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Designation and Simulating the FFSK Algorithms and Patch Work for Watermarking Data of Acoustic Systems in order to Increase the Channel Capacity and Reduce the Error by Matlab Software

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ABSTRACT

Digital watermarking in the Telecommunication systems especially acoustic systems has a significant role that causes to create security and improving the data and reducing the error rate. After watermarking ,compression operations will be done on data and significant point is this that the receiver system for the purpose of disclosing needs data before Digital watermarking.

In this paper , two corrected algorithms of FFSK and Patch work in the area of time which can be helpful in digital watermarking , will be studied and simulated . Spread Spectrum algorithms in the scope of frequency (FSK) can be in two forms of F-FH (FFSK) and S-FH(SFSK) , of course with pre-emphasis filter which causes to increase the channel capacity is considered here .

Also , modified patchwork in the area of time with LPF which is considered to reduce the error significantly . in this paper , we will review and simulate these two different method in order to watermark the data using the Matlab software and show and prove their useful results in reducing the error rate and increase of the channel capacity and network .

KEYWORDS : FFSK, Patch work , watermarking , channel capacity.

INTRODUCTION

Information security and data optimization and reducing the error rate as one of the main and most important parts in the telecommunication systems specially in the acoustic systems will be counted . one of the ways in creating the security in the acoustic systems , is the data watermarking in speech signals.

In this paper , two significant algorithms i.e, FFSK and corrected patchwork in the field of time to watermark the data were introduced and studied and consequently , we will investigate and simulate their results and impact rate in reducing the error and increasing the channel capacity and network security.

Coding is one of the common methods used in preparation of security and preventing from unauthorized access to the special and secret information. The best way to code the data and creating the security is use of wide spectrum techniques and regarding to the speech signals span and compression operation at the time of posting , watermarking algorithms of data should have the best resistance against the compression [3].

Activities which are being done in the wide span spectrum , also will consider the data compression and one of the reasons id selecting the wide span spectrum algorithm in this paper is the capability of wide span algorithms in the scope of time and frequency. It is worth to be mentioned that in this paper for exposing the original signal of speech , there is no need for the original signal and exposing by ICA technique will be done [2].

In this paper , one technique in the scope of frequency (FFSK) with the pre-emphasis filter for increasing the channel capacity and one algorithm in the scope of time (corrected patchwork) with LPF for recusing the error will be studied and investigated . Moreover , two watermarking techniques with algorithms and their relations will be reviewed.

2- Watermarking using the FFSK algorithm

One of the wide span spectrum systems is based on the frequency hopping (FF/SS) . frequency hopping is the orderly change of carrier frequency between a set of frequencies based on a pattern that a quasi-random signal will propose. In this case , there is no need, the PN-sequence to be composed of ± 1 values. This sequence in the frequency hopping systems only will control the hopping pattern.

Hopping rate can be faster or slower than the data sending rate that the first case in known as the fast frequency hopping and the second case is known as the slow frequency hopping . Time interval of each bit is supposed to be **T** and usually for this systems is used of FSK .due to a fast change of the carrier frequency , we cannot use of the **co-phase**modulation , therefore we suppose that system uses of the **cross-phase** modulation.

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As we know , data compression causes the distortion in the signal to be produced . according to the reviews which were done , distortion amount compared to the compression in the field of frequency is very low than the time field and because of this we emphasize that the wide span spectrum algorithms were used in the field of frequency this paper. [7]. Also , it is needed to be mentioned that , the desired data will be distributed on DFT4 coefficients . moreover, coding and decoding of these systems will be assessed .

2.1 speech signal is composed of two parts as with data and without data (silence).

Therefore watermarking only for the part having data or in other words for the part having the speech will be selected .In this case to identify the part which includes data, speech signal must be converted to the frames in length of N and for each frame , the absolute value of frame sizes must be calculated and then the result must be compared with the Threshold (T), if the result to be greater than or equal with the threshold , the data is available ,otherwise the speech signal is not available . Data sizes using the equation (1) and will be calculated as the following:

1)$\sum_i^N |S|] > T[\frac{1}{N}$

The next stage is the watermarking that this function based on the wide span spectrum technique of fast frequency hopping will be done , in this case the random strings will be indicted by U and conversion size DFT by F that in the equation (2) is shown. α parameter in this equation will affect the audio clarity and error and precisely must be selected.

2) Fi=[Fi(1+α)] i-0,1,2,3,…..

In the next stage , silence frames or the frames without information were added to the main signal and after compression is going to send via the channel that the sent signal is shown in the equation 3.

$$x = \sum_i (\frac{1}{\alpha_i}\mathbf{a}_i)(s_i\alpha_i)$$

The next stage includes the recipient part that the important operation which are done in this part , includes the decoding or decryption of the signals .The decompression is being performed in this part and decrypting will be done on the basis of equation 4.

In this case , in order to control the decrypting error is used of the coding method . the cod-ing method which are being suggested in this paper include the following : the iteration coding and BCH coding, we don't explain them here due to their simplicity and basic nature .

4) $\varphi_c(x) = \frac{1}{c}\varphi(\frac{x}{c}) = \frac{1}{\sqrt{2\pi c}} e^{-\frac{x^2}{2c^2}}$

3- watermarking the data using the corrected patchwork algorithm in the area of time

One of the other methods in watermarking the information is based on the patchwork algorithm. This algorithm is designed based on the higher-order signal statistics. In the issue of blind signal separation is required to use of the higher-order statistics than 2. The main reason is this that with having the second-order statistics of several signals , we cannot offer any comment about their statistics independence.

The first and second torques respectively are the average and medium power of the random variable of the studied signal , note that the central torques i.e, μ=1 and μ=0 are not very important and the second central torque is our regarded signal variance .the third central torque is a proper standard of symmetry related to the density function. And so, the torques or in other words higher boards statistics will put in our hands the proper information. [1]

This algorithm act based on a change in the main signal statistics than the received signal .In the designation of this algorithm, a change in the statistics of the received signal by an instant parameter like ***d*** will be created that this parameter adaptively will be derived from the main signal that this method is called as the corrected patchwork algorithm in the field of time and more is examined.

3.1 encoding and decoding section

In this case also like the previous technique , information must be divided into frames in length of N . Now we suppose that want to cloak a bit within a frame , the method of watermark-ing is as below . First of all , we suppose the two quasi-random strings corresponding to two dif-ferent state of a bit that can be both zero and one and each string include 2n numbers that their values according to the equation 5 have been selected as below.

In this case , the sequence of F represents the samples of main data and two vectors of A,B related to the samples of vector of F that was shown in the equation of 6 . Then ,the average val-ue and standard deviation by the equation of 7 will be achieved and finally , data watermarking will be done base on the equation of 8.

5) 1<K1<K2<N

6) F={F1,F2,F3,.....}

$\tilde{x} = (C - \Sigma)^{-1/2} x = Bs + \tilde{n}$ 7)

8) $$\log L(A, s(1), \ldots, s(T)) = -\sum_{t=1}^{T} [\frac{1}{2} \| As(t) - x(t) \|^2_{\Sigma^{-1}} + \sum_{i=1}^{n} f_i(s_i(t))] + c$$

The next stage is the decrypting stage that in this state, inverse form of the strings which were used in the coding , will be produced and according to it , the statistical samples will be predicted and statistics on the basis of standard deviation will be calculated and decrypting occur via it.

4- The increase of the channel capacity using the pre-emphasis

In the FFSK technique , signal ratio to the channel noise for the higher coefficients compared to the lower coefficients is less. For the removal of this problem and also use of the higher coefficients , we use of the pre-emphasis technique . In this technique , that part of the channel which has the low signal to the noise , can be sent in the sender by more power than other parts and this leads to an optimized usage of the channel and channel capacity too. Results obtained from these techniques are showed in the simulation section.

5- Error reduction using LPF

Assuming that the watermarking bit is zero and because of the bit receiver to be properly retrieved , it is necessary that T0 quite square to be greater than T1 and result of the sampling square difference to be a small amount . For this purpose, we can pass the speech signal via a low-pass filter that its drastic changes to be removed, and then perform the watermarking , and this causes that the error significantly to be reduced and performance and efficiency to be increased.

6-simulation using the Matlab

In the first part (FFSK Method),we convert the speech to the discrete form in order to simu-late the watermarking algorithm using the frequency hopping method , then the signal framing will occur. The sample frequency with 8 kH and ample with 16 Bit will be indicated. In this state, au-dio standard (criterion) will be supposed as α=0.15 and the number of samples equal with 200 . then the simulation will occur.

Fig.1 shows the error graph for the frames with various lengths . Fig.2 indicates the compari-son of two suggested error and Fig.3 shows both the usage effect of the pre-emphasis filter and significant reduction of error and channel capacity increase .

In the second part (patch work method) , the decryption upon the samples will be done in the area of time and the frequency and the results of simulation will prove that this method in the area of time is useful , but for the frequency is not efficient and this is due to the intensive loss of the audio quality in the field of frequency.Fig.4 shows the results obtained from the water-marking and the error existing in the second method and fig.5 shows the review of low-pass filter usage and intensive reduction of error .

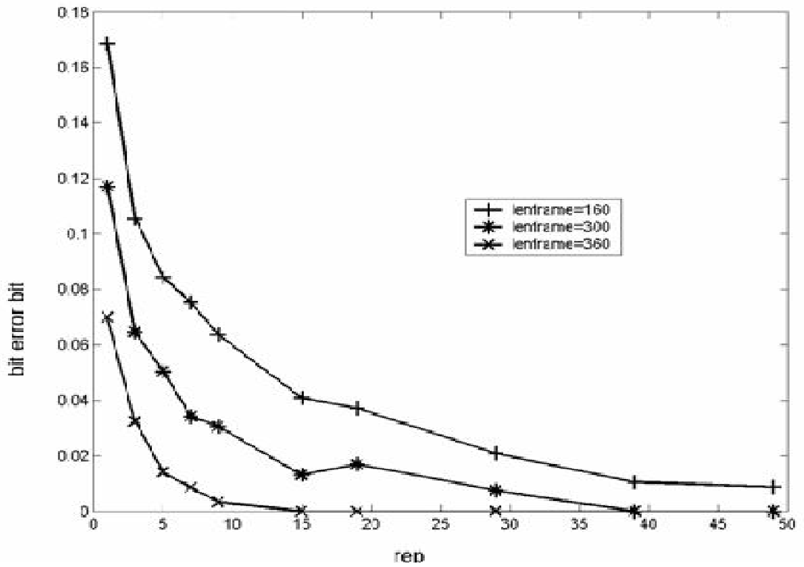

Fig 1 . Error graph for frames with various lengths in the method of FFSK

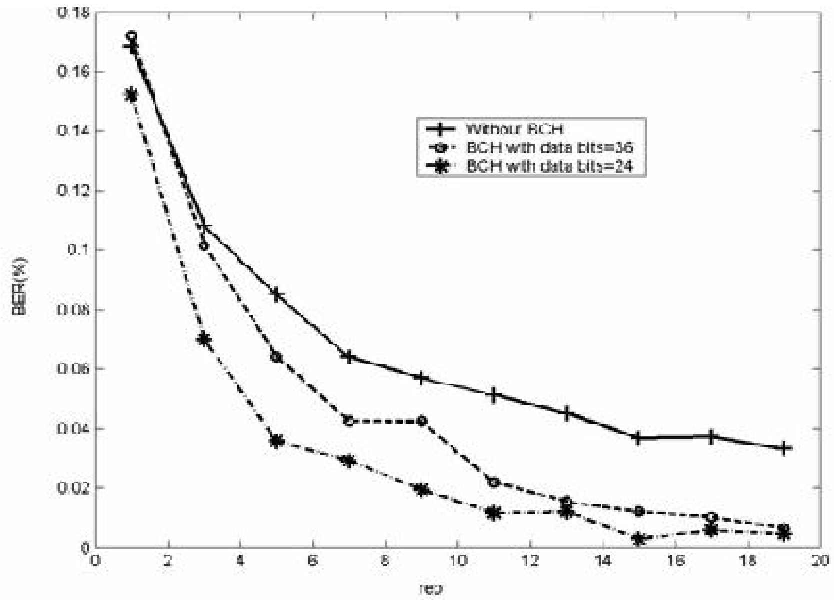

Fig 2 .comparing the two proposed error in the method of FFSK

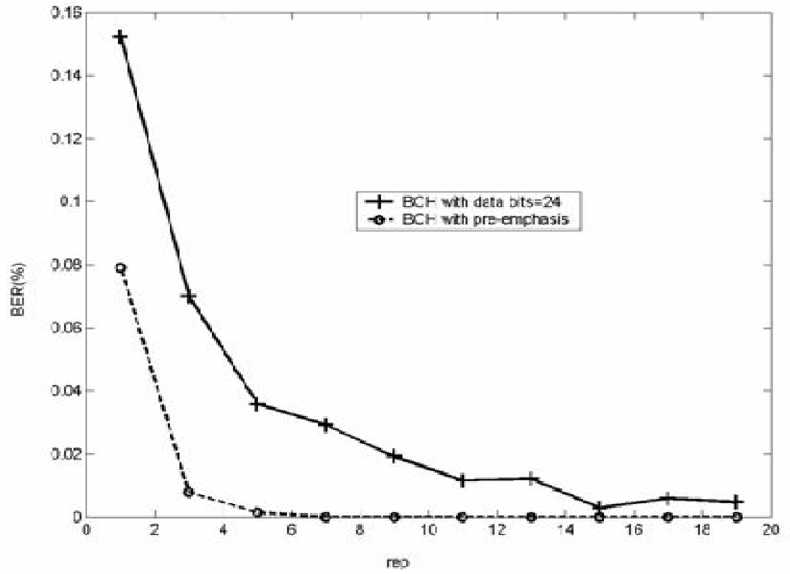

Fig 3.using the pre-emphasis filter and significantly reducing the error in FFSK method.

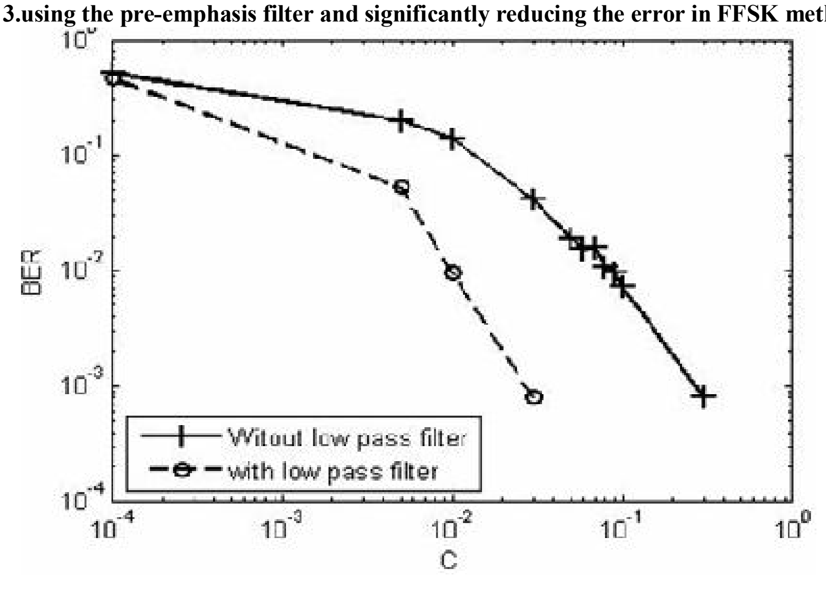

Fig.4 Using the low pass filter, and its effect in the Patch work method.

7- in this paper , two very important method in order to watermarking the data in the acoustic systems was studied. From the most significant features related to the these two method is their resistance against the compression , good audio quality , blind decrypting and their possibility of promptly simulation . In this paper , one method in the area of time and other one in the area of frequency was measured and assessed.

The results of simulation will determine that the method with is used in the area of time , reduces the error ratio to the additional amount and wide spectrum method causes the quality of audio to be increase . Therefore , in the systems which error rate reduction is important , the corrected patchwork method in the area of time will additionally decrease the error rate in comparison with the wide spectrum method.

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