



International Conference on Emerging Trends in Engineering, Technology & Management (ICETM-2025)

Conducted by Viswam Engineering College (UGC—Autonomous Institution) held on 11th & 12th, April- 2025

IMPLEMENTATION OF SPEECH DENOISING USING SPECTRAL SUBTRACTION AND KALMAN FILTERING

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ABSTRACT: In this paper, I have described about the speech denoising using spectral subtraction and kalman filtering. Speech denoising aims to improve the speech quality and intelligibility by using various techniques and algorithms. Speech communication is the process to have noise unrestricted speech signal with good clarity and quality along with high performance. In real world, it is very difficult to get clean signal all time for speech communication system. Speech processing is required in communication systems to apply noise reduction techniques so as to get desired speech signal from the corrupted speech signal. The proposed method, spectral subtraction and kalman filtering algorithms uses to suppress the additive background noise. It is easy for implementation and widely used in simplicity. Denoising is used to estimate average noise spectrum for noisy speech signal spectrum. The performance of spectral subtraction and kalman filtering algorithm for speech is evaluated by using Mean square error, Peak Signal to Noise Ratio, Normalized Root mean square error and Distortion. In this method Simulink model and hardware implementation in raspberry pi has been used.

Keywords: Speech signal, Spectral subtraction, Kalman filtering, AWGN channel, Voice activity detector, Noise canceller and Raspberry pi.

1. INTRODUCTION

In order to share any information from one place to another place, communication systems play an important role. Speech is a fundamental, common media to communicate. The observed speech signal to be measured is of good quality and clarity. The main objective of speech denoising is to improve one or more aspects of speech such as overall quality and clarity. Speech processing systems are usually designed for noise free environment. In real world environment, presence of back ground noise is unavoidable. Speech denoising algorithms are applied to the problem as background noise removal and multi speech separation in modern speech communication system.

Every case of speech communication involves listener, speaker and various communication systems. Therefore, background noises and noisy surroundings are possible to affect people particularly for hearing loss. The area inspects removing the noise from corrupted speech employing various signal processing methods known as speech processing. Different procedures of speech communication such as speech enhancement, speech recognition and speech coding are present.

Related Work

The noise presented in speech signal can affect the signal quality, reduce the noise and increase listener fatigue. In many noises present in recording speech, the problem of noise reduction is essentially in the world of telecommunication. In noise reduction, there are few algorithms to be presented. When their input or output signals are corrupted by noise, noise reduction algorithms are used to improve the performance of speech communication systems. The main objective of speech denoising is to improve one or more aspects of speech such as overall quality and intelligibility. Speech processing is designed for noise free environments



but in real world environment, the presence of background noise is inevitable. It requires speech processing systems such as Speech coding, speech recognition and speaker recognition. The spectral subtraction algorithm are based on the principle that estimate clean speech signal which can be obtained in subtracting by estimate the noise signal spectrum from noisy speech signal spectrum.

Enhancement systems are considered as a single channel and multi-channel methods. Single channel enhancement is used for spectral subtraction algorithm and ease of implementation. Multi-channel enhancement is difficult to operate. Noisy speech is obtained from some algorithms which uses noise estimation operation to calculate the overall noise in original speech signal. Then noise estimated is too slow, unwanted residual noise will be audible else too high, speech will be uncleaned. As these algorithms involve a forward and inverse Fourier transforms, they are very easy to implement. We presented different algorithms that mitigated the Musical noise distortion.

In different approaches in spectral subtraction algorithm is presented for improving the speech signal from the boisterous situations. The creators say that the clean signals quality is debased by the added substance foundation noise. All the accessible methods in the spectral subtraction calculation is traditionally one of the first calculations proposed for foundation noise diminishment. In different strategies to diminish the background noise from the speech signal are presented. Background noise is decreased to certain cutoff by the spectral subtraction strategies exploiting regulation domain and geometric approach. When subjective and destination tests were performed on the adjustment approach, we get enhanced speech quality.

2. SPEECH DENOISING ALGORITHMS

2.1. Spectral Subtraction:

Speech denoising is used in different algorithms as one of the algorithms, is based on spectral subtraction. Spectral subtraction was first proposed by S.F. Boll. To subtract the estimation of average noise spectrum from noisy speech spectrum based on the basic principle of spectral subtraction.

The noisy signal model in time domain is given by

$$y(m) = x(m) + n(m) \quad (1)$$

where $x(m)$, $n(m)$ and $y(m)$ are the speech signal, the additive noise and the noisy signal respectively, and m is the discrete time index.

In the frequency domain, the equation for noisy signal model is

$$Y(f) = X(f) + N(f) \quad (2)$$

where $Y(f)$, $X(f)$ and $N(f)$ are the Fourier transforms of the noisy signal $y(m)$, speech signal $x(m)$ and noise $n(m)$ respectively and f is the frequency variable.

Spectral subtraction may be expressed as

$$|\hat{X}(f)|^b = |\hat{Y}(f)|^b - \alpha |\hat{N}(f)|^b \quad (5)$$

Where $|\hat{X}(f)|^b$ is an estimate of the original signal spectrum $|X(f)|^b$ and $|\hat{N}(f)|^b$ is the time-averaged noise spectra.

2.2. Kalman filtering:

Kalman filter is the mathematical method named after Rudolf E. Kalman (1960). It was developed as a recursive solution to the discrete data linear filtering problem and simply recursive data processing

algorithm. Using feedback control Kalman filter algorithm estimates the previous process i.e., it estimates the process to a moment over the time. It is probable to separate them in to two sets:

1. Those which update the notice data and update equations.
2. Those which update the period or prediction equations.

The first one is to take care of feedback and they add new information inside the previous estimation to achieve an improved estimation of the state. The second one has to throw the state to n moments taking reference state on n-1 moment. State prediction equation is

$$x_{n/n-1} = A_{n,n-1} \cdot x_{n,n-1} \text{-----} (8)$$

$$R_{e,n/n-1} = A_{n,n-1} \cdot R_{e,n-1} \cdot A_{n,n-1}^T + u \cdot R_w \cdot u^T \text{-----} (9)$$

state correlation equation is given below

$$R_{e,n} = R_{e,n/n-1} - R_{e,n/n-1} \cdot C^T \cdot F_n^{-1} \cdot C \cdot R_{e,n/n-1} \text{-----} (10)$$

$$x_n = x_{n/n-1} + R_{e,n/n-1} \cdot C^T \cdot F_n^{-1} (y_n - C \cdot x_{n/n-1}) \text{-----} (11)$$

$$F_n = C \cdot R_{e,n/n-1} \cdot C^T + R_v \text{-----} (12)$$

3. IMPLEMENTATION AND RESULTS

Speech denoising is implemented in Simulink design and in raspberry pi.

In Spectral Subtraction and Kalman Filtering methods designed in which speech signal is added to AWGN channel to produce noisy speech. This method is used to de-noise the noisy speech.

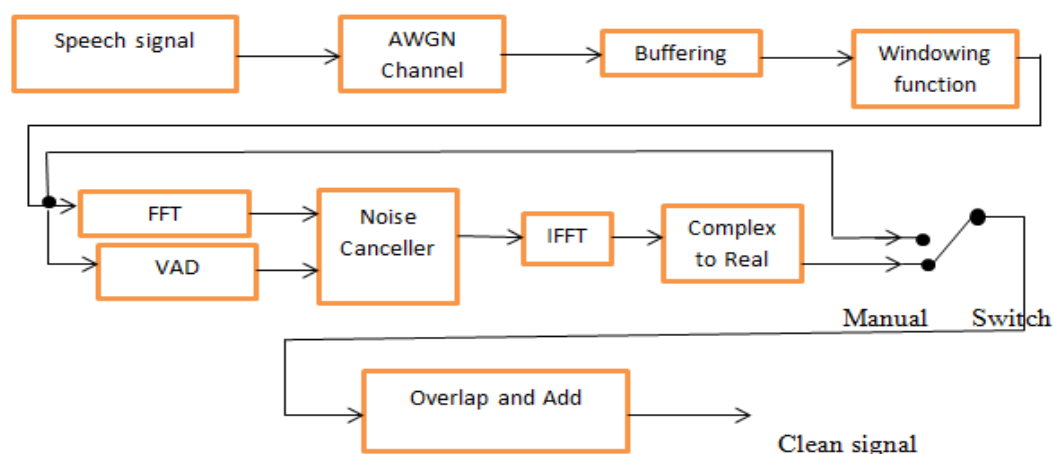


Fig.1 Spectral Subtraction Method

Spectral subtraction method is used for speech de-noising. Speech signal is added to AWGN channel to produce noisy speech. Noisy speech uses 1024 samples and then the samples are converted into frames in buffer. Windowing function is used as hamming window in this method. FFT used to convert time domain to frequency domain. IFFT used to convert frequency domain to time domain. Voice Activity Detector used three blocks Feature extraction, Decision module and Decision smoothing. Feature extraction is used to implement full band energy, zero crossing rate. Decision module is used to represent in logical algebraic

module. Noise Canceller is used to remove the noisy speech in noise. It consists of Bias Removal, Half Wave Rectification and Residual Noise Reduction.

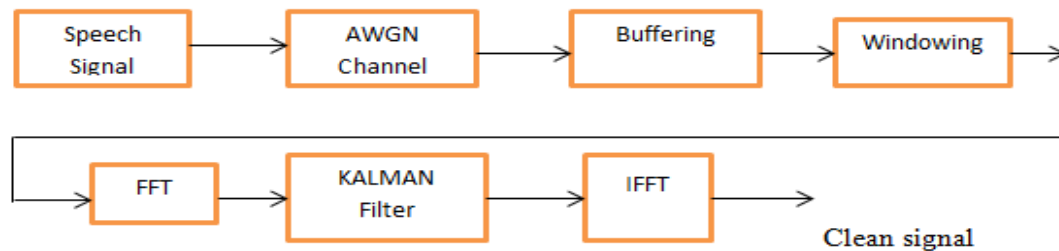


Fig. Kalman filtering

In kalman filtering, speech signal is a pass through an AWGN channel to produce noisy speech signal. When noisy speech samples are converted into data buffering used in frames, frames pass through a hamming window function.

FFT is used to convert in time domain to frequency domain. After passing through kalman filtering, Estimates the state of a dynamic system from sequence of imperfect and noisy measurements. It can be used the previously estimated state to predict the current state and also used for the current measurement and predicted state to estimated current state values.

3.2 Objective quality measurements:

1. Signal to Noise Ratio (SNR) :

Signal to Noise Ratio is defined as the power ratio of noisy signal and clean signal.

$$SNR = 10 \times \log_{10} \frac{\text{mean}(\text{noisy signal}^2)}{\text{mean}(\text{noisy signal}^2 - \text{clean signal}^2)} \quad \text{-----} \quad (20)$$

2. Peak Signal to Noise Ratio (PSNR):

It is used to represent the ratio of maximum possible value of a noisy speech signal to the clean signal.

$$PSNR = 10 \times \log_{10} \frac{\text{length} \times \max(\text{noisy signal}^2)}{\text{noisy signal}^2 - \text{clean signal}^2} \quad \text{-----} \quad (21)$$

3. Mean Square Error (MSE) :

The Mean Square Error (MSE) is measured in dB is one of the ways to determine the difference between noisy values and clean values of the quantity are being estimated.

$$MSE = \frac{1}{\text{length}(\text{noisy signal})} \times \sum (\text{clean signal} - \text{noisy signal})^2 \quad \text{-----} \quad (22)$$

4. Normalized Root Mean Square Error (NRMSE) :

The Root Mean Square Error is the measure of determining the normalized value of difference between the values predicted by a clean and noisy signal. The RMSE provide with an average of the magnitude of errors calculated over a period of time. RMSE is a good accuracy.

$$NRMS = \frac{\sqrt{\text{mean}[(\text{noisy signal} - \text{clean signal})^2]}}{\sqrt{\text{mean}[(\text{noisy signal} - \text{mean}(\text{noisy signal}))^2]}} \quad \text{-----} \quad (23)$$



Distortion:Distortion is typically unwanted and often efforts made to reduce it as possible. The addition of noise or other outside signals may lead to distortion of signal.

$$D = \frac{1}{\text{length}(\text{noisy signal})} \times \sum(\text{clean signal} - \text{noisy signal}) \text{ ----- (24)}$$

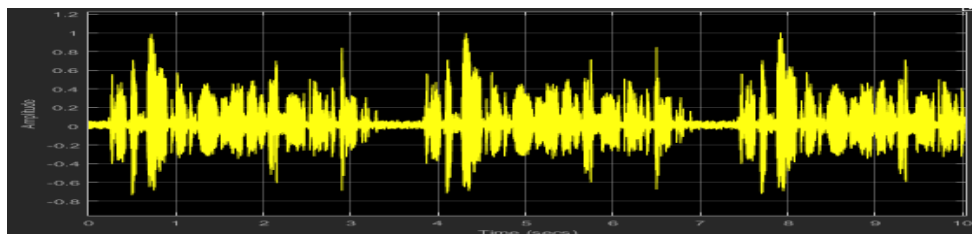


Fig.3.speech_dft.wav in Time scope for noisy speech signal in spectral subtraction

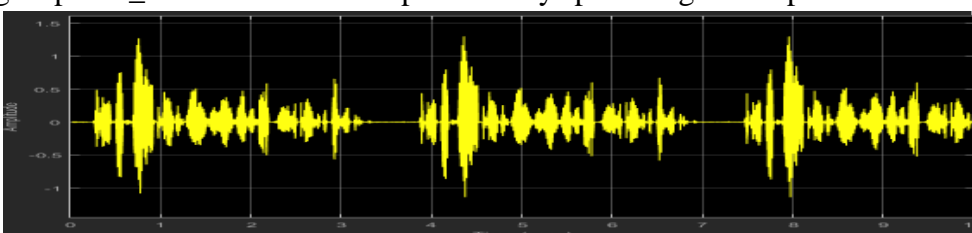


Fig.4. speech_dft.wav in Time scope for clean speech signal in spectral subtraction

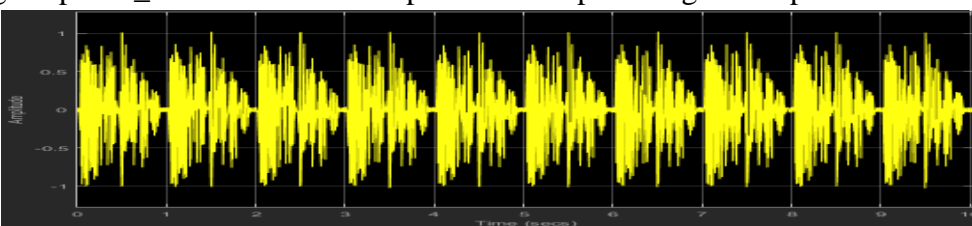


Fig.5. about_time.wav in Time scope for noisy speech signal in spectral subtraction

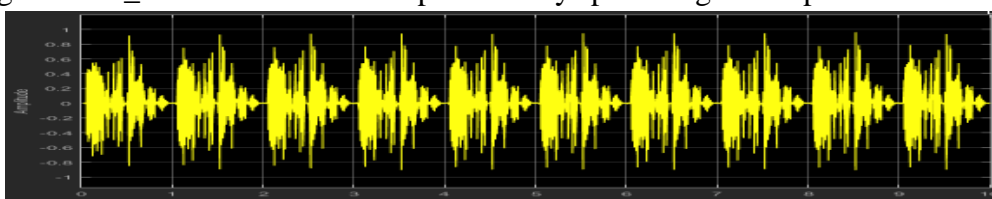


Fig.6. about_time.wav in Time scope for clean speech signal in spectral subtraction

S. No	Noise Power	PSNR	MSE	SNR	NRMSE	Distortion
1	0.02	136.3	1.666e-07	1.223	2.785e+16	-0.01821
2	0.04	142	9.543e-08	1.218	9.293e+14	-0.01827
3	0.06	147.7	5.42e-08	1.213	9.-57e+14	-0.01831
4	0.08	153.4	3.09e-08	1.209	1.553e+15	-0.01835
5	0.1	160.7	1.5e-08	1.204	2.313e+15	-0.01838

Table1.Quality measurements for speech_dft.wav at different noise powers in spectral subtraction



S. No	Noise Power	PSNR	MSE	SNR	NRMSE	Distortion
1	0.02	107.9	7.35e-08	0.4736	6.94e+15	-0.0002942
2	0.04	116.9	3.19e-08	0.334	3.422e+15	-0.0003507
3	0.06	127.6	1.155e-08	0.2585	8.922e+14	-0.0003941
4	0.08	144.3	2.257e-09	0.2112	1.072e+15	-0.0004307
5	0.1	188.4	2.839e-11	0.1780	6.961e+15	-0.0004629

Table2. Quality measurements for speech_dft.wav at different noise powers in spectral subtraction

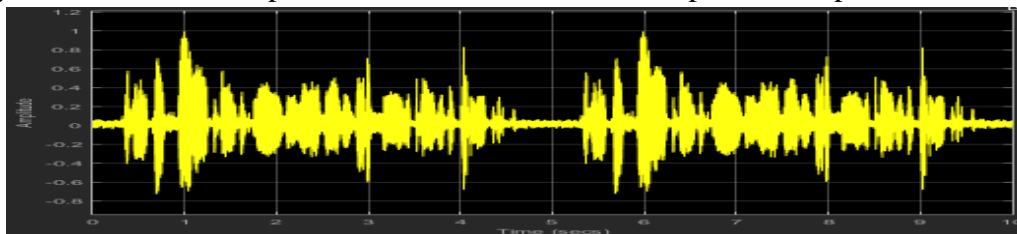


Fig.6.speech_dft.wav in Time scope for noisy speech signal in kalman filtering

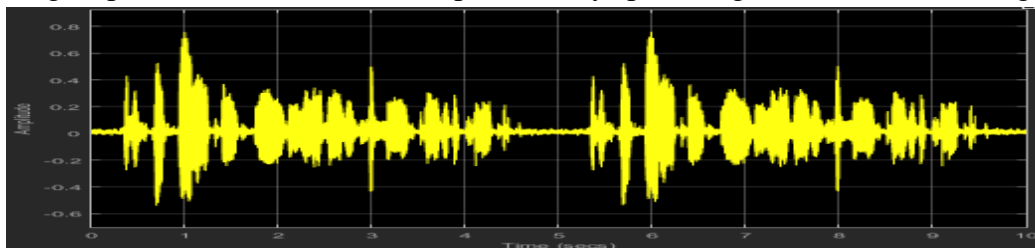


Fig.7.speech_dft.wav in Time scope for clean speech signal in kalman filtering

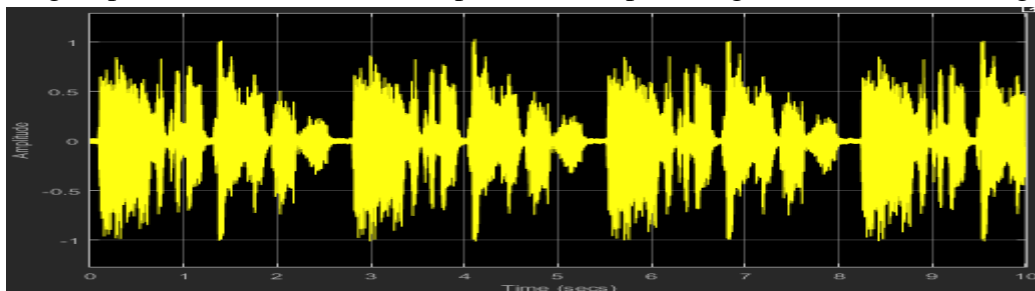


Fig.8.about_time.wav in Time scope for noisy speech signal in kalman filtering

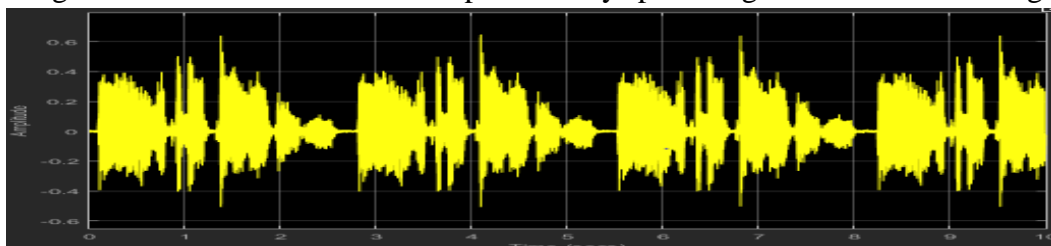


Fig.9.about_time.wav in Time scope for clean speech signal in kalman filtering



S. No	Noise Power	PSNR	MSE	SNR	NRMSE	Distortion
1	0.02	38.61	1.375e-05	0.7608	1.119e+16	-0.010
2	0.04	46.97	6.967e-07	0.8039	5.413e+15	-0.009391
3	0.06	55.53	3.31e-06	0.8164	9.933e+15	-0.008616
4	0.08	65.99	1.272e-06	0.8174	4.59e+15	-0.007962
5	0.1	82.28	2.688e-07	0.8071	inf	-0.007387

Table3. Quality measurements for speech_dft.wav at different noise powers in kalman filtering

S. No	Noise Power	PSNR	MSE	SNR	NRMSE	Distortion
1	0.02	29.16	0.005859	0.5858	9.77e+15	-0.2034
2	0.04	29.51	0.005717	0.588	1.465e+16	-0.2033
3	0.06	29.78	0.005608	0.5896	1.464e+16	-0.2032
4	0.08	30	0.005517	0.591	2.927e+16	-0.2031
5	0.1	30.2	0.005438	0.5922	1.463e+16	-0.203

Table4. Quality measurements for about_time.wav at different noise powers in kalman filtering

4.CONCLUSION AND SUMMARY

This paper presents implementation of speech denoising using spectral subtraction and kalman filtering. These algorithms are used to improve in speech quality, clarity and intelligibility. Objective quality measurements are used to produce different values. Spectral subtraction method is used to improve speech quality. Kalman filtering is used to improve speech intelligibility. These algorithms used to reduce in noise and improve in SNR, MSE, PSNR values. Wave signals are used to Noise power is varied with observed in quality measurements. In spectral subtraction, noise power varies in increase PSNR, decrease SNR, and improve MSE and Distortion in Table 1&2. In Kalman filtering, noise power varies in improve Distortion and MSE in table 3&4. In this method, to represent wave signals in particular time scope that is sampling time 10 are used. Summary in these two methods used to implement in speech denoising. These two algorithms are used to improve the clean signal and quality. Future extension speech denoising is used in adaptive filters to improve speech quality.

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