$$

Where $w(n)$ is zero mean Gaussian noise signal vector which is uncorrelated with v(n). Then the correlation matrix $Rw=\sigma_w^2(n)IL$ and I_L is the identity matrix of L × L .the variance captures the uncertainties in h(n).

$$Re(n) = XT(n)\ .\ Rm(n).X(n) + \sigma_v^2(n)Ip$$

$\sigma_e^2(n) = X^T(n)Rm\ x(n) + \ \sigma_v^2(n),$
Which is the correlation matrix of priority error signal

$K(n) = Rm(n).X(n).R_e^{-1}(n)$or

$K(n) = \frac{1}{\sigma_e^2(n)}Rm(n).x(n)$

$e(n) = d(n) - X^T(n).\hat{h}(n-1)$

$\hat{h}(n) = \hat{h}(n-1) + K(n).e(n)$

$R\ \mu(n) = [I\ L - K(n)X^T(n)]\ Rm(n)$

At initially consider $\hat{h}(n) = 0$ and $R\ \mu(n) = \varepsilon I_L$,where ε is the small positive number .for the $\sigma_w^2(n) = 0$ the Kalman filter have poor tracking and for the large value of $\sigma_w^2(n)$ the Kalman gain matrix never goes to zero which is allow to update the impulse response $\hat{h}(n)$.

4. APA
In this part we will show how the General Kalman filter is having good approximation with APA algorithm [main paper] Initially we assume that the GKF has started to converge then Rm tends to become the diagonal matrix with all elements equal to small positive number so we get the approximation:

$Rm(n) \approx \sigma_m^2(n)I_L$

As the result the Kalman gain simplifies to
$K(n) \approx K_{APA}(n) = X(n)[X^T(n).X(n) + \partial(n)Ip]^{-1}$
Where,$\partial(n) = \frac{\sigma_v^2}{\sigma_m^2(n)}$,
$\partial(n)$ Is the regularization parameter then we deduce the GKF into the APA as:
$K_{APA}(n) = X(n)[X^T(n).X(n) + \partial(n)Ip]^{-1}$,

$e(n) = d(n) - X^T(n).h(n-1)$,

$\hat{h}(n) = \hat{h}(n-1) + K_{APA}(n).e(n)$,
Simulation confirm that the GKF behaves as the APA at convergence.

5. NLMS
As from the equations of GKF the algorithm is striking resemblance with the classical RLS algorithm [7] the two parameters $\sigma_w^2(n)$ and σ_v^2 in the Kalman filter allows better control but the RLS is does not depend on.

After the convergence and with the approximation of Rm(n) $\approx \sigma_m^2(n)I_L$ We get

$e(n) = d(n) - X^T(n).\hat{h}(n-1)$

$\hat{h}(n) = \hat{h}(n-1) + \frac{x(n).e(n)}{X^T(n)X(n)+\partial(n)}$

Which becomes the classical NLMS algorithm when the regularization parameter is defined as in APA.

6. Simulation Results for echo cancellation
For getting the original signal when in between we clapped which can be shown graphically as

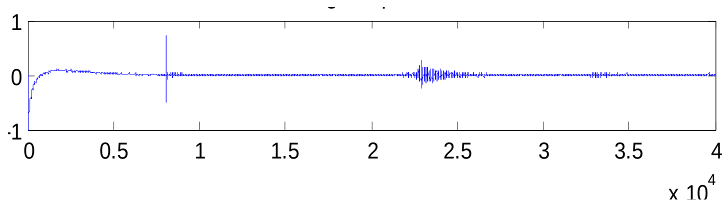

Fig.6 Original speech signal
Where the horizontal x axis shows the time and y axis shows the amplitude. We recorded the input signal for the 4 second

After the getting the original signal we added echo in real time by scaling the output as channel selected as stereo (multichannel) case.

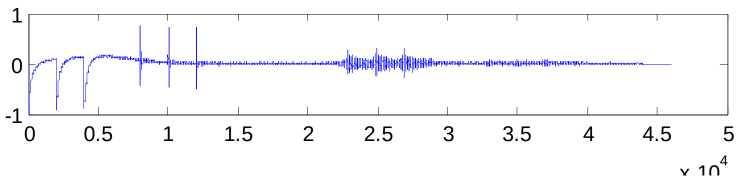

Fig.7 Echoed speech in multichannel case
In this we added the echo three times manually, By adding the room impulse response the echoed speech can be look like

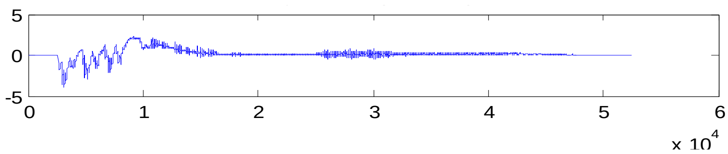

Fig.8 Echoed speech with addition of room impulse response
Up to here we processed the original signal to fit to remove echo from the noisy environment original speech taken to observe the peaks in original and echoed signal
After the removal of echo using each filter as mentioned above the response is as shown below for the one more observation

ISSN: 2278 – 1323 All Rights Reserved © 2015 IJARCET

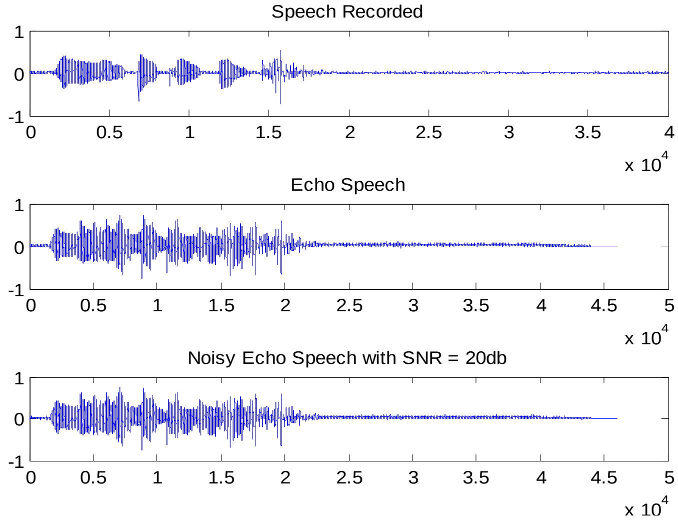

Fig.9 Speech recorded and added the noise 20db SNR AWGN

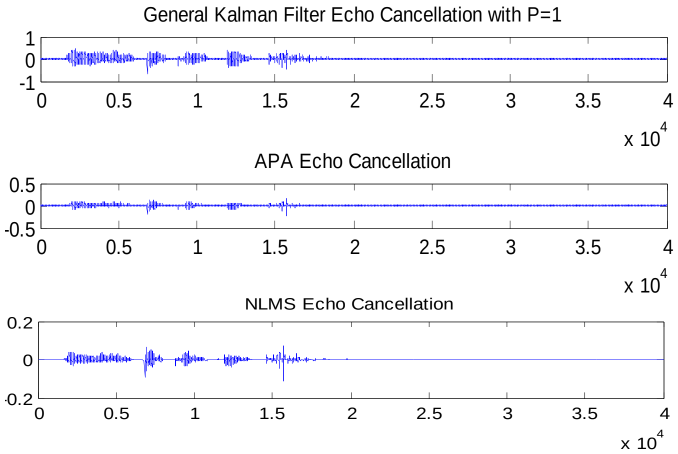

Fig.10 Echo cancellation speech

As in the first observed that P = 1 means most recent time sample

7. Misalignment analysis

Misalignment means the something is wrongly aligned with the other

For the filters the normalize misalignment in dB evaluated as

$$\text{Mis}(n) = 20\log_{10}\frac{\|h-\hat{h}(n)\|2}{\|h\|2}$$

in all the experiment the results are averaged with 20 independent trials.

8. Comparison of Misalignment results

As the simulation results shows that the normalize misalignment is less for GKF as here we performed for the most recent valued samples as P = 1 and for the NLMS the value of alpha i.e. step size parameter is selected equal to 0.1.

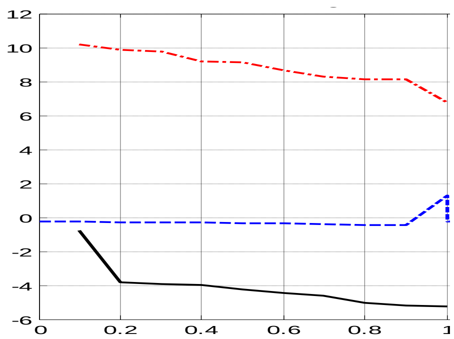

Fig.11 Misalignment analysis of GKF, APA, NLMS

For step size parameter equal to 1 then we get high misalignment than the previous one.

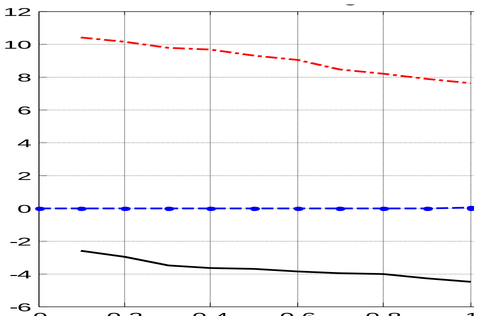

Fig.12 Misalignment analysis of GKF, APA, NLMS

9. Conclusion

In this paper, we studied the General Kalman filter in context of echo cancellation and by observing the simulation result in misalignment analysis the GKF shows the more optimum performance as compared to APA and NLMS.

10. REFERENCES

[1] R. E. Kalman, “A new approach to linear filtering and prediction problems,” *J. Basic Eng.*, vol. 82, no.1, pp. 35–45, Mar. 1960.

[2] Stephen G. McGovern,” A Model for Room Acoustics”, book, 2003,2004

[3]Frank L. *Pedrotti*, "*Introduction to Optics*", 3rd Edition, Prentice Hall Inc.

[4] Allen, J and Berkley, D. *'Image Method for efficiently simulating small-room acoustics'*. TheJournal of the Acoustical Society of America, Vol 65, No.4, pp. 943-950, 1978

[5] Constantin paleologue*t al ,”* Study of the general kalman filter for echo cancellation“Vol. 21, no. 8, august 2013

[6] H. Buchner, "Acoustic echo cancellation for multiple reproduction channels: From first principles to real-time solutions," *Proc. ITG Conf. on Speech Communication*, Aachen, Germany, Oct. 2008.

[7] A. H. Sayed and T. Kailath, “A state-space approach to adaptive RLSfiltering,” *IEEE Signal Process. Mag.*, vol. 11, no. 3, pp. 18–60, Jul.1994.

ISSN: 2278 – 1323 All Rights Reserved © 2015 IJARCET