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Removal of Impulsive Noise from Speech by Using Different Filters

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ABSTRACT

This Paper presents survey of impulsive noise from speech. In this paper we are using different filters to remove impulsive noise. Use of image is becoming popular in every fields astronomy, space exploration, medical imaging, scanning techniques an many others. This paper gives the brief introduction of noise, types of noise,and about the techniques used for noise removal. All the techniques used in this paper for removal of impulsive noise from speech were implemented in MATLAB.

Keywords : Laplacian filter,Gaussian filter,Fuzzy filter,Convolution filter,Impulsive noise,rank order mean,Wiener filter.

I. INTRODUCTION

An impulse noise filter can be used to enhance the quality of noisy signals, in order to achieve robustness in pattern recognition and adaptive control systems. A classic filter used to remove impulse noise is the median filter,at the expense of signal degradation.Thus it's quite common,in order to get better performing impulse noise filters, to use model-based systems that know the properties of the noise and source signal (in time or frequency), in order to remove only impulse obliterated samples. All recording devices, both analogue and digital, have traits which make them susceptible to noise.The fundamental problem of image processing is to reduce noise from a digital color image. The two most commonly occurring types of noise are i) Impulse noise, ii)Additive noise (e.g. Gaussian noise)and iii) Multiplicative noise (e.g. Speckle noise).

Impulse noise is usually characterized by some portion of image pixels that are corrupted, leaving the remaining pixels unchanged.Examples of impulse noise are fixed-valued impulse noise and randomly valued impulse noise.We talk about additive noise when value from a certain distribution is added to each image pixel, for example,a Gaussian distribution.Multiplicative noise is generally more difficult to remove from images than additive noise because the intensity of the noise varies from the signal intensity (e.g., speckle noise).

II. METHODS AND MATERIAL

TYPES OF NOISE

a) White noise

. It can be any value between 0 and 2π, and its value at any frequency is unrelated to the phase at any other frequencyWhite noise is a sound or signal consisting of all audible frequencies with equal intensity. White noise is a random signal with a constant power spectral density. At each frequency, the phase of the noise spectrum is totally uncertain. White noise is a discrete signal whose samples are regarded as a sequence of serially uncorrelated random variables with zero mean and finite variance; a single realization of white noise is a random shock. The following figure represents the white noise

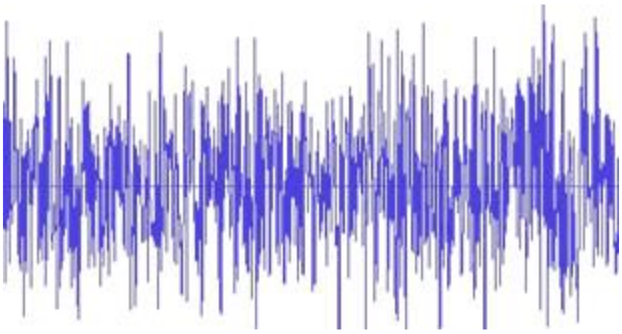


Fig 1:white noise

b) Coloured Noise:

Any noise that is not white can be termed as coloured noise. Coloured noise has frequency spectrum that is limited within a range unlike white noise which extends over the entire spectrum.

There are different types of coloured noise (brown noise, pink noise, etc.) depending upon the gradation in the Power Spectral Density (PSD) of the noise. Coloured noise can be generated by passing white noise through a filter with required frequency response.

c) Brown noise:

Brown noise is the kind of signal noise produced by Brownian motion, hence its alternative name of random walk noise. The term "Brown noise" comes not from the colour, but after Robert Brown, the discoverer of Brownian motion

The following figure represents the brown noise

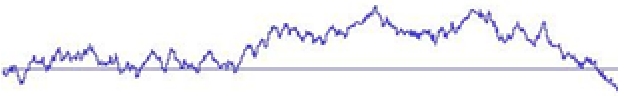


Fig 1.1a:Brown noise

d) Pink noise:

Pink noise is a signal or process with a frequency spectrum such that the power spectral density (energy or power per Hz) is inversely proportional to the frequency of the signal. In pink noise, each octave (halving/doubling in frequency) carries an equal amount of noise power. The name arises from the pink appearance of visible light with this power spectrum.

The following figure represents the pink noise:

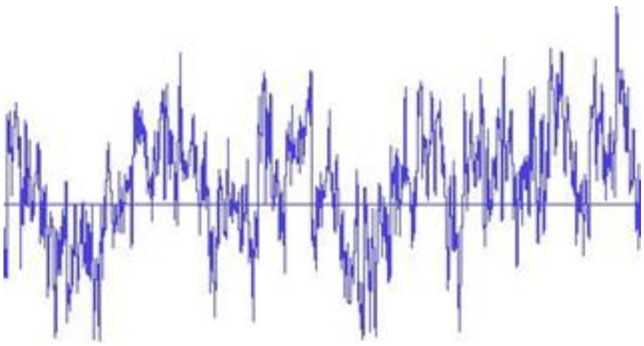


Fig 1.1b:Pink noise

e) Impulsive noise:

Impulsive noise refers to sudden bursts of noise with relatively high amplitude. This type of noise causes click sounds in the signal of interest.Impulsive noise is generally modelled as contaminated Gaussian noise. The ratio of the variances of the two Gaussian noises decides the impulsive Character of the noise generated.

The following figure represents the impulsive noise

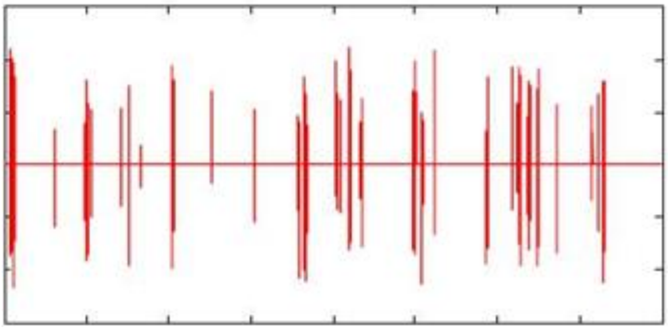


Fig 1.1c:Impulsive noise

2.<u>LITERATURE REVIEW:</u> From the past few decades,most of the attention was given to system modeling of speech signals and has been incorporated in several commercial speech coding standards. There are mainly two types of Algorithms that are used for adaptive filtering.

(1) The first is Stochastic Gradient based algorithm known as Least Mean Square (LMS) Algorithm and the

(2) second is Recursive Least Square (RLS) Algorithm. adaptive noise cancelling as an alternative method of estimating signals which are corrupted by additive noise by employing LMS algorithm. A great deal of research has been carried out in subsequent years for finding new variant of these algorithms to achieve better performance in noise cancellation application.The proposed method reduces the computation time drastically without degrading the accuracy of the system. When compared to the LMS based Windrow model, it was shown to have superior performance. The LMS based algorithms are simple and easy to implement but the convergence speed is slow.

According to Abhisekh Tondon. introduced an efficient low complexity Normalized Least Mean Square (NLMS) algorithm for echo cancellation in multiple audio channels. The performance of the proposed algorithm was compared with other adaptive algorithms for acoustic echo cancellation. it has reduced complexity while providing a good overall performance.

As proposed by Dr. M. Satya Sai Ram a 3-stage 2-switch 3-part Multi Switched Split Vector Quantization (MSSVQ) has provided better performance in term of spectral distortion, complexity and memory requirements is less when compared with Split Vector Quantization (SVQ) and Multi Stage Vector Quantization(MSVQ). Whereas MSVQ has better spectral distortion but complexity and memory requirements are more but the quality of the signal is less. So in order to enhance the performance of MSVQ to get better quality of the signal without losing the information, speech enhancement techniques are proposed before transmitting the signal.

Many researchers have given a detailed review of how to remove impulsive noise(salt and pepper) from speech which can be found in a major issue in noise is the quantization of LP parameters.

By Tsai and Yang [12], Comparison of the PCA-based fast search method using PCA-LBG-based VQ codebooks and the full search method using TY-LBG codebooks developed.

The implementation procedure of LBG algorithm is given below:

Design a single vector codebook; which is the centroid of the complete set of training vectors and hence there is no iteration.

Split each current codebook into two parts and according to the rule. Where the value of can vary from 1 to the current size of the codebook. is a splitting parameter (choose =0.01).

Each training vector is analyzed to get the code word which is close to Nearest-Neighbor Search in the current codebook by looking into similarity measurement parameter and assign that vector to the corresponding cell which is allied with the closest code word.

Find the centroid of the training vectors assigned to that cell to update the code word in each cell.

Iteration 1: repeat steps 3 and 4 until the average distance falls below a preset threshold level.

Iteration 2: repeat steps 2, 3 and 4 until a codebook size of M is achieved

FLOW CHART FOR LBG ALGORITHM:

The LBG algorithm is used to design codebook of M stages. The algorithm starts with a single large code book and the process starts by splitting the codebook into two group vectors as per splitting technique on the code and this process of splitting will continue till the desired vector codebooks of size M are obtained.

3.REMOVING NOISE FROM SPEECH BY FILTERING

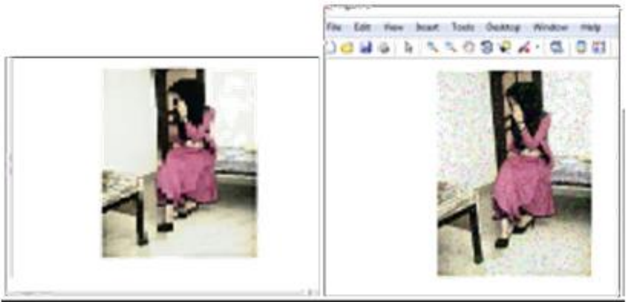
1. **RANK ORDER MEAN:** A computationally efficient algorithm is proposed to remove noise impulses from speech and audio signals while retaining its features and tonal quality. The proposed method is based on the SD-ROM (Signal Dependent Rank Order Mean) algorithm. This technique has successfully been used to remove impulse noise from images. It has the advantage of being relatively fast, simple and robust. The algorithm estimates the likelihood the sample under inspection is corrupt relative to the neighboring samples and replaces a sample detected as corrupted by a value based on the neighboring samples. This algorithm also has the advantage of being 'configurable' to the type of noise impulses in the sample, as the thresholds used to detect noise impulses can be varied to suit the signal. Standard global filtering techniques like low pass filtering do not differentiate between impulses corrupted samples and uncorrupted samples.
2. **LAPLACIAN FILTER:** The Laplacian is a 2-D isotropic measure of the 2nd spatial derivative of an image. The Laplacian of an image highlights regions of rapid intensity change and is therefore often used for edge detection. The Laplacian is often applied to an image that has first been smoothed with something approximating a Gaussian smoothing filter in order to reduce its sensitivity to noise. The operator normally takes a single graylevel image as input and produces another graylevel image as output.
3. **GAUSSIAN FILTER:** A **Gaussian filter** is a filter whose impulse response is a Gaussian function (or an approximation to it). Gaussian filters have the properties of having no overshoot to a step function input while minimizing the rise and fall time. This behavior is closely connected to the fact that the Gaussian filter has the minimum possible group delay. It is considered the ideal time domain filter, just as the since is the ideal frequency domain filter.These properties are important in areas such as oscilloscopes and digital telecommunication system.

Mathematically,a Gaussian filter modifies the input signal by convolution with a Gaussian function,this transformation is also known as the Weierstrass transform.

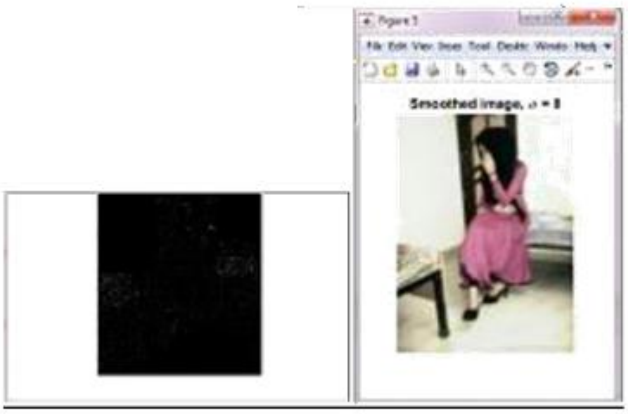
4. **FUZZY FILTER:** A fuzzy filters for noise reduction deals with fat-tailed noise like impulse noise and median filter.Only impulse noise reduction uses fuzzy filters. The proposed system presents a new technique for filtering

narrow-tailed and medium narrow-tailed noise by a fuzzy filter. The system first estimates a "fuzzy derivative" in order to be less sensitive to local variations due to image structures such as edges. Second, the membership functions are adapted accordingly to the noise level to perform "fuzzy smoothing." A new fuzzy filter is presented for the noise reduction of images.

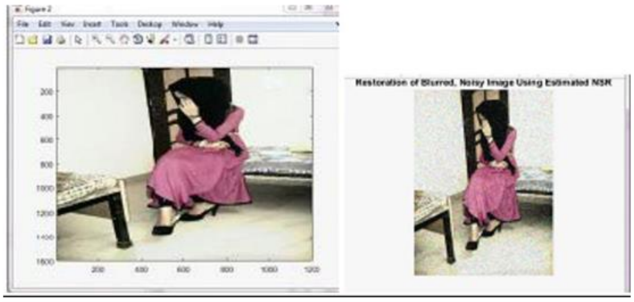
5. **WIENER FILTER:** The **Wiener filter** is a filter used to produce an estimate of a desired or target random process by linear time-invariant (LTI) filtering of an observed noisy process, assuming known stationary signal and noise spectra, and additive noise. The Wiener filter minimizes the mean square error between the estimated random process and the desired process. The goal of the Wiener filter is to compute a statistical estimate of an unknown signal using a related signal as an input and filtering that known signal to produce the estimate as an output.



(a)Real image **(b)Rank order mean filter**



(c) Laplacian filter **(d)Gaussian filter**



(e) Fuzzy filter **(f) Wiener filter**

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