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High Capacity Audio Watermarking Based on Fibonacci Series

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ABSTRACT

Watermarking is used to hide secret information for copyright protection of intellectual property from violation. Digital watermarking is process by which a watermark is embedded or hide into media .This paper contains high capacity audio watermarking system to embed the secret information and extract that secret information exactly by changing magnitude of the some frequency samples with minimal average error. Divide the FFT spectrum into small segments and then change magnitude of the frequency sample based on Fibonacci series is the key idea of this scheme. Maximum change for related FFT sample is less than 61% and the 25% is average error of each sample. Capacity, robustness and transparency are regulated by using two parameters frequency band and frame size. Possible to adaptively changing frequency samples and robust, transparent technique can be achieved by using Fibonacci series. Fibonacci series is the key idea for this watermarking technique to implement robust and transparency technique. This method has minimal perceptual distortion and robustness to signal processing attacks and high payload capacity.

Keywords: Fibonacci Series, Frequency Spectrum, Audio Watermarking, Remarkable Capacity

I. INTRODUCTION

In the present world, various communication techniques development occurs rapidly and digital multimedia content transferring more useful. However distribution and illegal copy of multimedia content becomes easier and many of publishers' and authors' suffers from violation of intellectual property copyrights, which damages financially in many applications. thus nowadays people must focus on copyright management and protection. Embedding or hiding secrete information in multimedia content is known as watermarks, this is considered as the potential solution to copyright infringement [3]. Watermark is embedded and hide in a media (cover data) is known as the digital watermarking. For example digital content such as audio, image, electronic document, video. This embedded information can be later extracted from the watermarked signal forvarious applications. There are several applications of audio watermarking includingcopyright protection, copy protection, content authentication, fingerprinting and broadcast monitoring.

An audio watermarking system may have various properties but it must satisfy the basic requirements of following:

1. Imperceptibility: audio signal quality should not be change after embedding watermarks within the audio .imperceptibility evaluated using both subjective and objective measure
2. Security: Watermarked audio signals should not give any clues of hidden watermarks in them. Also, the security of the watermarking procedure should not depend on secrecy of the watermarking algorithm ,but it must depend on secret keys
3. Robustness: robustness nothing but after various signal processing attacks or malicious attacks, has ability to extract watermarked data from watermarked audio signal.
4. Payload: payload refers to the how amount of data that can be embedded or hide into the host audio per unit of time signal without losing imperceptibility. Payload can measured in bits per second (bps).

Considering the embedding domain, classification of audio watermarking Techniques can be two methods ,one is time-domain and another one is frequency-domain .watermarking In frequency-domain [2]–[15], after applying one of the usual transforms such as the Discrete/Fast Fourier Transform (DFT/FFT) [5]–[7], [11], [12], the Modified Discrete Cosine Transform (MDCT) or the Wavelet Transform (WT) from the signal [8], [10], [13], the hidden bits are embedded into the resulting transform coefficients. In frequency-domain methods, the Fourier transform (FT) is most popular. Among various Fourier transform, the Fast Fourier transform (FFT) is used because of its reduced computational burden and so that these FFT has been the chosen for the proposed scheme. This fast Fourier transform is also used by various authors, such as in [16], which proposes a multi-bit spread-spectrum audio watermarking scheme based on a geometric invariant log coordinate mapping (LCM) feature. In Ref. [5], [7], [11], [12], which were proposed by the authors of this paper, the FFT domain is also selected to embed or hide watermarks to take advantage of the translation-invariant property of the frequency sample (FFT coefficients) to resist small distortions occur in the time domain. In fact, with help of transforms based method we can achieve better perceptional quality and robustness against common attacks in time domain.

In this paper the algorithm suggested is, for embedding secret bits we select a part of the frequency of FFT spectrum. The selected frequency spectrum is divided into small segments each segment consists of certain number of samples, each segment we call as frame and a single secret bit is embedded or hide into each frame as we segmented. The largest Fibonacci number which is lower than each single FFT magnitude in each segment should be calculated and, depending on that the corresponding secret bit to be embedded in each frame , all samples in each frame are changed. If the watermarked bit is "0", all FFT samples in that frame should be changed to the very closest Fibonacci number with even index because watermarked bit is "0". If the watermark bit is "1", all FFT samples in a frame should be changed to nearest Fibonacci number with odd index.

As discussed above, the fast Fourier transform is used to design a scheme in Most of watermarking systems. To the best of our knowledge, this is the first audio watermarking scheme based on Fibonacci series. advantage of Using Fibonacci numbers for embedding the secret bits is that increases transparency and robustness against attacks, where as embedding a secret bit into a single FFT sample of frame is usually very fragile. Almost every watermarking method rely on experimental results to prove the fidelity of watermarking system. However, in this article, the fidelity of suggested system is proved mathematically and this is addition to the experimental results The results of this method show that this method achieves a huge capacity (about 0.54 to 2.5 kbps), and also provides robustness against common signal processing attacks or malicious attacks (even for strong disturbances) and minimal perceptual distortion. The remaining of the paper is organized as. In Section II, Fibonacci series are presented. Section III describes the proposed scheme. Section IV provides discussion about the fidelity. In Section V, the experimental results of this method are shown. Finally, Section VI summarizes the most relevant conclusions of this total research.

II. FIBONACCI NUMBERS AND GOLDEN RATIO

The numbers 1, 1, 2, 3, 5, 8, 13, 21, 34, 55, 89, … , are called as the Fibonacci numbers. This Fibonacci numbers have been named by the nineteenth-century French mathematician Eduardo Lucas after Leonard Fibonacci of Pisa, he was one of the best mathematicians of the Middle Ages, he referred to them in his book Liber Abaci (1202) in connection with his rabbit problem. The Fibonacci sequence has fascinated both amateurs and professional mathematicians for few centuries for their abundant applications and their ubiquitous habit of occurring in totally surprising and unrelated places [17].

In this paper we are using this Fibonacci numbers for the first time for audio watermarking scheme.
The equation to generate Fibonacci sequence is given below.

$$fn = \begin{cases} 0, & if\ n < 0, \\ 1, & if\ n = 1, \\ fn-1+fn-2, & if\ n > 1 \end{cases} \qquad (1)$$

Fibonacci series features are very interesting .one of that feature we using in this article, is the ratio of two consecutive Fibonacci numbers [25]

$$F_n = F_{n-1} + F_{n-2}\ ; \qquad (2)$$

$$\frac{F_n}{F_{n-1}} = \frac{F_{n-1} + F_{n-2}}{F_{n-1}} = 1 + \frac{F_{n-2}}{F_{n-1}}$$

$$= 1 + \frac{1}{\frac{F_{n-1}}{F_{n-2}}}\ ; \qquad (3)$$

$$\lim_{n\to\infty} \frac{F_n}{F_{n-1}} = \lim_{n\to\infty}\left(1 + \frac{1}{\frac{F_{n-1}}{F_{n-2}}}\right)$$

$$= 1 + \frac{1}{\lim_{n\to\infty} \frac{F_{n-1}}{F_{n-2}}} \qquad (4)$$

$$if \lim_{n\to\infty} \frac{F_n}{F_{n-1}} = \varphi;\quad \varphi = 1 + \frac{1}{\varphi}; \qquad (5)$$

$$\varphi^2 + \varphi - 1 = 0; \qquad (6)$$

$$\varphi = \frac{1 \pm \sqrt{5}}{2} \ (7)$$

if φ is positive then, φ=1.618

In fact,φ is the Golden Ratio and is an irrational number Having several curious properties. The Golden Ratio is not a transcendental one (like), since it is the solution of a polynomial equation. The name of Golden Ratio obtained from the Golden Rectangle, a rectangle whose sides are in the proportion of the Golden Ratio. The Golden Rectangle philosophy is an aesthetic one: the ratio is an aesthetically pleasing one and it can be found spontaneously or deliberately turning up in a great deal of art. Therefore, for instance, the front of the Parthenon can be flexible framed with a Golden Rectangle. What a beauty the Golden Rectangle is, how often it really does turn up in art, and whether it does really frame the front of the Parthenon, may be largely a matter of interpretation and preference. Each number of Fibonacci series can be represented by the Golden Ratio [25]. Equation (8) represents how each number of Fibonacci series is generated by the Golden Ratio.

$$F_n = \frac{\varphi n \ \_\ \varphi - n}{\sqrt{5}} \qquad (8)$$

Where is the negative solution of Equation (7).

III. PROPOSED SCHEME

Human auditory system is very sensitive one . For understanding this human auditory system (HAS) characteristics Extensive work has been performed over the years continuously. Then acquired knowledge is applying to audio watermarking and audio compression.

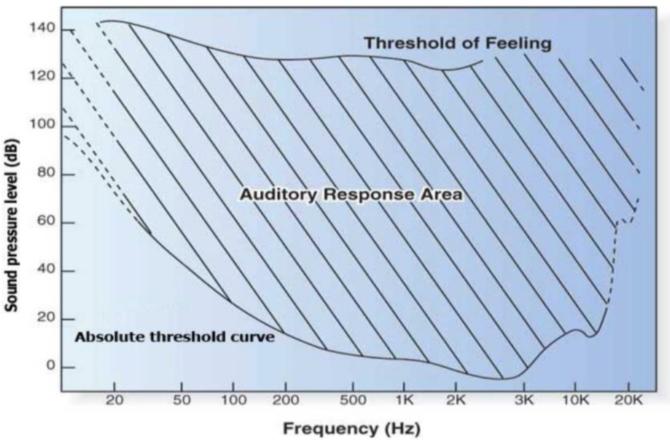

Figure 1. Absolute threshold curve of human auditory system response

Fig. 1 illustrates human auditory system responds for which range of frequencies and which intensities of sound. Human ear minimum sound level detect ability is strongly depend on frequency. At the levelOf pain, sound levels are about six orders of magnitude above the minimal audible threshold. A decibel (dB) is the measuring units for the sound pressure level (SPL). The sound pressure level (SPL) is measured in decibels (dB).

Decibels constitute a logarithmic scale, which is used to measure SPL. Such that each six decibels increment represents a doubling of previous intensity. Intensity refers to the perceived loudness of a sound. Basically human ear hears lowest frequency is 20 Hz and similarly highest frequency approximately 20 kHz. But hearing is best at the range of 3 kHz to 4 kHz and sensitivity of the human ear decrease at lower frequency and higher frequency but more sensitive at higher frequency when compared with the lower frequencies. so it is clearly gives that, what idea we want to use in our proposed system. So in our proposed system we embedded the information in high frequency range. At this high frequency range distortion will be imperceptible to human ear so that we can achieve transparency [11].

In this watermarking scheme, we use the following algorithm to embed or hide watermarks (secret bit stream) into the frequency coefficients. Before that, all the parameters of the scheme should be adjusted. Based on our required payload capacity, transparency and robustness. The frequency band or frequency spectrum and frame size are two parameters that are used to set the properties of the proposed watermarking method. The selected frequency band is segmented into small segments; each of these segments is called as the frame. Then each and every secret bit of the watermark stream is hide into all samples of a frame, this techniques makes our method more robust against some attacks.

A. Tuning

There are two parameters are used to adjust the three properties of watermarking scheme are provided in our suggested system. To adjust the frequency band, robustness and perceptional distortion here we use two parameters one is frequency band and another one is frame size (d).in this scheme we have some general rules this rules help us to meet requirements or to get very close to them. Robustness affected by changing frame size while capacity and distortion effected by frequency band. Better robustness is achieved by increasing the frame size and further increasing the frequency band we can achieve high capacity and more distortion.

Note that frequency band and frame size allow regulating the ODG between0 (not perceptible) and-1 (not annoying), with payload about 540 bits to 2700 bits per second (bps) and allowing robustness.Against MP3-128, which are extremely better than typical requirements of watermarking system.

Most of MP3 cut –off frequencies are greater than 16 kHz, so we set high frequency band $f_h$to 16 kHz or less .To select frequency band based on requirements we adjusting the low frequency bandf_l.12 kHz is the default value for the low frequency bandf_l . Reducing this lower frequency band f_l leads to increasing capacity and distortion .better robustness is achieved increasing frame size d but capacity decrease.
d = 5 is the default value for frame size.

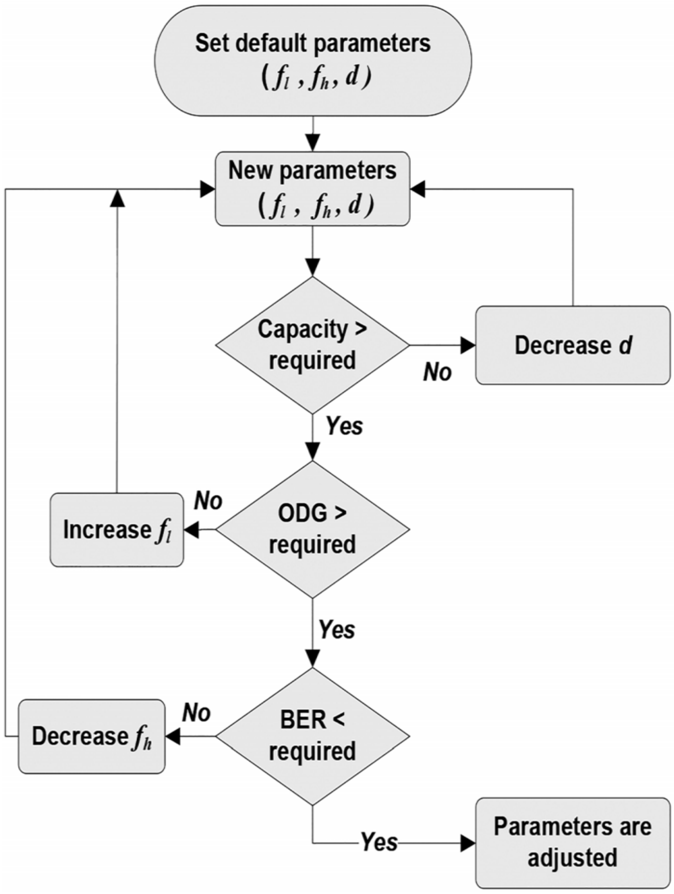

Figure 2. flow chart of tuning proces

selection of tuning parameters flow chart is shown in fig.2 .in the initialization f_h is 16 kHz and $f_l$is 12 kHz and frame size is d = 5. The flow chart gives the adjustment of parameters based on our requirements. However based on some demand adjusting parameters is very difficult and considering trade- off between capacity, robustness and transparency is always necessary.

B.EMBEDDING

Fig. 3 provides the flowchart of the embedding algorithm. Here two parameters frequency band and frame size (d) are used in the embedding process by adjusting this parameters we can achieve our requirement .for simplicity we consider them fix and we don't go with the regulation of this parameters. The effect of this parameters regulation annualized in the experimental part.

Before embedding the watermark bit stream, first the FFT is applied
To the original audio signal and then, the frequency (FFT) samples are modified based
On the Fibonacci series and the watermark bits. Finally the inverse Fourier transforms
(FFT) is applied to gene

1. Apply FFT to calculate the FFT coefficients of the original audio Signal from time domain to frequency domain. We can use the whole file (for short clips, *e.g.* with less than one minute) or blocks of a given length (*e.g.*10 seconds) for lengthy files.
2. Segment the FFT samples in the selected frequency band based on Frame size d.
3. For all the FFT samples or frequency samples in the current frame, find the Largest Fibonacci number$\{fib'_{n,i}\}$, the nth Fibonacci number for ith FFT sample$\{f_i\}$, which is lower than the magnitude of the FFT sample.

It is worth to mention that we use the following Fibonacci series:

$$F = \{1 ,2,\ 3,\ 5,\ 8,\ 13, 21\ \ldots\ldots\ldots..\}$$

Actually In the original Fibonacci series there are two ones, one of Which is removed and remaining used in our algorithm.

4. By using equation (9) we obtain marked FFT samples $\{f_i\}$ as

$$f_i \quad = \begin{cases} fib_{n,i} & if n mod\ 2 = 0\ and w_l = 0, \\ fib_{n+1,i}, & if n mod\ 2 = 1\ and w_l = 0, \\ fib_{n+1,i} & if n mod\ 2 = 0\ \ and w_l = 1\ , \\ fib_{n,i} & if n mod\ 2 = 1\ and w_l = 1, \end{cases} \quad \ldots(9)$$

Where,$l = \left[{}^{i}/_{d}\right] + 1$,$w_l$is the l-th bit of the secret bit stream and $[x]$denotes the largest integer value lower than or equal to$[x]$. Each watermark in secret bit stream is embedded into a suitable frame or each frame represents a single secret bit within it.

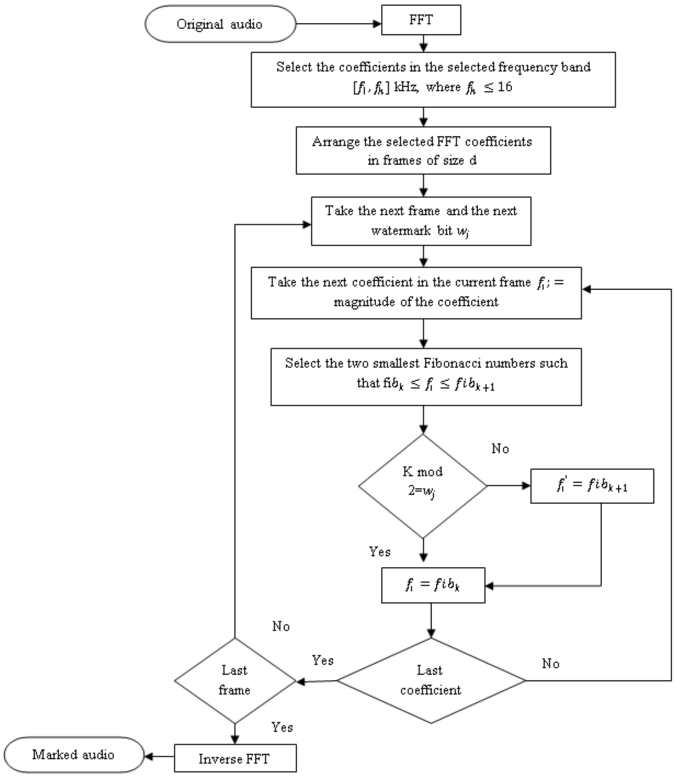

Figure 3.provides the flowchart of the embedding algorithm

5. At last, we use the inverse FFT to obtain the watermarked audio signal.

By increasing the frequency band by changingf_l, the capacity and distortion will be increase and robustness decreases vice versa. Also, by increasing the frame size, we can the strength robustness against signal processing attacks and but reduction in the capacity. In addition, to this the use FFT magnitudes results in better robustness against attacks compared to the use of the real or the imaginary parts only.

C. *Extracting the Secret Bits*

Here the detector is blind; therefore the host audio signal does not required in detection process of watermarks. here detection parameters frequency band and frame size ,which are send to detector in secrete way or some standard parameters are can used for the all audio signals.

The following steps summarize detection process

1. Apply the FFT to calculate the FFT coefficients of the marked audio signal which is in time domain.
2. Segment the FFT samples in the selected frequency band into frames of size d.
3. For each single FFT sample in current frame, find the Nearest Fibonacci number $\left\{fib'_{n,i}\right\}$, the n th Fibonacci number for i th FFT sample, to the magnitude of the FFT sample$\{f_i\}$. If the FFT sample has the same distance from two Fibonacci numbers, we select the lower Fibonacci number. We us $\{1 , 2,\ 3,\ 5,\ 8,\ 13, 21\ \dots\dots\dots..\}$ as the Fibonacci series.
4. To extract a secret bit in each and every a frame, each sample in a frame, should be examined to check if it is a zero (“0” embedded) or a one (“1” embedded). Then, depending on the examine of all samples in the current frame, a secret bit can be extracted. Extraction of watermark bit w_l,achieved by following equation;

$$B_i^{'} = \begin{cases} 0, & if n mod\ 2 = 0, \\ 1, & if n mod\ 2 = 1, \end{cases} \qquad (10)$$

Where$B_i^{'}$ is the watermark bit which is extracted from each sample, After getting information from all samples, based on this information, how many number of samples which represent “0” or “1” (voting scheme) a watermark bit can be extracted for each single frame, then If the number of samples identified as “0” is equal to or larger than half of the frame size, the extracted bit is “0”, else the extracted bit is “1”. For example if five is the frame size and we detect two “0” and three “1”, then the extracted watermark bit of the frame would be “1”.

D. Security

In this system first level security is provided by tuning parameters. Without knowing the frequency band and frame size it is difficult to extract secret data. Therefore it is difficult for attacker if he tries to erase, replace or extract embedded watermark. For suppose if the attacker guess or knows this secret values further security provided with the cryptography. Further increasing security, secret bit stream can be changed into another stream by using pseudo random number generator (PRNG), which makes more difficult for attacker to extract the secret information. Based on the requirement of our watermarking system, a cryptographic method should be chosen.

IV. DISCUSSION

The main idea behind using of Fibonacci numbers is to keeping the modification error for each sample in an acceptable range. Here, we prove that the maximum modification error is about 60% of the correlated FFT sample in the typical case. Assume that we want to convert the original value of a sample,
To the nearest Fibonacci number.

$$F_n \le s \ge F_{n+1}, \qquad (11)$$

Where F_n ,F_{n+1} , are two Fibonacci numbers, based on the distance to each other, $s$can be converted to closest one.

$$e_1 = s - F_n\ , \qquad (12)$$

$$e_2 = F_{n+1} - s, \qquad (13)$$

$$maxerror = max(e_1, e_2) = (F_{n+1} - F_n)$$
$$= F_{n-1} \qquad (14)$$

Error ratio can be calculated, by finding the ratio between two Fibonacci numbers. Assume that ratio between two Fibonacci numbers is.

$$r_n = \frac{F_{n+1}}{F_n}(n = 1,2,.\ .\ .\ .)$$
$$r_1 = 2, r_2 = 1.5, r_3 = 1.6, r_4 = 1.66, r_5 = 1.65,.\ .\ .\ . \ (15)$$

In the above equation, when $n$is very large, then r_n is equal to golden ratio I.e. φ . Even If $n > 3$ it is very close to Thus we can summarize the max error as given below;

$$maxerror = (F_{n+1} - F_n) = (r_n F_n - F_n)$$
$$= (r_n - 1)F_n \quad (16)$$

Theorem 1: According to the result presented in the above Equation (16), the "typical maximum" distortion introduced in the magnitude of a FFT sample using this embedding system is between 0.38 and 0.61.

Proof:

1. Ifs is converted to F_{n+1}

$$maxerrorrate = \frac{\max error}{F_{n+1}} = \frac{(r_n - 1)F_n}{F_{n+1}}$$
$$= \frac{(r_n - 1)F_n}{r_{nF_n}}$$
$$= \frac{r_n - 1}{r_n} \qquad (17)$$

2. If s is converted to F_n

$$maxerrorrate = \frac{mxerror}{F_n} = r_n - 1 \qquad (18)$$

Therefore, if we assume that the "typical" value is ,$r_n = 1.61$, the maximum error rate would be between 0.38 to 0.61. Thus the average of "typical" maximum error rate would be 0.50. This completes the proof.

Note that for small values of n this "typical maximum" distortion may be exceeded. However, for the most of cases, the maximum distortion for original magnitude would be below 50% of that original magnitude. FFT samples will typically have high values but in some case if they are less than 3, then r_n value would be equal to 2 or 1.5. Thus in the mostly worst case, which occurs rarely, the maximum error would be between 0.66 and 1. If all the FFT samples have uniform in distribution, in other words,All values have equal probability then the average error rate is 0.25. It means for each FFT sample average change is 25% only. Hence this fact has a remarkable effect on imperceptibility of watermarked signal.

The distance between 0 s and 1 s automatically adapts to modify the magnitude of the FFT coefficient.

However, this is the one of the method to obtain an "exponential like spacing "of the marked coefficients .assume that we use another sequence instead of Fibonacci numbers as bellow :

$$F_n = [k^n], \quad for\ n = 1, 2, \ .\ . \qquad (19)$$

Where

$$\frac{F_{n+1}}{F_n} \approx k, \qquad for\ n = 1,\ 2,\ .\ .\ .$$

TABLE I. *FIBONACCI* SERIES WITH DIFFERENT K

n	1	2	3	4	5	6	7	8	9	10
$F_n(k = 1.3)$	1	2	3	4	6	8	10	13	17	23
$F_n(k = 1.5)$	1	2	3	5	7	11	17	25	38	57
Fibonacci	**1**	**2**	**3**	**5**	**8**	**13**	**21**	**34**	**55**	**89**
$F_n(k = 1.7)$	1	2	4	8	14	24	41	69	118	201
$F_n(k = 1.9)$	1	3	6	13	24	47	89	169	322	613

Here k has following options:

1-if |k|< 1, this condition is not suitable to proposed scheme, because the generated sequence is formed by numbers lower than one.

2- if |k|>2 is not suitable to suggested method because increasing rate is too high.

3-if 1 < |k| < 2 this results are practical. Table 1 show these sequence when k is 1.3, 1.5, 1.7, and 1.9. If k is 1.5, 1.7 the sequence is very nearest to Fibonacci series which is shown in table 1. When n >3 the Fibonacci numbers is very close to φ (golden ratio) which is proved in discussion part. In the other words when k = φ the Fibonacci numbers are identical to the generated sequence. For the Fibonacci series the spacing between values of marked coefficients is neither neither too small nor too large, this can have convenient effect on trade- off between watermarking properties is concerned. Hence In terms of trade-off between watermarking properties, particularly Fibonacci numbers are good choice.

V. EXPERIMENTAL RESULTS

To show effectiveness of our system, we perform simulation on audio signal which is sampled at 44.1 kHz with 16 bits per sample. Most of All audio clips are sampled at 44.1 kHz with 16 bits per sample and two channels. In these scheme audio signal changing from time domain to frequency domain by using the FFT. then with help of parameters embedding the secret bit stream in the audio signal , while embedding secret data we meet some requirements, which are capacity ,transparency and robustness. The secrete data which is embedded will be not disturb the original audio; by adjusting parameters of this scheme we meet the basic requirements of watermarking.

A. Transparency, Capacity and Robustness

Here we provide imperceptibility results both as SNR and SDG. SNR is provided for comparison of this work with other work.

Here SDG means subjective difference grading which is calculated based on human acoustic perception. To this subjective listening test we select five participants with original audio signal and watermarked signal , and then asked them to repot dissimilarities between original and watermarked signal. Here five-point subjective grading (SDG) is used. SDG=5 means excellent quality, SDG= 4 means good quality, and SDG =1 bad quality. Averaging of outputs of SDG test quality rating is called mean opinion score (MOS). Perceived quality of marked is good when greater than 3.5 (in all cases). Imperceptibility results also provided with the ODG. Values of ODG calculated using advanced ITU-R BS.1378standard which is more accurate measure of distortion.

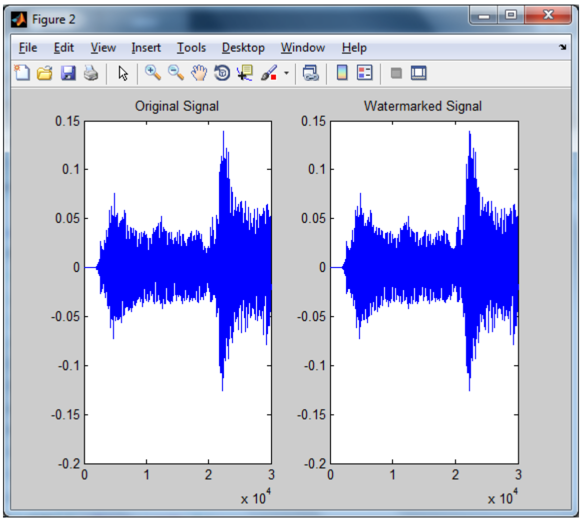

Figure 4. Original and Watermarked Audio

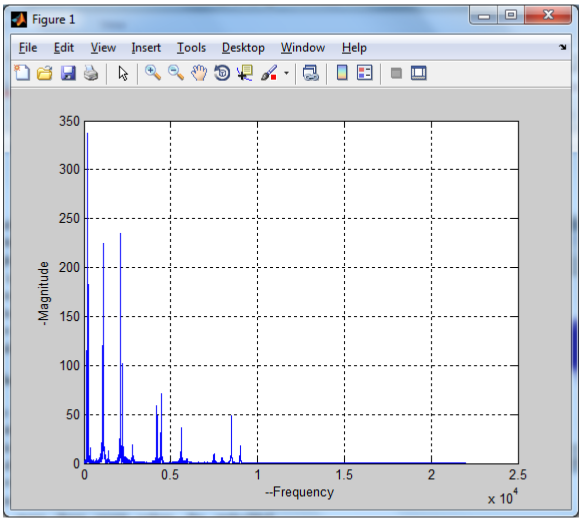

Figure 5. Magnitude plot of original Signal

TABLE II

Reference	SNR	SDG	Robustness to MP3 (Kbs)
proposed	29 to 39	4	128
Emd[1]	25	3.2	32
Lie and chang	28	Not reported	80

TABLE III.
Comparison of different algorithms

S no	Algorithm	Capacity (bps)	Imperceptibility in SNR (dB)
1	[2]	2	42.8 to 44.4
2	[15]	4-512	Not reported
3	[6]	2k-6k	Not reported
4	Emd[16]	46.9 – 50.3	26.5
5	Proposed	540 – 2.5k	31 to 43

A. Comparison

in this article the suggested system was compared with many different recent audio watermarking . Each watermarking system must have different properties; therefore it is difficult to establish fair comparison suggested system with some of other watermarking schemes. Table III provide the comparison of proposed algorithm with a few recent audio watermarking methods. Watermarking methods can be classifies as two groups.

1. Low capacity these methods [1],[2],[3],[4],[9],[14],[25] provided low capacity, payload about few hundred bits, normally robust against some attacks and acceptable transparency.ref [14] contains a very robust scheme against compression and r sampling, but capacity(7-30bits) of this scheme is very low .
2. High capacity these methods [5],[6],[10], presents high capacity solutions to audio watermarking. Payload is few thousand bits, robustness against some attacks and transparency are properties of this system. The proposed system valuable achievement is roust against difficult attacks such as echo, filtering and noises. ref [12] scheme is robust against MP3 -64 and provides 512 bps which is half of the capacity of proposed scheme.

The proposed scheme can be embedded huge information and also less distortion introduced in the marked audio file, in shortly the proposed method achieve huge capacity when compared with it similar imperceptibility and robustness, and when compared with same capacity methods proposed method achieves more imperceptibility and robustness.

In the table III the proposed system is compared with the other techniques this comparison proves the superiority in both imperceptibility and capacity in the literature.

VI. CONCLUSIONS

In this article, a high-capacity and transparent watermarking system for digital audio, that watermarking is robust against common audio signal processing attacks. The suggested method guarantees that for each FFT sample maximum change is less than 61% for typical value of FFT samples and each sample has average error is 25%. in this system two adjustable parameters frequency band and frame size these determine the capacity, the perceptual distortion and the robustness trade-off of the system accurately. Furthermore, the suggested system does not need the original signal because the suggested scheme is blind. The experimental results of this system show that It has a high capacity (540 bps to 2.5kbps) without significant perceptual distortion (ODG about) in marked audio and provides robustness against common signal processing attacks like echo, added noise, filtering or MPEG compression (MP3) even with rates as low as 64 kbps. In addition, the proposed method clearly overcomes the robustness results of recent methods that can be compared with it in terms of capacity.

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