Reduction Gaussian Noise of Speech Sound Signals Using Technical Mean Digital filter Traditional and Enhanced

S. S. Salman
Department of Clinical Laboratory Science, College of Pharmacy , University of Baghdad
Received in : 6 October 2010
Accepted in : 13 July 2011

Abstract

Many techniques concerning processes of verbal audio signal were examined aiming to eliminate noise, amelioration and analysis. Imitation for Collective Gaussian Noise using a computer software for different situations of standard deviation of noise (σg =1,2 ….10) and with zero average. This noise was added collectively to verbal signal components, audio signals impure with noise were processed by using mean filter and statistical properties of signals resulted from process in order to determine the quality of signal and filter efficiency. Notably, process leads to lose some significant features of audio signal. Therefore, filter was ameliorated by adding some additional conditions to the components under process in order to keep the significant features of signal. Audio signals impure with noise were processed by using the ameliorated Mean filter and then examining the statistical properties resulted from process. Ameliorated version of Mean filter made good results in improving signal impure with noise. Changes on pure signals were examined through statistical standard i.e. [Mean μ, standard deviationσ, Mean Square Error MSE] of impure, pure and ameliorated signals. Great amelioration was noticed in these characteristics along with keeping significant and distinguishing features of audio signal.

Key word :- Gaussian Noise, Distortion of signals ,Mean filter ,smoothing audio sound

Introduction

Speech is a product of complicated processes. The "verbal message" formed in the brain is transferred somehow into nervous signal, transmitted to audio system which is responsible for sound generation which comes out of mouth generating verbal wave. Verbal signal includes different information about speaker. By means of verbal wave high level of characteristics about the speaker such as accent, sequence, manner of speaker, emotional state and other characteristics[1,2,3]. Verbal wave is series of changes in air pressure which the human ear may pickup as a sound. Usually, digital representation of speech (information included in verbal signal) requires a large number of n bits [4,5]. In audio systems noise may appear, accompanying the audio signal, as a random audio signals or undesired sounds. When processing or analyzing the signal noise is deemed no more the than meaningless data. It is a by-product resulting from the effect of other effects, not from actual signal. It is a data enters with signal data causing the deviation, penetration an change in the content of carried message in the digital signal. Whereas signal is transmitted from receiver to transmitter, undesired effect may take place which weaken the signal. These effect is defined as follows [6]:

1. Distortion
2. Interference
3. Noise:
4. Gaussian Noise : Gaussian distribution and given by the following formula [7-8]:

$$f(N) = \frac{1}{\sqrt{2\pi}\sigma_N} \exp - \frac{(N - \mu_N)^2}{2\sigma_N^2} \quad \ldots\ldots\ldots\ldots\ldots(1)$$

Whereas N represents the Random parameter which is distributed in natural distribution (represents noise element).

μ_N :- represents the arithmetic mean of random parameters (mean of N parameter)

σ_N :- represents standard deviation of random parameters around the mean μ_N (standard deviation of noise)

Consequently, for the purpose of solving this problem, we have examined optimal filters in order to eliminate noise and then to reduce noise with quantitative standards and new methods.

In 2004, Ahmed Humood Faliyih have examined speaker recognition systems, identifying engines sound and recognized separated words of fixed text within a closed group starting from reference database represented by reference layers of sounds [9].

In 2005, Ahmed Kamil conducted a study to eliminate noise form sound using wavelet transform and other different procedures to eliminate natural noise. He adopted a procedure to separate vocal syllable non-vocal syllable [10].

In 2007, Noor Ali Hassen conducted a study to reduce noise of audio signal using techniques applied of digital signal on the basis of wavelet transform. Three types of wavelet transform and five types of threshold procedures were used. Their efficacy of reducing noise form two types of audio signal were examined. First type was of high amplitude and other with low one [11].

Sound and Accompanying Noise

Digital Smoothing Filters

Digital smoothing filters is used, primarily, to eliminate the fake effects that may be present in signal such as noise resulting from signal coding or errors accompanying the recording, transmission and receiving process.

Traditional filters are one of the most simple techniques in signal smoothing. It may be applied directly on the impure signal, without any need to have previous knowledge of statistical characteristics which controls the intensity in signal regions. Most significant filters are Mean Filter and Median Filter. The efficiency of these filters are obvious in suppressing noise effect in uniform regions of signal [12,13].

Mean Filter

This filter operates using small movable window parallel to the signal length. Smoothing signal is obtained by taking the mean values of the signal content included slide window and replacing the valued central window element with mean. In other word, the ameliorated signal is obtained through the following formula [12]:

$$S(t) = \frac{1}{N} \sum_{i=-\frac{N-1}{2}}^{\frac{N-1}{2}} NOS\ (t+i) \quad \ldots\ldots\ldots\ldots\ldots(2)$$

Whereas $S(t)$ represents the mean value of the element n the window (resulted ameliorated element)

NOS (t + i) represents the value of signal element in the window (signal element impure with noise).

Statistical Characteristics of Audio Signal

Processing the digital audio signal is based, basically, on the information included in the signal and the way of information distribution. This information, sensed according to the physical principle, is subject to mathematical modeling and statistical distributions. Therefore, signal statistics are most significant in the process of analyzing processing the digital signal. Also, these statistics shows features and attributes of signals nature and how to distribute the information inside them.

Mean (μ): mean of intensity is represented by (μ) which is statistical term which denotes to mean value of signal. It comes from sum of all samples of audio signal, divided by the total number of samples N and the mean μ is calculated from the following relation [13-15]:

$$\mu = \frac{1}{N} \sum_{t=0}^{N} I(t) \quad \ldots\ldots\ldots\ldots\ldots (3)$$

whereas $I(t)$ is a general sign

N is total number of signal elements

Standard Deviation (σ): is the amount of signal deviation of mean. Standard Deviation (σ) is calculated from the following relation [13-15]:

$$\sigma_s = \sqrt{\frac{1}{N} \sum_{t=0}^{N} (I(t) - \mu)^2} \quad \ldots\ldots\ldots\ldots\ldots(4)$$

Mean Square Error (MSE): is accumulative square error between output signal and original signal. MSE is on of the standards through it the signal quality is identified. It is calculated from the following relation [13-15]:

$$\text{MSE} = \sum_{t=0}^{N} [O(t) - f(t)]^2 \frac{1}{N} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots (5)$$

Whereas $O(t)$ represent original signal data

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

Probability Density Function (PDF)

Is explained by the formula p (I) which represents probability density distribution of signal elements I (t). The function I (t) takes the range ∞ < I (t) < - ∞. The probability is confined with the relation 0 < p(I) < 1, the relation between I (t) and p (I) is known as signal Histogram [13-15].

The following audio signal along with their classifications:

A- Signals free form noise $O(t)$

B- Signals impure with noise NOS (t): are the signals which Gaussian noise n(t) was added to for different states of standard deviation (σg = 1,2,3,4…..,10) according to the following formula:

$$NOS(t) = O(t) + n(t) \qquad \ldots\ldots (6)$$

C- Signals processed by Mean Filter, denoted with S(t), which represents the ameliorated signal.

Algorithm of mean digital filter Operation, Traditional and Enhanced

The traditional smoothing filter was adopted to eliminate the collective Gausian noise from audio signal. However, this filter mostly causes a smoothing operation at significant parts of signal which leads to a deformation in audio signal data. Therefore, we directed ourselves in the work to enhance filter operation through determining additional conditions to improve the operation of this filter. Let's assume that figure (1) represents a waveform of audio signal, the center line is zero of audio signal. The elements over zero line are equal to 128, they represent still moments, while the elements under zero line are less than 128, they represent sounds with high amplitude. Form figure (1) we note that the point (1,6) may be processed using the traditional modifier filter and it doesn't cause deformation of audio signal. We may see the significant difference in values before and after process when applying traditional modifier filter to process the points 2, 3 and 4, this may change the common characteristics of audio signal and leads to loss of primary points of audio signal, i.e. points high and low. Hence, the traditional modifier filter was improved by adding additional conditions of enhancement (showed in filter algorithms) to avoid such problem, since the conditions may exclude such points from processing and will process the other points of audio file . Most of algorithms used in the work are as follows:

algorithm mean filter

Input:

- noisy sound signal NOS(t) t=1,2,3……………..J
  J= total number of elements
- filter block size (w) =1,3,5.7 …………

output:

- smoothed sound signal S(t)

1- drag the audio signal which added noise Gaussian to processing.
2- Take the number integer for the result of dividing the number of elements of a window processing
   w1 = w/2.
3- select the first element of the audio signal being processed by the following
   w2 = w1 +1.
4- Selected the last element in the audio signal which is processed using filters according the equation follow j1=j-w1 .
5- Open loop for to scan the elements of the audio signal being processed by the window which is used as follows
   For k = w2 to j1 .
6- Select the first and last element for the window which is used for processing as follows
   t1 = t -1 , t2 = t +1
7- Put value initial for sum of the elements used in the Processing window .

K=0.

8- drag the audio signal which added noise Gaussian to processing.
9- Take the number integer for the result of dividing the number of elements of a window processing
   w1 = w/2.
10- select the first element of the audio signal being processed by the following
   w2 = w1 +1.
11- Selected the last element in the audio signal which is processed using filters according the equation follow j1=j-w1 .
12- Open loop for to scan the elements of the audio signal being processed by the window which is used as follows
   For k = w2 to j1 .
13- Select the first and last element for the window which is used for processing as follows
   t1 = t -1 , t2 = t +1

14- Put value initial for sum of the elements used in the Processing window .
   K=0
15- Open loop for elements window , according to the following equation
   For t3=t1 to t2
16- Accumulation of elements of the window in S
   S(t) = k + N(t3)
17- Calculating average sum elements of the window which represents the value of the element central to processing according to the following Cint = k / w
18- End loop (Next)
19- Save audio signal
20- End the loop for the window processing (Next)

adaptive mean filter algorithm

Input

- noisy sound signal NOS(t) t=1,2,3……………..J
  J = total number of elements
- filter block size W =1,3,5.7…………..
- filter optimization threshold Th1 , Th2

output:

- smoothed sound signal S(t)

Procedures

1- Input the noisy audio signal with noise Gaussian.
2- Input thresholds for filter .
3- Select the size of filter window .
4- Calculate value of displacement for window from right and left for signal as follow :
   W_1 = int (w/2).
5- Calculating sequence of the last element in audio signal to being processed j_1=j-w_1 .
6- Select sequence of the central element for the window as follow w_2=w_1 +1.
7- open for loop to scan elements of audio signal which beginning from w and ending j as follow
   a- determine the beginning of window b – determine the end of window
8- put value initial for sum of elements window

9- open loop for elements of the window from t_1 to t_2 as follow
for $t_2 = t_1$ to t_2
10- Accumulation value the elements for the window $s = s + Nos(t_3)$
11- End the loop in step 9 .
12- Calculating average elements of window according the equation
Av = Round _ Integer (s/w)
13- Select the central element for the window $a_1 = Nos(t_3)$.
14- Determine condition modify for the process as follow
- If the value difference between the central element and the level zero for the audio signal (level zero = 128) greater than threshold (Th_1) we apply the condition for adaptive filter which mean if $|a_1 - av| > Th_1$ then.
- If the difference between the central element and the average greater than the threshold (Th2) replace the average to the central element as following
If $|a_1 - av| > Th_2$ a1 = av
- If the value of the central element of the largest of the value of the element before and after that we av = a_1 in other word if $a_1 > Nos(t-1)$ and $a1 > Nos(t+1)$ then av = a1
- If the value of the central element of the smallest of the value of the element before and after that we av = a_1 in other word if $a_1 < Nos(t-1)$ and $a1 < Nos(t+1)$ then av = a1

15- End the if statement in step 14.
16- value of audio signal improved will be as follows :
$S(t) = av = a_1$
17- End loop in step 7 .

Practical Part

Verbal audio signal (sound syllable consisting of two words), illustrated in Figure (2), was recorded. This signal was impure with Gaussian Noise which was generated in laboratory by using a computer software with Visual Basic language. Random noise imitating the natural noise was generated according to mathematical models i.e. "Gaussian Noise", with a zero average (gμ = 0) and different values of standard deviation (σg = 1,2,3,…..,10). Te noise (noise files) generating the audio signals was added in order to obtain audio signals impure with noise. These signals impure with noise was processed using traditional Mean Filter and the digital filter which was added to improve its operation:

The type of audio files dealt with in this study was (wave, 8 bit). Wave file is general file known by Microsoft Company which is used currently in Widows operating systems. Computer software was designed to regulate the filter operation and to find statistical characteristics of audio signals in order to compare between the results of signals impure with noise and those ameliorated.

Results and Discussion

Gaussian Noise was added to the audio signal for different states of standard deviation (σg = 1, 2, 3, 4 ….., 10). The noise was added to 100% of audio signal elements. The same amount of noise is added to each element of audio signal elements. In processing, the widows (3, 5, 7, 9,) were adopted for each value of σg values of noise added to signal. The filter efficiency and audio signal quality resulted from processing can be evaluated through examining the chart of statistical parameters as shown in figures (3, 4).

Using traditional digital mean filter M leads to obvious amelioration in the verbal audio signal and this amelioration is accompanied by loosing some significant features of verbal signal. The mentioned conditions given in algorithms (1, 2) for ameliorated mean filter AM have made better results for processing in terms of keeping the significant features in signal. The AM filter kept the significant heights and drops in the audio signal which lead to good percentage of amelioration. Chart relations were drawn between standard deviation of Gaussian Noise added to the signal σg and the following parameters ($s\mu$, σs, MSEs) of signals resulted from processing using the traditional digital and ameliorated filters. The filter efficiency and audio signal quality resulted from digital processing can be evaluated through examining the charts of these statistical parameters.

Amelioration Results Using Mean Filter M

The melioration Results for verbal audio signals using Mean Filter M seemed to be good. It is obvious from examining the statistical parameters of signal from the figure (3) that the filters has, notably, kept the mean intensity of audio signal μS and for the all of the widows used in the processing.

Notably, from figure (3) using mean filter in processing has reduced the resulted increase in standard deviation σs for the signal impure with noise. Using the windows (7, 9) has made better results than using the windows (3, 5). For the MSEs, it is obvious that using the filter has reduced its resulted increase in the signals impure with noise. Form figure (3) it may be noticed that using the windows (7, 9) has made better results than using the windows (3, 5). It may be noticed form figure (4) that using the filter to ameliorate the second verbal audio signal made similar results similar to those obtained from processing first verbal audio signal.

Amelioration Results Using Ameliorated Mean Filter AM

This filter is deemed better that traditional mean filter. This is obvious from examining all statistical parameters of audio signal. Form figure (3) it may be noticed that AM filter has kept the mean intensity of audio signal μS. this means that the audio power of signal was kept. However, form figure (3) it may be noticed that AM filter has notable reduction the resulted increase in standard deviation (σ) resulting from adding to noise it. Using windows (5, 7, 9) made better results than using the window 3. this notable for all added noise percentages. Form the figure (3), using AM filter has reduced the resulted in crease in MSEs. The filter results of windows (7, 9) were better than those of windows (3, 5). The figure (4) shows that amelioration of the two verbal signal was the same.

Repetition charts of audio signals

From figures (5, 6), 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 σ 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 σg = 5 of Gaussian Noise, it is a mean value of σg values. The symbols in figures (5, 6) are: [W3 = window 3, W5 = window 5, W7 = window 7, W9 = window 9].

Results of deduction of uniform parts of verbal audio signal

Uniform intensity parts (silent regions) of verbal audio signal were deducted from verbal audio signal in order to examine the effect of processing with traditional digital and ameliorated filters in audio signal impure with noise. The parameters (σR, μR, SNRR) for the deducted parts of pure, impure with noise and ameliorated signal. The results of three signals were compared in order to know approximation of these parameters of ameliorated signals from original signal those of original signal.

From the tables (1, 2), notably, that the M and AM filters operated with efficiency. This is obvious from improvement in σR and SNRR of ameliorated signals comparing with those of impure "with noise" signal. Also, is was noticed that μR value of pure, impure and ameliorated signals was fixed. This doesn't represent an exact and efficient measure in examining the quality of filtration process. It may not be used alone to identify the efficiency of filtration process because since it takes into consideration the uniform parts of signal or regions that represent silent moments only.

The optimal filter is that filter that make a high smoothing since these regions don't include significant information. However, these results show how thwe filters are powerful in eliminating noise in uniform regions or those that don't include intensive and more fluctuations in signal

Conclusion

The ameliorated mean filter AM occupied the first rank out of the four digital filters. 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, mean filter M showed high efficiency in verbal signals processing. This appears clearly from charts of verbal signal. This is that the verbal signal include intervals as well as some of the word letters are audible, with high amplitude while other are not audible, with low amplitude.

References

1- Fontaine, Jean-Marc (2002),Sound recording Evolution through its history some stages University Paris 6-CNRS - MCC -Laboratory' Acoustique Musicale E-mail : jmfontai@ccr.jussieu.fr .
2- Kintzel, T. (1998), A programmers Guide Sound , Addison -Wesly , England ,
3- Plataniotis, K.N.D.; Androutsos, A.N.and Venet Saopoulos, (1998),Multi channel filter for color image" , signal processing, image communication , 11:171-177.
4- Ahmed Kamil Hassan ,(2005), Speech Denoising Using Wavelet Transform,M.Sc. thesis , Electrical Engineering Department , College of Engineering , Mustansiriya University .

5- Alan, H.(2008),Sound Nature and Measurement , Report , Cornell University, January ,.Kolar and Kishnap " speech Compression " .

6- 6.بوش،أساسيات الفيزياء (1998) , الطبعة السابعة الدار الدولية للنشر والطباعة

7- Nabeel Mubarak Meerza Al-Dalawy (2008),A study of TV images quality for channels Broadcast Television Terrestrial" M.Sc. thesis , Physics Department , College of Science , AL-Mustansiriyah University .

8- عبد الله حمود محمد الجبوري، (2006) , دراسة الضوضاء المرافقة لصور البث التلفزيوني الأرضي وطرق تقليل . تأثيرها ، أطروحة دكتوراه \ قسم الفيزياء \ كلية العلوم \ الجامعة المستنصرية.

9- Tariq Ahmed Al-Amree ,(2005), Speaker Identification Using Wavelet Transform And Probabilistic Neural Network" M.Sc. thesis , Computers Department , collage of sience , Al-Mustansiriyah University .

10- Al-Saisia ,A. H. F. (2004), voice and speech recognition systems. A thesis submitted to the collage of science in Al- Mustansiriyah University .

11- Plataniotis, K.N.D.; Androutsos, A.N.and Venet Saopoulos, (1998) , Multi channel filter for color image,signal processing,image communication,11:171-177.

12- Gonzalez , R.C. and Wintz, P. (1987)Digital image processing , Addison-wistey.
Niblack, .W.(1986),An Introduction to digital image processing, prenticeHall

13- International (UK) LTD .

14- Mukul v.shiraikar (2004),An optimal measure of camera focus & Exposure, electrical engineering Department ,university of Texas.

15- CH 2 Steven W. Smith ,(2006), Statistics, Probability and Noise, Ph.D. Thesis , scientist and Engineers Guide to Digital Signal Processing .

Table (1): Results of deducting uniform regions of first signal

Standard deviation Gaussian σg	Window for processing	type	Mean of region μ r	Standard deviation region σr	The ratio of Signal to Noise SNRr	Standard deviation Gaussian σg	Window for processing	type	Mean of region μ r	Standard deviation region σr	The ratio of Signal to Noise SNRr

R1	R1	SIGNAL	130	0.87	149.23	R2	R2	SIGNAL	130	1.24	104.27
2		NOISY	130	1.96	66.20	2		NOISY	130	2.33	55.75
2	3	M	130	1.58	82.48	2	3	M	130	1.88	69.33
2	3	A M	130	1.58	82.48	2	3	A M	130	1.88	69.33
2	5	M	130	1.39	93.90	2	5	M	130	1.71	75.98
2	5	A M	130	1.39	93.90	2	5	A M	130	1.71	75.98
2	5	AMD	130	2.06	63.26	2	5	AMD	130	2.37	54.91
2	7	M	130	1.26	103.46	2	7	M	130	1.61	80.82
2	7	A M	130	1.26	103.46	2	7	A M	130	1.61	80.82
6		NOISY	130	5.78	22.53	6		NOISY	130	6.17	21.09
6	3	M	130	3.93	33.14	6	3	M	130	4.36	29.86
6	3	A M	130	3.93	33.14	6	3	A M	130	4.36	29.86
6	5	M	130	3.33	39.12	6	5	M	130	3.76	34.57
6	5	A M	130	3.33	39.12	6	5	A M	130	3.76	34.57
6	7	M	130	2.99	43.56	6	7	M	130	3.42	37.99
6	7	A M	130	2.99	43.56	6	7	A M	130	3.42	37.99
10		NOISY	129	9.68	13.34	10		NOISY	129	8.88	14.56
10	3	M	129	6.47	19.97	10	3	M	129	6.15	21.04
10	3	A M	129	7.25	17.83	10	3	A M	129	6.61	19.55
10	5	M	129	5.38	24.02	10	5	M	129	5.22	24.76
10	5	A M	129	6.39	20.20	10	5	A M	129	5.82	22.21
10	7	M	129	4.82	26.80	10	7	M	129	4.63	27.97
10	7	A M	129	6.03	21.43	10	7	A M	129	5.30	24.39

Table (1): Results of deducting uniform regions of first signal

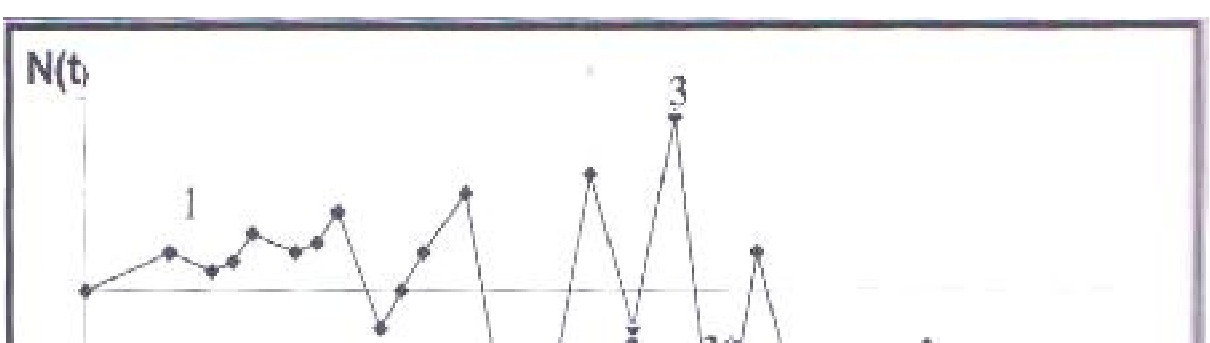
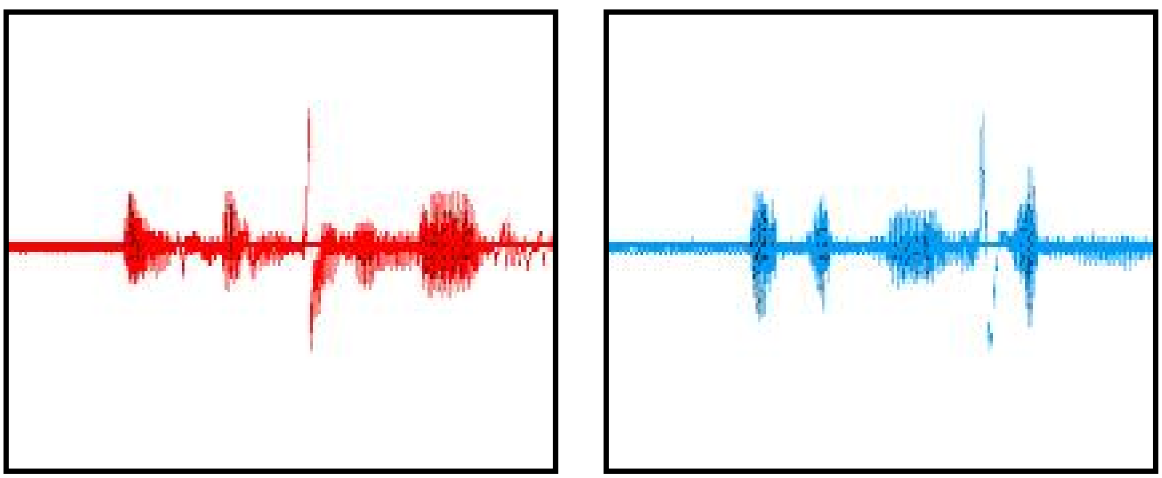


Fig. (1): Represents a waveform of audio signal



First verbal audio signal: (Alhamdu Lillah) Second verbal audio signal: (Arrahman Arraheem)

Fig. (2): Verbal audio signals used in this study

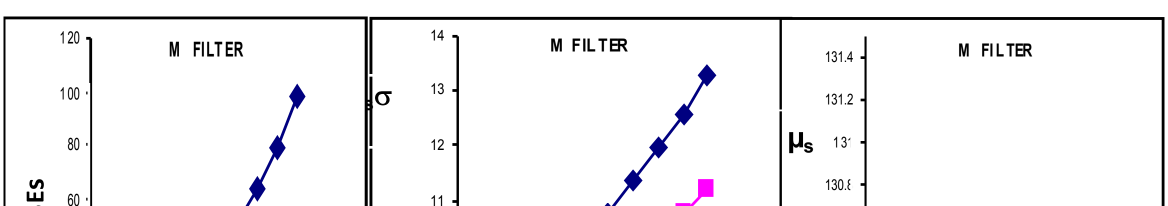


Fig. (3): Amelioration results of statistical standards adopted in study – first signal

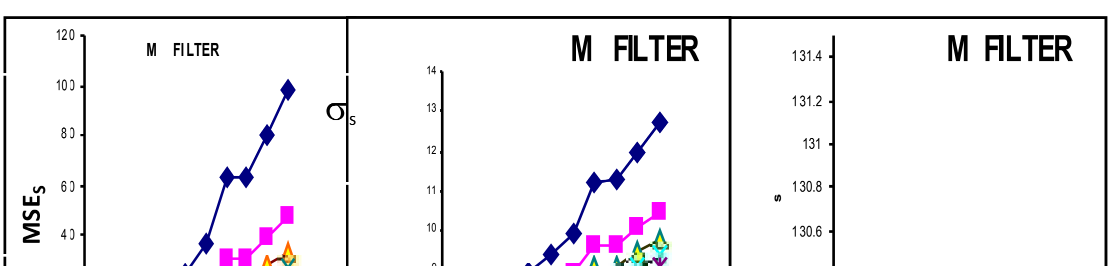


Fig. (4) Amelioration results of statistical standards adopted in study – second signal

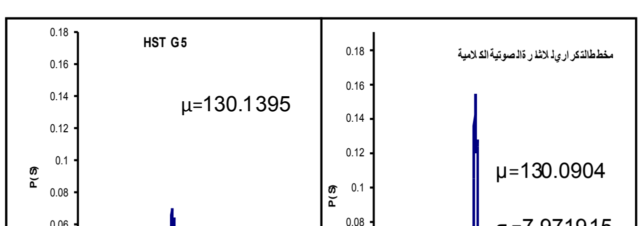
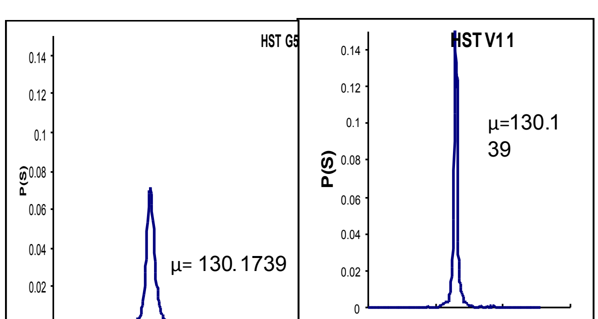


Fig. (5): First signal repetition chart for pure and impure with noise and ameliorated signal



Standard deviation Gaussian σg	Window for processing	type	Mean of region μ r	Standard deviation region σr	The ratio of Signal to Noise SNRr	Standard deviation Gaussian σg	Window for processing	type	Mean of region μ r	Standard deviation region σr	The ratio of Signal to Noise SNRr

Fig. (6): Second signal repetition chart for pure and impure with noise and ameliorated signal

R1	R1	SIGNAL	130	2.00	64.92	R2	R2	SIGNAL	130	1.38	94.19
2		NOISY	130	2.82	46.14	2		NOISY	130	2.47	52.77
2	3	M	130	2.43	53.47	2	3	M	130	1.98	65.95
2	3	A M	130	2.43	53.47	2	3	A M	130	1.98	65.95
2	5	M	130	2	56	2	5	M	130	2	72
2	5	A M	130	2.31	56.35	2	5	○AM	130	1.81	72.14
2	7	M	130	2.23	58.17	2	7	M	130	1.71	76.06
2	7	A M	130	2.23	58.17	2	7	A M	130	1.71	76.06
2	9	M	130	2.17	59.75	2	9	M	130	1.66	78.69
2	9	A M	130	2.17	59.75	2	9	A M	130	1.66	78.69
6		NOISY	130	6.29	20.67	6		NOISY	131	6.36	20.52
6	3	M	130	4.57	28.46	6	3	M	131	4.50	28.99
6	3	A M	130	4.58	28.39	6	3	A M	131	4.58	28.52
6	5	M	130	3.95	32.90	6	5	M	131	3.87	33.75
6	5	A M	130	3.97	32.75	6	5	A M	131	3.95	33.03
6	7	M	130	3.62	35.92	6	7	M	131	3.49	37.41
6	7	A M	130	3.64	35.73	6	7	A M	131	3.59	36.38
6	9	M	130	3.40	38.26	6	9	M	131	3.25	40.13
6	9	A M	130	3.42	38.02	6	9	A M	131	3.36	38.85
10		NOISY	130	10.18	12.76	10		NOISY	130	9.91	13.15
10	3	M	130	7.30	17.81	10	3	M	130	6.97	18.70
10	3	AMD	131	9.60	13.60	10	3	AMD	131	9.27	14.09
10	5	M	130	6.34	20.49	10	5	M	130	5.94	21.94
10	5	A M	130	7.53	17.27	10	5	A M	130	7.03	18.54
10	7	M	130	5.76	22.56	10	7	M	130	5.34	24.38
10	7	A M	130	7.10	18.31	10	7	A M	130	6.60	19.74
10	9	M	130	5.38	24.17	10	9	M	130	4.93	26.40
10	9	A M	130	6.82	19.06	10	9	A M	130	6.36	20.51

Table (2): Results of deducting uniform regions of second signal

اختزال الضوضاء الكاوسية من الاشارة الكلامية
باستعمال تقنية المرشح الرقمي المعدل التقليدي والمحسن

سليمة سلطان سلمان
فرع العلوم المختبرية السريرية ، كلية الصيدلة، جامعة بغداد

استلم البحث في:6 تشرين الاول 2010
قبل البحث في: 13 تموز 2011

الخلاصة

درس العديد من تقنيات معالجة الإشارة الصوتية الكلامية لغرض إزالة الضوضاء منها وتحسينها وتحليلها عملت محاكاة للضوضاء الكاوسية الجمعية باستخدام برنامج حاسوبي ولحالات مختلفة من الانحراف المعياري للضوضاء

(10.... σ_g =1,2) وبمعدل صفري واضيفت هذه الضوضاء بشكل جمعي لعناصر الإشارة الكلامية وتمت معالجة الإشارات الصوتية المشوبة بالضوضاء باستعمال مرشح المعدل، و دراسة الخصائص الإحصائية للإشارات الناتجة من المعالجة لتحديد جودة الإشارة بعد المعالجة وتحديد كفاية المرشح ولوحظ إن عملية المعالجة تؤدي إلى فقدان بعض السمات المهمة للإشارة الصوتية ، لذا حسّن المرشح بإضافة بعض الشروط الإضافية على العنصر الذي تتم معالجته للمحافظة على السمات المهمة للإشارة وتمت معالجة الإشارة المشوبة بالضوضاء باستعمال المرشح المعدل المحسن ومن ثم دراسة الخصائص الإحصائية للإشارة الناتجة من المعالجة وأعطى مرشح المعدل بصيغته المحسنة نتائجا" جيدة في تحسين الإشارة المشوبة بالضوضاء ودرست التغيرات التي طرأت على الإشارات النقية من خلال المعايير الإحصائية وهي [المعدل μ ، الانحراف المعياري σ، ومعدل مربع الخطأ المعياري MSE] للإشارات المشوبة والإشارات النقية والمحسنة، اذ لوحظ تحسن كبير في هذه الخصائص مع المحافظة على السمات المهمة والمميزة للاشارة الصوتية .

الكلمات المفتاحية : – الضوضاء الكاوسية ،تشويه الاشارة ، مرشح المعدل ، تنعيم الاشارة الصوتية