Removing the Noise from the audio Signal using a Technique Median Filter

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Abstract

Audio signal processing is one of the most important branches of digital signal processing. it has wide application in many fields, especially in communication systems and mobile phones. The aim of this study is to extract and analyze information from audio signals including the mean µ ,standard deviation σ and Signal Noise to Ratio SNR) of the recorded audio signals as a function to the different distances. In addition to studying the histogram of these signals and comparing them with the sound signals which have been added noise to it , and improved using the median filter. The results showed high efficiency in noise removal and in maintaining the characteristics of the recorded voice signal.

Keywords: Audio Signal, Median Filter, The statistical characteristics, Noise

Introduction

The audio signal is a representation of sound, typically using either a changing level of electrical voltage for analog signals, or a series of binary numbers for digital signals. Audio signals have frequencies in the audio frequency range of roughly 20 to 20,000 Hz, which corresponds to the lower and upper limits of human hearing. Audio signals may be synthesized directly, or may originate at a transducer such as a microphone, musical instrument pickup. Loudspeakers or headphones convert an electrical audio signal back into sound [1]. It is possible that the signal is a function of one dimension, such as audio signal is a function of the amplitude change with time also is the noise data is undesirable result from the changes that occur on the signal or signals that interfere with the audio signal which causing distorted and lack of clarity in Microphone that works to record sound and record it with acoustic signals that come from sources of sound other near the source of sound is required, which is the background of noisy, add to random values to each element of the audio signal and this is the first effective for the emergence of noise in the audio signals [2]. The process of converting the voice signal to an electrical signal and then into a digital signal subject to a process of sampling and quantization in addition to the changes that occur due to conditions of environment registry as they lead to that the reference is not clear in addition to the context in which the audio signal for example when you run an electrical device near the TV radio or note distort the audio signal and the picture together and that the result of overlapping signal of the device electrical signals connecting to the TV, thereby causing the appearance of noise mixed to indicate TV , It is noteworthy that the fluctuation electric wires carrying the signal plus the resistance wire that turns into heat energy in the wires also lead to the generation of signal noise, unwanted distortion signal passers-by [2,3]. In many applications, sound processing is handled the presence of the background noisy and sound is one of the important branches to deal with speech or sound is focused primarily on the best guess for the voice vestiges of noise or the signal distorted and the process of improving the sound issue is not easy and the total removal of the background noise is virtually impossible, and distortions in the content of the voice signal is essential [4]. There are many previous studies that involved studied the noise of audio signal and analysis it. The following are the most important of these studies:

- In 2001, C. Chandra , M.S. Moore and S.K. Mitra , The researchers proposed an algorithm to remove noise impulses from speech and audio signals while retaining its features and tonal quality[5].

- in 2010. Derry FitzGerald, present a fast, simple and effective method to separate the harmonic and percussive parts of a monaural audio signal. The technique involves the use of median filtering on a spectrogram of the audio signal with median filtering[6]
- in 2013 , Ching-TaLu presented study to reducing the quantity of musical residual noise by a two-stage speech enhancement approach. In the first stage a preprocessor enhances noisy speech . In the second stage the enhanced speech signal is post-processed by an iterative-directional-median filter to significantly reduce the quantity of residual noise[7].
- in 2015 Jyoti Jaybhay & Rajveer Shastri are evaluate study to the Different filters have been developed as Mean and Median filters, Statistic Lee filter, Statistic Kuan filter, Frost filter, Srad filter used to remove speckle noise[8].

- in 2018 Bhawna Dhruv; Neetu Mittal; Megha Modi , studied the attempts to analyze the efficacy of different filtering techniques on the image containing 04 types of noises Gaussian, Poisson, Salt & Pepper and Speckle. The performance of filtering techniques Median, Average and Wiener is evaluated by performance measuring parameters execution time and entropy[9].

The statistical characteristics of audio signal

Based signal processing digital audio primarily to the information contained in the reference and method of distribution, and this information was that has been observed and in accordance with the principle of physical subject to the distributions of statistical Therefore, the statistics indicate to be very important in the process of analysis and digital signal processing, as these statistics gives the features and qualities of the nature of the signals and how to distribution of information and things and the statistics are linked to the principle of probability distribution of information and statistical measures of the signal be linked to the principle of probability distribution of intensity in the signal and the most important statistical measures [10]:

1 - Average

Average intensity of the signal known as the symbol μ, a single statistical mean average value of the signal comes from the collection of all samples the voice signal and dividing by the total number of samples N and computes the average μ of the relationship of the following: -

$$\mu=\frac{1}{N}\sum_{i=0}^{N} f(t) \qquad -1-$$

As f (t): t Ps in the audio signal.

It can be calculated based on the average probability distribution of intensity in the relationship as the following:

$$\mu=\sum_{g=0}^{L-1} g \cdot P(g) \qquad -2-$$

$P(g)$: probability distribution of intensity g in the signal

L: number of levels of intensity in the signal and is usually 256 levels.

2-Standard deviation:

Defined as the amount of deviation of the signals values for the rate and calculate the standard deviation of the following relationship [10,11]:

$$= \sqrt{\frac{1}{N}\sum_{i=0}^{N} (f(t) - \mu)^2} \qquad -3-$$

3-Mean Square error

Defined as the cumulative square error between output signal and the original signal and is one of the criteria by which to know the signal quality, and the Mean square error from the following relationship [10,11]:

$$MSE = \frac{1}{N}(f(t) - f(t)^{\sim})^2 \qquad -4-$$

As $f(t)$: data represent the original signal.

$f(t)$: represents the processed signal, impure with noise or that MSE required for it .

N : total number of elements of the signal .

3-Signal to Noise Ratio

The difference between the reference and the reference within a noise outside so that each element in the emerging signal consists of a signal involved in addition to the noise If the reference is a homogeneous intermediate part of the audio signal, the Signal to Noise Ratio is given by the following:

$$SNR = 20Log\left(\frac{\mu}{\sigma}\right) \qquad -5-$$

Here you can refer to the values calculated for regions of homogeneous signal intensity of this standard ⌈of µ to be efficient in the evaluation of the signal and the relationship is clear that the reference to not depend on the noise, the signal is in various formats can fall between two values as in the specific relationship of the following:

Imin <I <Imax

Therefore, the proportion of signal to noise that we have proposed adoption of the signal audio is:

$$SNR = 20\ Log\left(\frac{I_{max} - I_{min}}{\sigma}\right) \qquad -6-$$

Since the ∫ standard deviation of signal , in the above two relationships need not be taken into consideration the amount (20 Log) because the figures that appeared to our small numbers do not show clear differences between them and enter the amount in the case of (20 Log) of the two relations[10,11].

The Noise in the audio signal

is signal at random and are the result of electrical signals randomly generated as a result of natural causes appear in the system or outside of these random changes add to the reference information can be hidden partially or entirely, and are these changes in the signal as a result of the numbering and the conversion from analogue to digital causes of noise also The output signal of the audio recording cannot be free of noise, including noise caused by fans of the computer or clicks the keyboard [12] and the process of reduction of noise is an important issue in applications such as mobile communications and speech recognition, signal processing, medical and many other applications where the signs are not can be separated from the noise , types of noise can be classified mathematically as follows [12,13]:

1- Noise additive
2 - Noise Multiplicative
3 - Impulse noise.

The Gaussian Noise Type

Gaussian noise takes the bell-shaped curve distribution as shown in figure(1), which can analytically described as [13]:

$$G(g) = \frac{1}{\sqrt{2\pi\sigma^2}} e^{-\frac{(g-m)^2}{2\sigma^2}} \quad \ldots\ldots\ldots\ldots\ldots \quad (7)$$

Where G(g) is the distribution of the random variable (g), m is the mean and σ is the standard deviation(σ^2 is variance) of (g). About 70% of all the values fall within the range from one standard deviation (σ) below the mean to one above, and about 95% fall within a distance twice the value of standard deviations. The Gaussian model is most often used to model natural noise process, such those occurring from electronic noise in the image acquisition system.

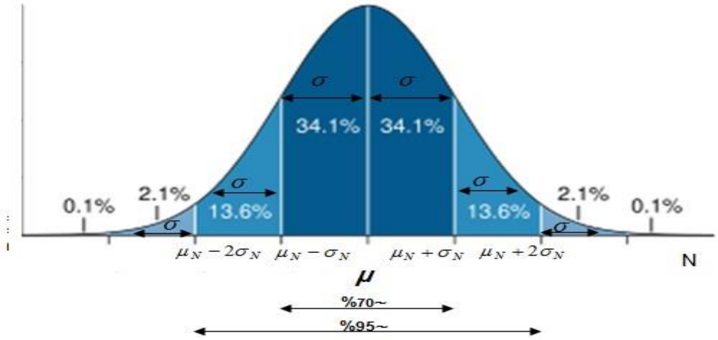


Figure (1) show the distribution of Gaussian Noise

Median Filter

The common names are: Median filtering and Rank filtering. The median filter is normally used to reduce noise in an audio signal , somewhat like the mean filter. However, it often does a better work than the mean filter of preserving useful detail in the audio signal [13].

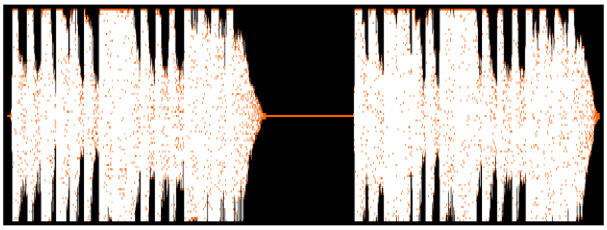
Is a median filter of the alternative methods for filter rate neighborhoods, which replaces the value of intensity for each element value of intermediate levels of intensity in the neighborhood rather than average, and not get the reference improved the application of sliding window are small, with resulting values of the elements of the reference within the window in ascending or descending order, as shown below

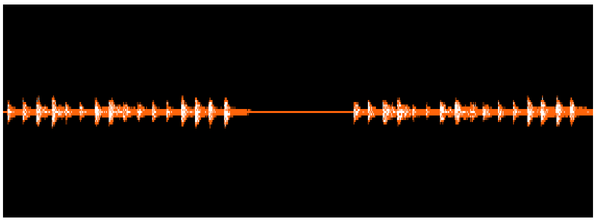
$$I_1 > I_2 > I_3 \text{ --------------- } I_N \text{ ------------ } \quad -8-$$

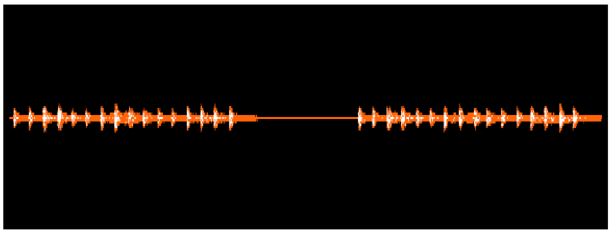
As the (I_1, I_2, I_3, ------, I_N) represents the values of the components of signal intensity within the window that the number of elements n, and then choosing the median value and compensation rather than the central element in the window animation as the median value for a set of elements I_M and the other half from smaller I_M For example, the median value of window size 3 is the second largest value in the window size 5 is the third largest component or a third smaller component [13]

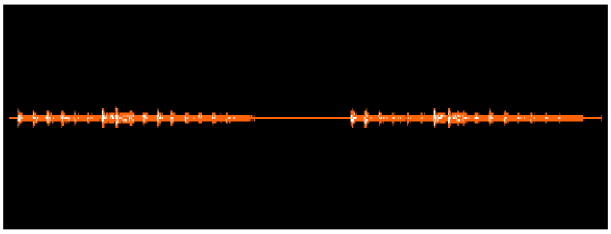
Practical Part

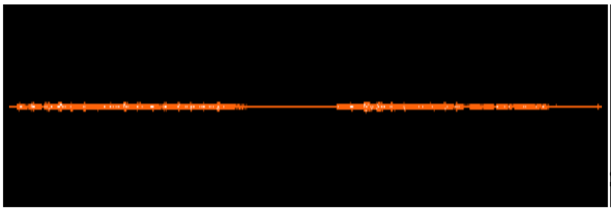
The audio signals shown in Figure (2) use a program to record sound using the microphone of the type C - 300 adoption of a single channel at a rate of (11025 sample / sec) were recorded signal depending on the distance between the source audio and microphone is a Mobile Ringtones Type Nokia where recorded tone depending on the distance for the purpose of comparison between them, and then added noise by 0.5 registered bluer signals of the resulting was then carried out by digital processing using digital filters of traditional and improved ,figure (3) shows the relationship between the average and standard deviation as a function of the distance (0,50,100,150,200,250,300) cm signals that were recorded before the process of adding noise. These signals impure with noise were processed using traditional Mean Filter and the digital filter which was added to improve its operation.

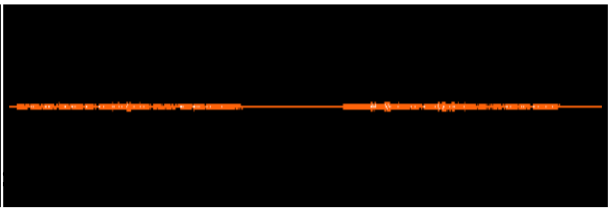

The sound wave where distance = 0 cm

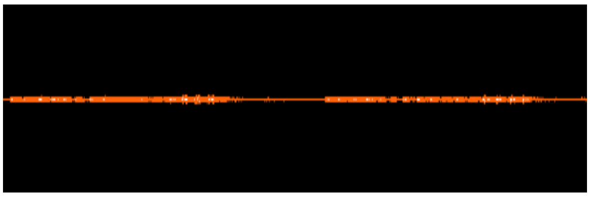

The sound wave where distance = 50 cm


The sound wave where distance = 100cm


The sound wave where distance = 150cm


The sound wave where distance = 200cm


The sound wave where distance = 250cm


The sound wave where distance = 300 cm

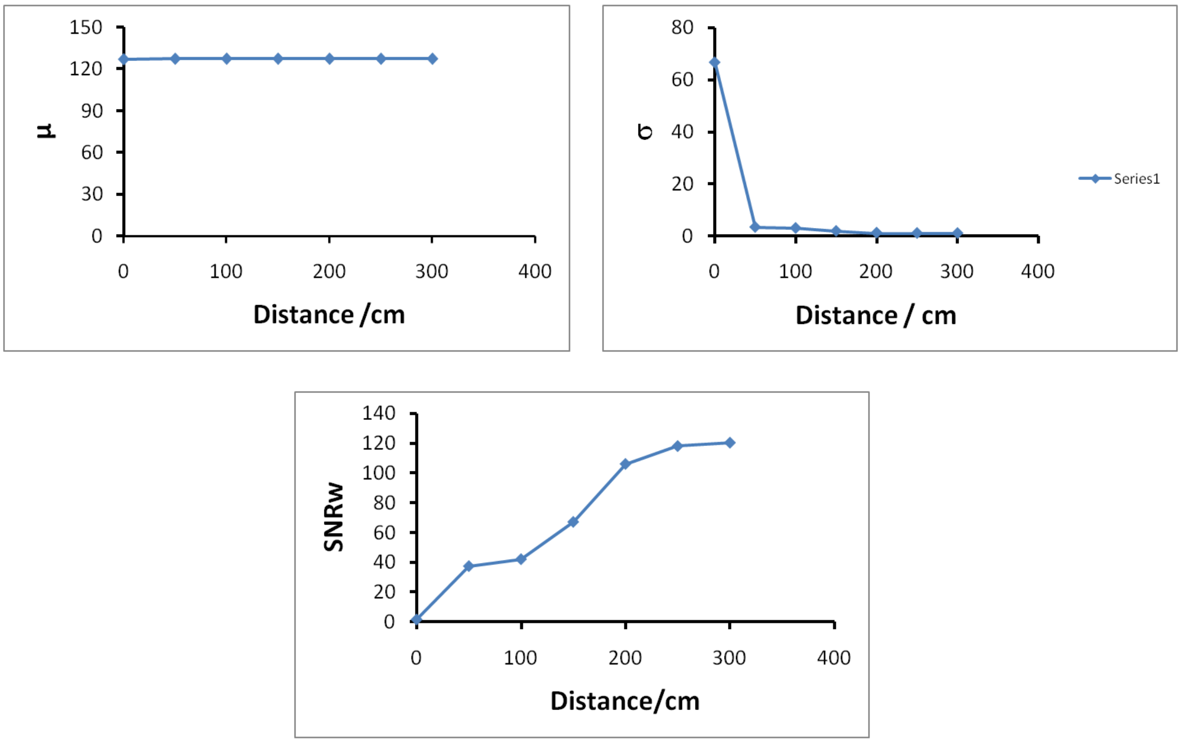


Figure (3) show the relationship between the average and standard deviation as a function of the distance (0,50,100,150,200,250,300) cm

Repetition charts of audio signals

From figures (4), difference in the shape of the repetition chart of both pure and impure with noise signals may be noticed. It seems to be that the curve width of impure signal is wider than curve width of pure signal. This results from the increase in the standard deviation ⌠ due to distortion in audio signal because of adding noise to it. When noticing the repetition charts of ameliorated signals, we may see that repetition chart shape closes to that of the pure signal and that the two windows (7, 9) gives us high smoothing compared with using (3, 5). This doesn't mean that they are the best for that the shape of repetition chart doesn't give the basic feature in smoothing efficiency of signal. It is only a statistical measure of all of the elements of signal which doesn't take its sequence and periodic appearance into consideration. This is the most significant thing in audio signal description.

Repetition charts of pure and impure with noise and ameliorated signals. Standard deviation g⌠ = 5 of Gaussian Noise, it is a value of g⌠ values. The symbols in figures (4, 5) are: [W5 = window 5, W7 = window 7, W9 = window 9].

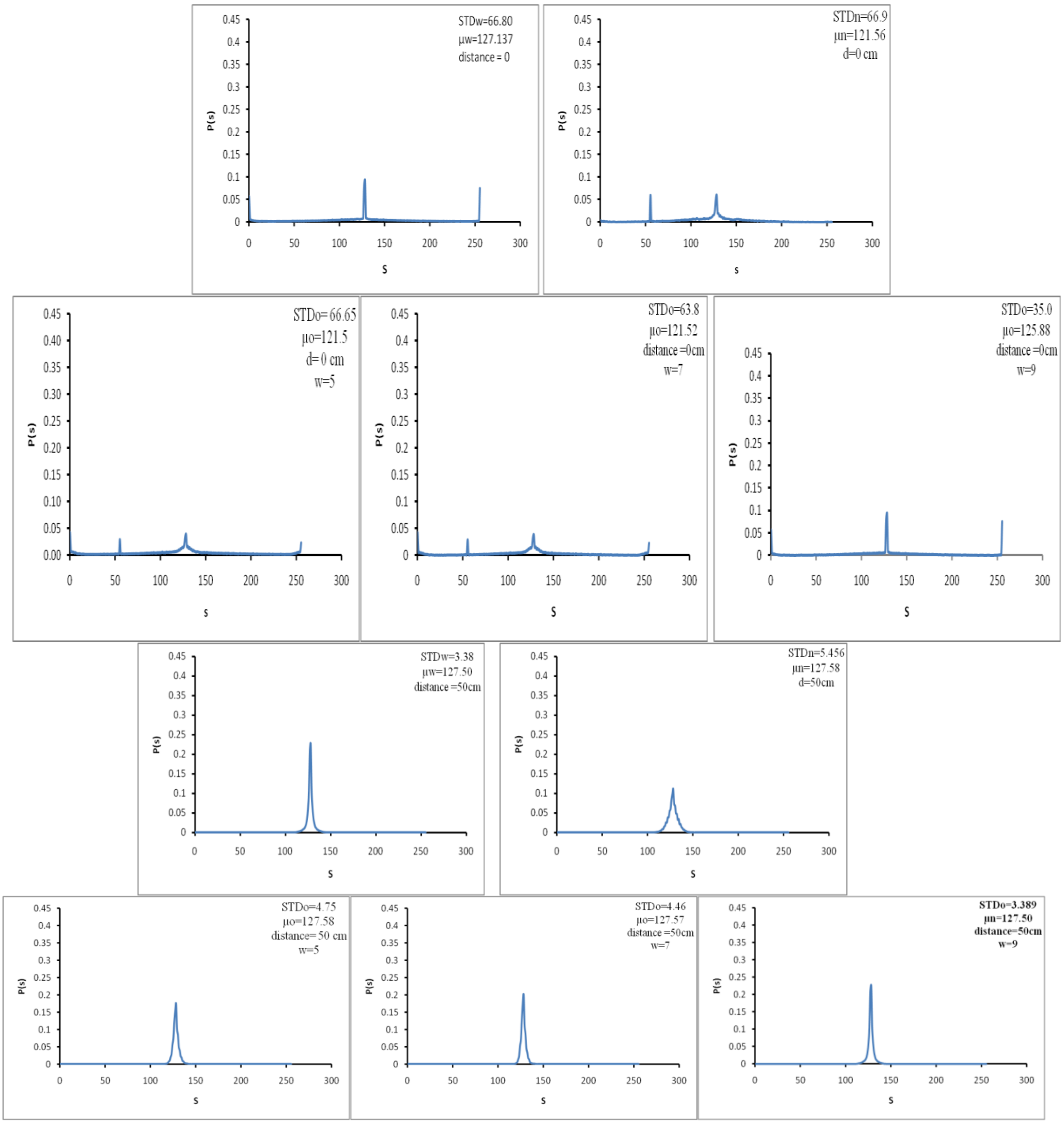
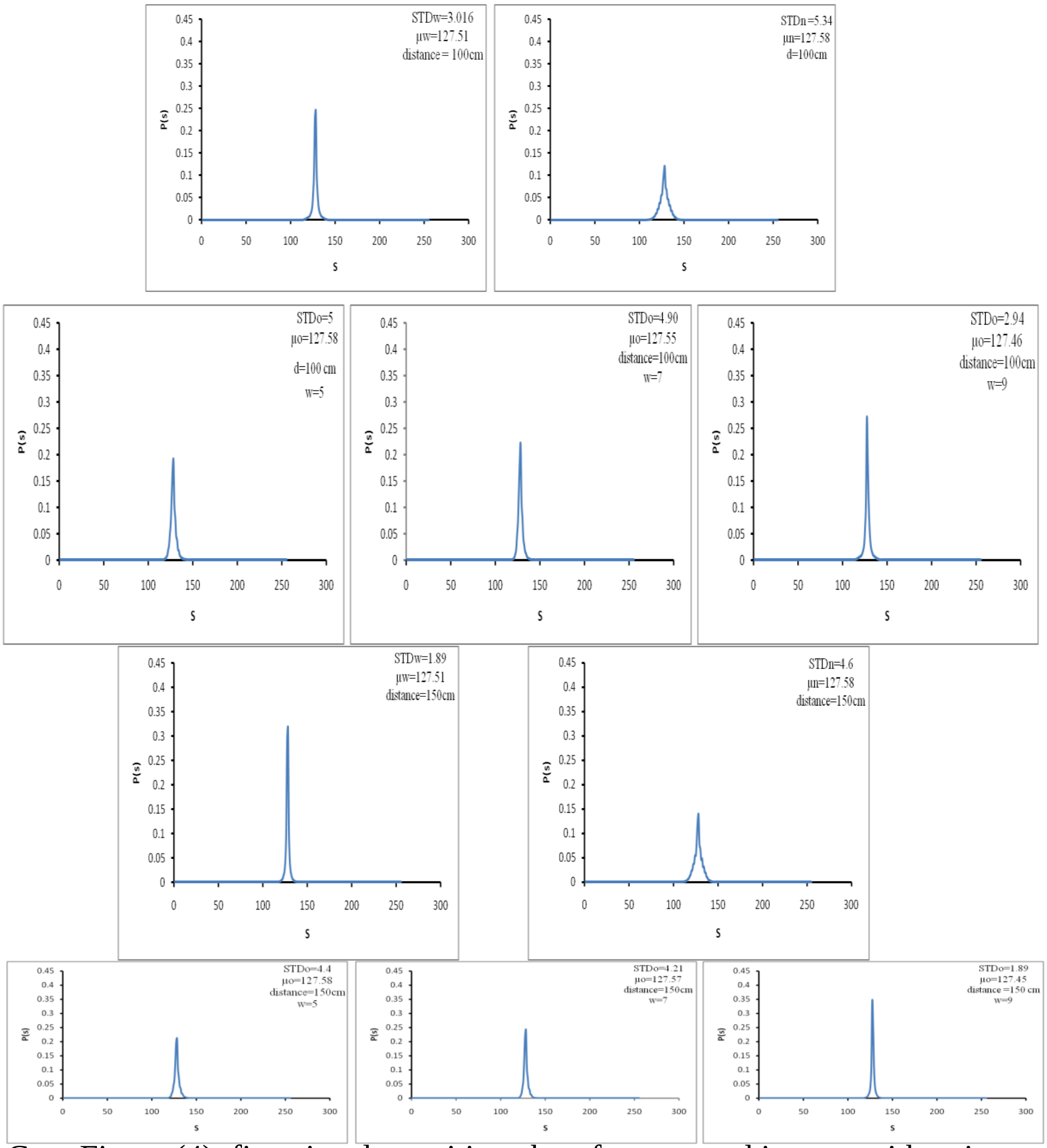
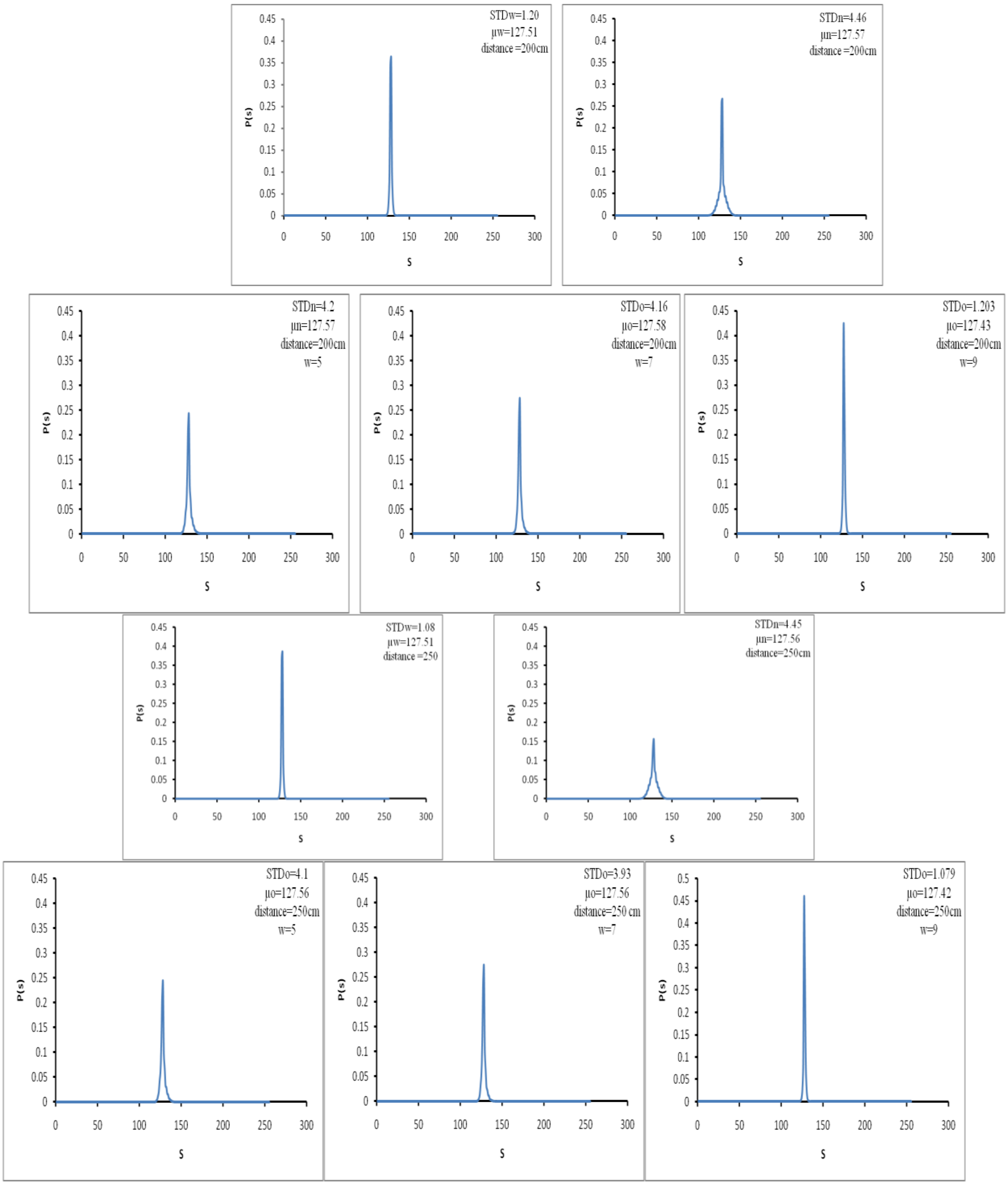


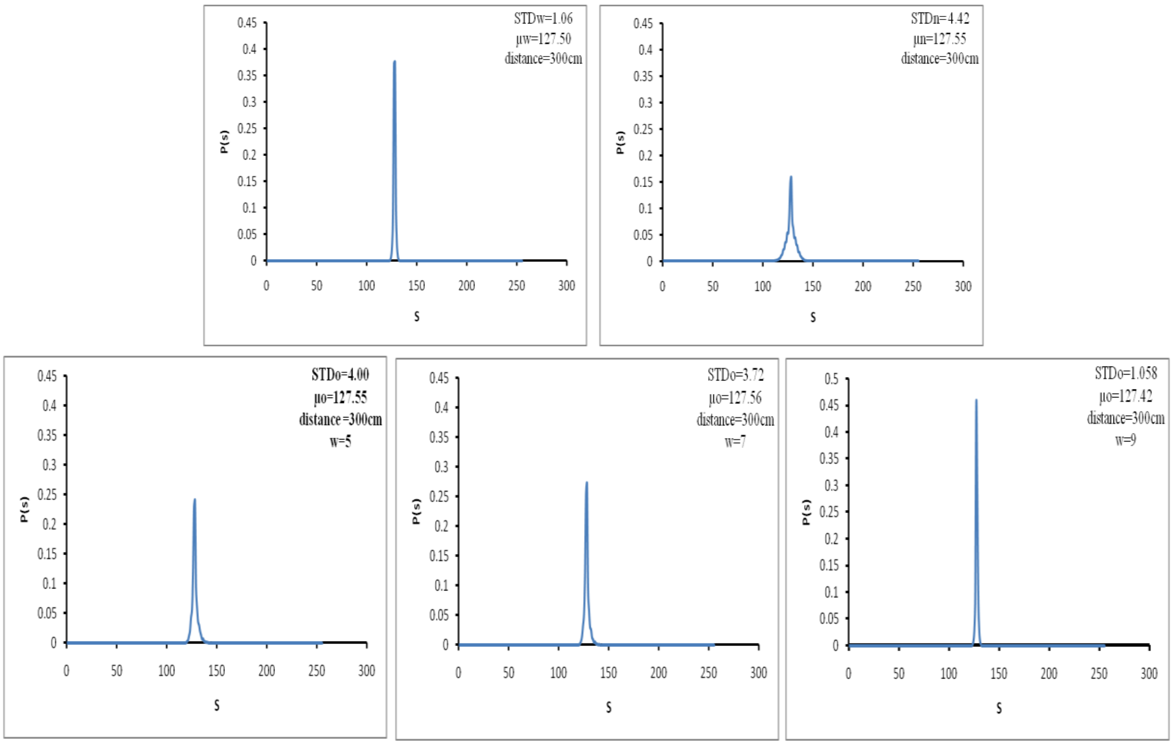
Figure (4): first signal repetition chart for pure and impure with noise and ameliorated signal with distance (0,50) cm and use the window (5,7,9) to process the impure audio signal



Cont Figure (4): first signal repetition chart for pure and impure with noise and ameliorated signal with distance (100,150) cm and use the window (5,7,9) to process the impure audio signal



Cont- Figure (4): first signal repetition chart for pure and impure with noise and ameliorated signal with distance (200,250) cm and use the window (5,7,9) to process the impure audio signal



Cont- Figure (4): first signal repetition chart for pure and impure with noise and ameliorated signal with distance (300) cm and use the window (5,7,9) to process the impure audio signal

Conclusion

The action power of filter appears for both types of audio signals through many standards were adopted to determine the quality of audio signal. After making the digital processing, median filter showed high efficiency in verbal signals processing. This appears clearly from charts of verbal signal

Acknowledgments

The author would like to thank Mustansiriyah University (WWW.uomustansiriyah.edu.iq), Baghdad –Iraq for its support in the present work.

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