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A Novel Spread Spectrum Algorithm for Audio Watermarking Based on Wavelet Transform

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Abstract

This paper presents a new wavelet-based spread spectrum audio watermarking algorithm. In this algorithm, new techniques are proposed. The first technique is used to specify periods of audio signal to insert a watermark into these periods. Selected periods must satisfy conditions designed to enable robust and inaudible embedding. The technique used for that purpose is called Block Selection Process (BSP). Another technique is proposed for watermark embedding. In this technique, embedding operation is achieved by generating signal-dependent watermark patterns and adding them to the signal in wavelet domain. For each watermark bit two patterns are generated; one to be embedded in approximation part and the other to be embedded in details part. Before adding the patterns into audio signal, patterns are modulated and filtered to reduce a possible distortion might be resulted from these patterns. In the detection stage, we must locate where the watermarks were embedded before start extracting them. BSP is used again to accomplish this task. Then, the watermark patterns are regenerated and correlated with the audio signal to decide whether the watermark exists or not. In the proposed algorithm, the watermark patterns are regenerated from watermarked audio signal and reshaped before correlation. A sort of whitening operation is used to enhance watermark detection. The proposed algorithm has been subjected to and survived robustness tests and the results are reported.

Keywords: Audio watermarking, Information Hiding, Wavelet Transform, Synchronization.

1. Introduction

With the rapid growth of the Internet, unauthorized copy and distribution of digital media become easier. Digital watermarking introduced as one of the growing area of research to solve the problem of copyright protection. Digital watermarking is the process of embedding imperceptible information (watermark) into digital multimedia work (such as images, audio and video) [1]. The embedded information should be recoverable in the detection stage to provide a proof for the ownership of copyright. In general, watermarking algorithms have to satisfy two conflicting conditions. First, the embedded watermark must introduce as less as possible distortion to the multimedia work. Second, embedded watermark must be robust against attacks and common digital signal processing operations. A number of digital audio watermarking algorithms have proposed that exploit different signal domains and different techniques to produce inaudible robust watermark. These algorithms can be classified according to the domain where the watermark embedded [2]. There are four domains which are used for this purpose; these are: time domain, frequency domain, compressed domain, and wavelet domain. In time domain, the watermark is embedded directly into digital audio signal, no domain transform is required [3]. In frequency domain, the watermarking requires to transform the audio signal to frequency domain where the watermark embedded [4]. Compress domain technique hides the watermark in compressed audio to avoid going through costly decompression and compression processes to embed the watermark [5]. Wavelet Transform is also used in audio watermarking to add the watermark in Wavelet domain [6].

In [7], another classification is presented which is based on a technique used in watermark embedding. In this classification, audio watermarking techniques are classified into five categories. The first technique is quantization based watermarking (Quantization Index Modulation (QIM)) [8] which modulate audio signal by quantizing audio samples to compute new values to these samples using quantization function. The second one is the spread spectrum method that is based on correlation in watermark detection [4]. The third one is the two-set method based on the difference between two, or more, sets of samples (i.e. patchwork watermarking technique [9], [10], [11]). The fourth one is the replica method using the close copy of the original audio and finally the self-marking method. Each technique of the above techniques can be applied in different domains of the audio signal.

The present work proposes a new algorithm to watermarking audio signal in wavelet domain using spread spectrum technique. The Discrete Wavelet Transform (DWT) is used to transform audio signal from time domain to wavelet domain. DWT is a technique for extracting information about non-stationary signals such as audio which was developed as an alternative to the Short Time Fourier Transform (STFT) to overcome problems related to its time and frequency resolution properties. In Fourier Transform, the signal is expressed as infinite sum of a series of sines and cosines referred to as Fourier expansion. The problem with Fourier Expansion is that it has no time resolution [12] which means that although we might be able to determine all the frequencies present in a signal, we do not know when they are present. DWT provides time-frequency resolution; therefore, we can reconstruct the signal perfectly. In Wavelet Transform, signal is decomposed to two parts, approximation which contains the low frequency components of the signal and details which contains the high frequency components of the signal. The approximation part can be decomposed again to low and high frequency parts. The data obtained from these decompositions are called DWT coefficients. The original signal can be reconstructed by applying inverse DWT (IDWT) on those coefficients. More information about wavelet transforms can be found in [12], [13].

Kim et al [6] proposed an audio watermarking technique using wavelet transform. Their technique uses patchwork algorithm to embed a watermark in wavelet domain by changing some statistical value related to the DWT coefficients. This technique modifies audio signal which is watermarked regardless of its characteristics. In other words, the watermark signal is audio-independent and depends only on the watermarking key (blind embedder). Consequently, fidelity of watermarked audio might be seriously affected. Cvejie et al [14] presented another technique for audio watermarking in wavelet domain; their technique is based on using frequency hopping and patchwork method in watermark embedding. Embedded watermark is detected using a statistical method which is similar to the method presented in [11]. Another algorithm for audio watermarking in wavelet domain is proposed by Li et al [15], in which they embed an image as a watermark inside the audio signal. Signal to Noise Ratio (SNR) is used to adjust the intensity of the watermark in order to balance between fidelity and robustness. The detection algorithm recovers the embedded image with as less as possible distortions, which means that we may have misdetection; however the recovered image is still recognizable. But it requires the original audio signal to be presented during detection. In [16], the authors combine the idea of embedding a binary sequence and the idea of embedding a still image. They claim that their system provides objective and subjective detection. However, desynchronization attack represents a serious threat to audio watermarking algorithms. Some embedding techniques use statistics of original audio signal as a secret key to locate the embedded watermark [17], which mean desynchronization attack will not defeat recovering of watermark signal. Such techniques eliminate the need to embed synchronization code.In this paper, the focus is on developing two new techniques to enhance the performance of digital audio watermarking in wavelet domain using spread-spectrum method. The first technique is used to generate signal-dependent watermark patterns by which watermark patterns are generated and shaped to produce inaudible watermark with high level of robustness to a wide range of watermarking removal attacks. The WaterMark Patterns Generation (WMPG) technique is used in embedding process and in detection process as well, and does not require psychoacoustic model to render the watermark inaudible. The second technique is Block Selection Process (BSP) which is a novel technique used to analyze and select periods of audio signal being watermarked according to the suitability of these periods for hosting inaudible robust watermark. The BSP is used in embedding process, to select watermarking-suitable block of audio signal, and in detection process to locate blocks where the watermark embedded. In detection process, another technique is used to whiten the watermarked audio signal in order to reduce the effect of audio signal on the watermark buried inside it. We have made use of these proposed techniques in developing wavelet-based watermark embedding and detection algorithms where these algorithms have been implemented and tested to show a high level of robustness as will be explained in following sections.The reset of this paper is organized as follows: next section is to describe the proposed embedding algorithm which uses BSP and WMPG techniques. Section 3 shows the proposed detection algorithm and the techniques used to enhance it, in addition to performance of the watermark detection. In section 4 results of robustness tests are shown, and finally conclusions are presented in the last section.

2. Proposed Embedding Algorithm

The proposed embedding algorithm begins with selecting watermarking-adequate blocks from audio signal being watermarked by using BSP, then generates a watermark pattern for each selected block. Fig. 1 shows the stages of watermark embedding while the subsections bellow give the details of the proposed techniques.

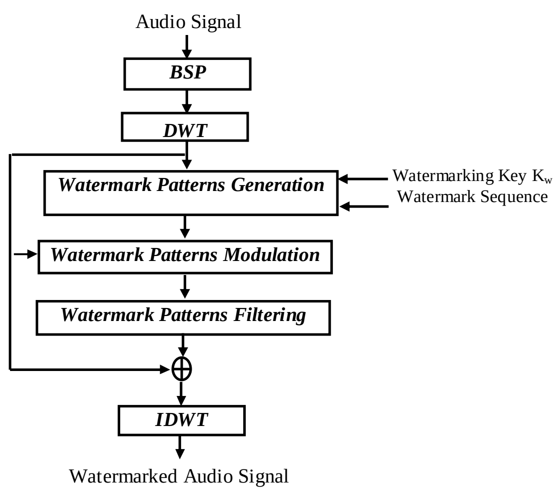


Fig. 1: Block diagram of watermark embedding algorithm.

2.1. Block Selection Process (BSP)

BSP is a process of selecting audio blocks which are located within most significant audio periods. To do so, BSP breaks up the audio signal to large segments of time-domain audio samples, each segment, in turn, will be subdivided to N_s small fragments called chunks. The chunks will be combined together to build audio blocks, every $N_B < N_s$ contiguous chunks will form one audio block. This means that one audio segment has $N_s - N_B + 1$ overlapped blocks as illustrated in Fig. 2 (in this figure each audio block is supposed to be consisted of 5 chunks).

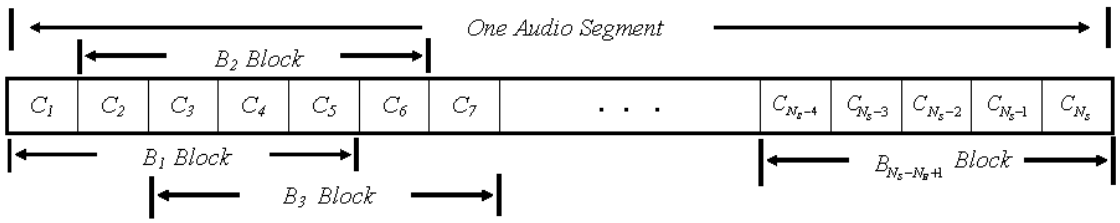


Fig. 2: Audio Signal Segmentation

The next stage of BSP starts processing segment's blocks and applying selection criteria to determine which blocks to be watermarked. To explain this operation, suppose we have a segment of audio signal consists of N_s chunks $(C_1, C_2, \ldots, C_{Ns})$, then the set of all possible blocks (that could be overlapped) contained in this segment are defined as follows:

$$B = \{B_1, B_2 \ldots B_{N_S-N_B+1}\} \ldots (1)$$

where block B_1 starts at chunk C_1 and consists of the chunks $C_1, \ldots, C_{N_B}$, $B2$ starts at chunk C_2 and consists of the chunks $C_2, \ldots, C_{N_B+1}$, and so on.

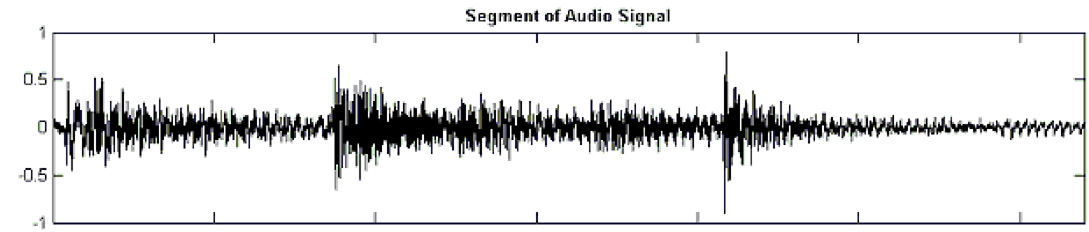
Let $Std(.)$ be a function returns the standard deviation of its argument, and let S_B be a sequence of standard deviations of the blocks in B given by

$$S_B = \{s_i \mid s_i = Std(B_i) \text{ and } B_i \in B, \text{ for } i = 1,2,\ldots,N_S - N_B + 1\} \quad (2)$$

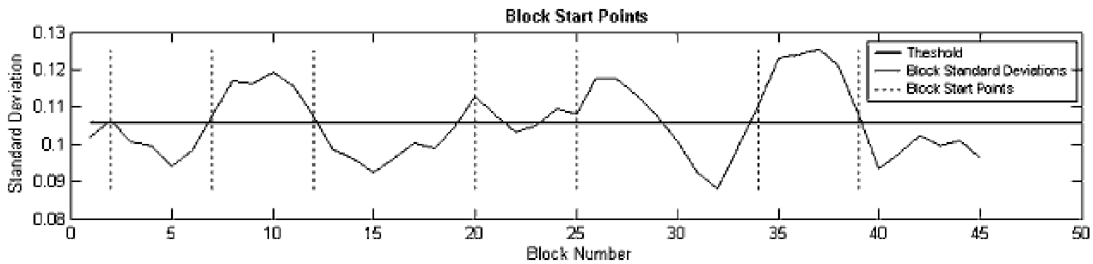
then the set of selected blocks B' is defined as:

$$B' = \{B_i \in B \mid Std(B_i) > \overline{S}_B \text{, for each two successive blocks } B_j \text{ and } B_k \text{, } j\text{-}k \geq N_B \text{, } j>k \text{ and } j \leq N_S - N_B + 1\} \ldots \quad (3)$$

$\overline{S}_B$ (the mean of standard deviations of all blocks within one audio segment) is used as a threshold in selecting the blocks with high standard deviations. The value of $\overline{S}_B$ will vary according to the audio segment being processed by BSP. Since that the standard deviation of audio block directly impacts the robustness of the embedded watermark [4], and then BSP picks the watermarking-suitable audio blocks out of all blocks contained in one segment. This means that BSP avoids the blocks of silence or weak audio signals (low loudness) to be selected for watermarking while watermarking such blocks may result in audible distortions and/or unreliable detection. Furthermore, watermark patterns generation technique (described in next subsection) depends on information extracted from audio signal in computing watermark patterns with energy proportional to that of audio signal. The purpose of this is to balance between robustness and fidelity requirements. Thus, introducing weak audio signal to watermarking leads to generating weak watermark patterns that might cause misdetection. On the other hand, introducing high energy audio signal results in computing high energy watermark patterns. Therefore, it seems to be a wise decision to select audio blocks with high energy to ensure reliable detection and high fidelity. Fig. 3 explains applying BSP on one segment of audio signal. It is noticeable that the start points of the selected blocks shown on the curve of blocks standard deviations S_B. The Fig. 3.a shows the audio signal while Fig. 3.b shows the standard deviation of all blocks in that signal. All the blocks that have a standard deviation greater than $\overline{S}_B$ (the threshold) will be selected for watermarking. The vertical doted lines mark the start points of each of these blocks.



A. Segment of audio signal under processing of BSP



B. Start point of the selected blocks marked by doted vertical line

Fig. 3: Block Selection Process, doted lines show selected blocks.

2.2. Initial Generation of Watermark Patterns

WMPG technique consists of two stages: initial generation of watermark patterns and watermark patterns shaping. The first stage will be discussed in this subsection while the second one will be discussed in next subsection. After applying BSP and selecting the blocks that will be introduced to embedding process, each block will be processed separately to embed one bit of the watermark. In order to embed one watermark bit, two signal-dependent watermark patterns are generated. These two patterns are parameterised by watermarking key K_w and information extracted from audio block. The watermark patterns generation algorithm can be described as follows:

1. Decompose the audio signal until the level k. So, we have A_k, D_0, $D_1 \ldots D_k$ resulting from decomposition. The watermark will be embedded into approximation A_k and details D_k parts.
2. Using the watermarking Key K_w, two sets of indexes $I = \{i_1, i_2, \ldots, i_{n1}\}$ and $J = \{j_1, j_2, \ldots, j_{n2}\}$ are generated.
3. Compute standard deviations S_1 and S_2 for A_k and D_k, respectively.
4. Generate two watermark patterns, WA and WD, such that:

If watermark bit being embedded = 1 then

$WA(k_1) = S_1$
$WA(k_2) = -S_1$
$WD(k_3) = S_2$
$WD(k_4) = -S_2$

Otherwise

$WA(k_1) = -S_1$
$WA(k_2) = S_1$
$WD(k_3) = -S_2$
$WD(k_4) = S_2$

End if

where k_1 and $k_3 \in I$ and k_2 and $k_4 \in J$.

The energy of resulting watermark patterns are proportional to that of audio block being watermarked. Thus, watermark patterns generation technique will produce a watermark that tries to balance between robustness and fidelity. Further operations are required to prepare the watermark patterns for embedding, as it is shown in next section.

2.3. Watermark Patterns Shaping and Embedding

In this stage of WMPG, two shaping operations are applied on watermark patterns WA and WD. First we scale the patterns then a simple filtering operation is applied.

Experimentally, applying these two shaping operations give very good results in maintaining fidelity of watermarked audio signal. In scaling operation the watermark patterns *WA* and *WD* are modulated by A_k and D_k, respectively,

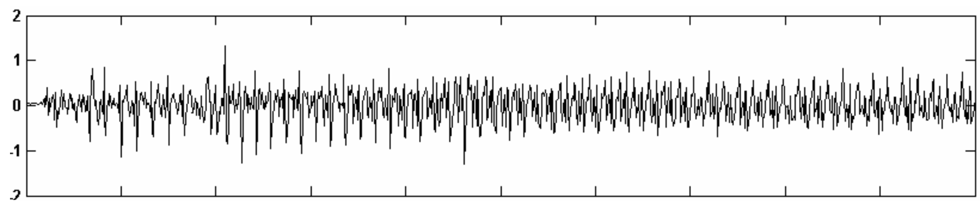
$$WA' = WA \,.\, A_k$$
$$WD' = WD \,.\, D_k \qquad \text{.... (4)}$$

The modulated patterns WA' and WD' will go through filtering operation using a moving average filter of the third order. Suppose w_i is a value in WA' or WD' , then moving average filter will be applied as follows:

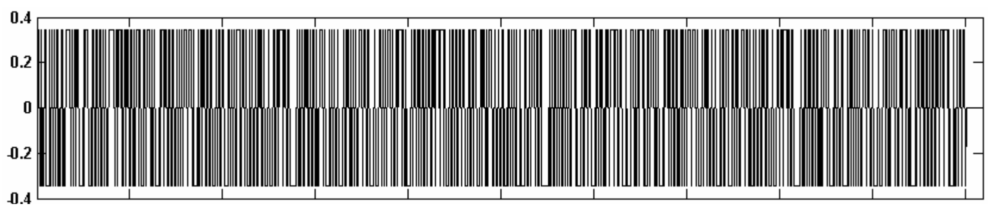
$$w_i' = \frac{1}{3}\sum_{j=-1}^{1} w_{i+j} \qquad \text{... (5)}$$

where w_i' is a value in the filtered watermark patterns WA'' and WD''.

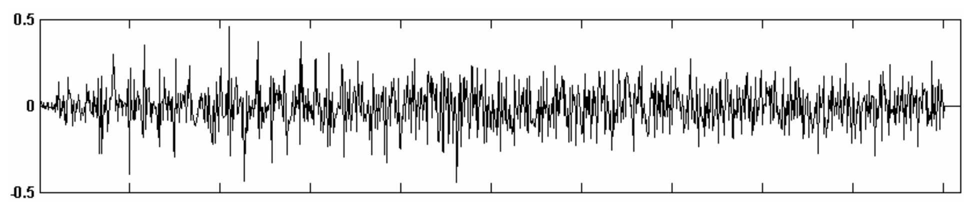
Every value in A_k or D_k has a certain capacity, which specifies the amount of watermark value that can be added without causing an audible distortion. Scaling operation has been used to achieve this goal by adjusting each value in the watermark patterns to be suitable for it's corresponding value in the audio signal. Moving average filter is used to smooth the watermark signal, in order to eliminate possibly existing audible noise. Furthermore, filtering has another job to do in regenerating watermark patterns during detection stage, this job will be explained later in section 3. Fig. 4 shows the original audio signal and watermark pattern over stages of generation.



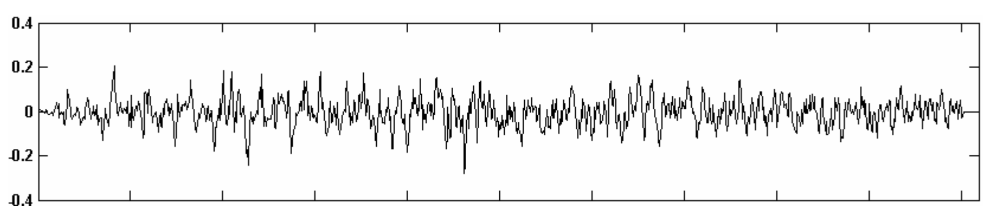
a. Original Audio Signal (A_k)



b. Original Watermark Pattern (*WA*)



c. The Watermark Pattern after Scaling (WA')



d. The Watermark Pattern after Filtering (WA'')

Fig. 4: Watermark patterns generating and shaping

After preparing the watermark patterns, we can embed them in the decomposed audio signal (wavelet domain). The shaped watermark patterns WA'' and WD'' will be added to A_k and D_k , respectively,

$$A'_k = A_k + WA'' . \alpha_1$$

$$D'_k = D_k + WD''.\alpha_2 \qquad \dots (6)$$

where α_1 and α_2 are two parameters used to control the trade-off between robustness and fidelity. Our experiments show that the best values for α_1 and α_2 are 1.2 and 0.7, respectively, and the reason is because of that the details part of the signal is more sensitive for modification that might result in presenting a high frequency noise, while that the approximation part is able to host more stronger watermark signal.

Finnaly, to get the watermarked audio signal we apply IDWT on the wavelet coefficients A'_k, D_1, D_2,… D'_k.

3. Proposed Detection Algorithm

To extract the embedded watermark, detection algorithm has to locate the watermarked blocks using BSP technique, then after, regenerate the watermark patterns and finally correlate them with watermarked audio blocks. Fig. 5 shows the stages of detection algorithm, while the following subsections describe the proposed techniques.

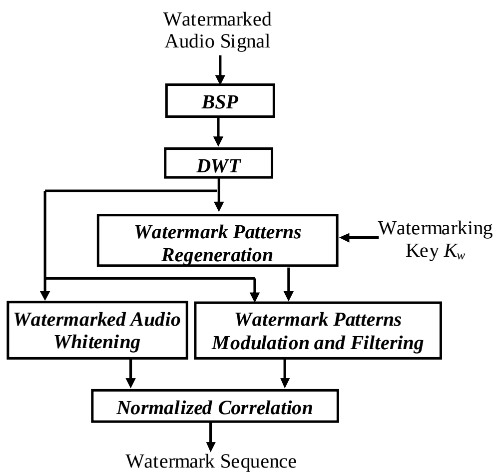


Fig. 5: Watermark detection algorithm block diagram.

3.1. Blocks Selection and Watermark Patterns Regeneration

The BSP technique is used again to locate the watermarked blocks, and then detection process starts regenerating watermark patterns from the watermarked audio signal. These patterns are correlated with watermarked signal using normalized correlation to extract the embedded watermark.

It is noteworthy that standard deviations of watermarked audio blocks are likely not equal to that of same blocks before watermark embedding. Consequently, standard deviations sequence of watermarked audio, S_{BW} (where BW is the set of all blocks in watermarked audio signal), will not be similar to standard deviations sequence of the original signal S_B. This change may affect both Blocks Selection Process and watermark patterns regeneration. However, BSP and patterns regeneration can avoid the damage caused by modifying standard deviations of original signal blocks. It is noticeable from equation 3 that BSP selects audio blocks according to a dynamic threshold $\overline{S}_B$, therefore, the rise in standard deviations will lead to increase $\overline{S}_{BW}$ (the mean of standard deviations of all blocks within one watermarked audio segment). Thus, the new threshold will be proportional to the new standard deviations and will success in selecting the same blocks involved in watermarking stage. As for pattern regeneration, scaling and filtering (smoothing) will help to produce similar watermark patterns lead to have a correct detection. Four watermark patterns are regenerated in detection process, two for the watermark bit 1 and the other two for the watermark bit 0. To regenerate these four patterns, audio block is decomposed using DWT to the level k, and then we compute the standard deviations S1 and S2 for Ak and Dk, respectively. The four patterns W1, W2, W3 and W4 are computed as follows:

$$W_1(k_1) = S_1 \;\; \& \;\; W_1(k_2) = -S_1$$
$$W_2(k_3) = S_2 \;\; \& \;\; W_2(k_4) = -S_2$$
$$W_3(k_1) = -S_1 \;\; \& \;\; W_3(k_2) = S_1$$
$$W_4(k_3) = -S_2 \;\; \& \;\; W_4(k_4) = S_2 \qquad \ldots (7)$$

where k_1 and $k_3 \in I$ and k_2 and $k_4 \in J$. I and J were generated using the watermarking key K_w.

W_1 and W_2 are two patterns for watermark bit 1, while W_3 and W_4 are for watermark bit 0. These patterns will go through modulation using audio signal and then filtering by moving average filter, using the same way of embedding process (i.e. equations (4) and (5)).

3.2. Correlation for Watermark Detection

As it is known, watermark signal must be inaudible. This means that the energy of audio signal is greater than that of watermark signal. As a result of this fact, extracting the watermark will never be an easy task.

In order to minimize the affect of audio signal on watermark patterns embedded inside it and enhance the detection process, a sort of whitening operation is applied on A_k and D_k, as follows:

$$A_w = A_k - f_n(A_k)$$
$$D_w = D_k - f_n(D_k) \qquad \ldots (8)$$

where $f_n(\,.\,)$ is a simple moving average filter similar to that in equation (5) but in this case the filter order $n > 30$. This whitening operation can reduce the effect of original audio signal on the embedded watermark, consequently yields in more reliable detection. After this preparation step, normalized correlation is used to measure the similarity between watermarked audio and regenerated watermark patterns. Let [1]

$$corr(W,X) = \frac{\sum_i w_i . x_i}{\sqrt{\sum_i x_i^2}} \qquad \ldots (9)$$

represents a correlation function.

Then we compute:

$$C_1 = corr(W_1, A_w)$$
$$C_2 = corr(W_2, D_w)$$
$$C_3 = corr(W_3, A_w)$$
$$C_4 = corr(W_4, D_w) \qquad \ldots (10)$$

C_1 and C_2 are to measure the similarity between the watermark patterns of bit 1 and the decomposed audio signal, and C_3 and C_4 are to measure the similarity between the watermark patterns of bit 0. Each two of these values will be involved in computing two detection values CW_1 and CW_0:

$$CW_1 = C_1 \,.\, 0.7 + C_2 \,.\, 0.3$$
$$CW_0 = C_3 \,.\, 0.7 + C_4 \,.\, 0.3 \qquad \ldots (11)$$

CW_1 represents the detection value of the watermark patterns of 1, while CW_0 represents that of 0.

In our algorithm, approximation part of audio signal has an important consideration than details part. The reason is that, the most significant components of audio signal lies in approximation which means that approximation part is more robust than details part, thus, correlation value of approximation has a higher weight in computing the detection values.

CW_1 and CW_0 should keep a reasonable distance between them to make a reliable decision about watermark existence. Let T be a threshold for watermark detection, then final decision about watermark is made such that:

If $|CW_1 - CW_0| < T$ *then*
No watermark is embedded.
Else if $CW_1 > CW_0$ *then*
1 is embedded.
Else
0 is embedded
End if

A further criterion is applied to increase watermark detection reliability. As explained above, four correlation values are computed during detection process, two correlation values for watermark 1 and the other two for watermark 0. If the embedded watermark was 1, then both C_1 and C_2 must be greater than C_3 and C_4, respectively, regardless that C_2 and C_4 may not have a significant difference. The reverse must be held in case of that 0 was embedded. If that logic cannot be held, this leads us to conclude that no watermark is embedded and that was just a false positive detection. For example, suppose that we have C_1 greater than C_3, but C_2 was less than C_4. It is clear that C_1 and C_3 values indicate that the pattern embedded in approximation is for 1, but C_2 and C_3 values indicate that the embedded pattern is for 0. While the two comparisons indicate existence of different watermark patterns, this means that no watermark is presented. This criterion will be called a consistency condition and it is summarized in Table 1.

Table1: Consistency condition

Comparison		Result of Detection
C_1 & C_3	C_2 & C_4	
$C_1 > C_3$	$C_2 > C_4$	1 is embedded
$C_1 > C_3$	$C_2 < C_4$	No watermark embedded
$C_1 < C_3$	$C_2 < C_4$	0 is embedded
$C_1 < C_3$	$C_2 > C_4$	No watermark embedded

To measure the level of performance of the proposed algorithm, distribution of detection values is computed. The probability distributions of watermarked and unwatermarked audio blocks are shown in Fig. 6.The solid curve shows the probability distributions of watermarked audio blocks while the doted one represents the probability distribution of the unwatermarked blocks. As shown, the two curves are completely isolated, which means that the probability of false positive is zero. In next section the affect of attacks on these distributions is shown.

4. Robustness Tests

The proposed watermarking algorithm has been subjected to a number of attacks in order to test its robustness. In this section shows results obtained from applying attacks in Audio StirMark [18], which is available on the Web. We used the same default parameters provided by the software. The effects of the attacks on the probability distributions of detection values of watermarked and unwatermarked audio are described below:

1- Low pass filter attack: The cut-off frequency of the low pass filter was 9000 Hz. The results of the attack are depicted in Fig. 7. A small number of detection values are degraded, even though watermark detector still able to discriminate between watermarked and unwatermarked audio blocks.

2- High pass filter attack: The cut-off frequency of high pass filter was 200 Hz. This attack has more affects than low pass filter. That is because high pass filter impacts approximation part of audio signal. Fig. 8 shows the affect of the attack.

3- Adding dynamic noise attack: Dynamic white noise is added to the watermarked signal. As shown in Fig. 9, attack has no considerable impact on watermark detection.

4- Compressor attack: In this attack the loudness of certain samples in watermarked audio signal will be changed. A threshold is used to determine these samples. Fig. 10 depicts the effect of this attack. As we can see, this attack changed the shape of detection values distribution. However, false positive probability is still zero.

5- ZeroCrossing attack: This attack sets all the samples values to zero if they are less than a given threshold. The result of this attack is shown in Fig. 11.

6- Exchange samples attack: For all samples in audio signal, every two successive values will be swapped. Fig. 12 shows the result of this attack.

7- Amplifying attack: In this attack, the loudness of all audio signal samples will be changed (i.e. increased or decreased). Fig. 13 shows the impact of the attack on detection.

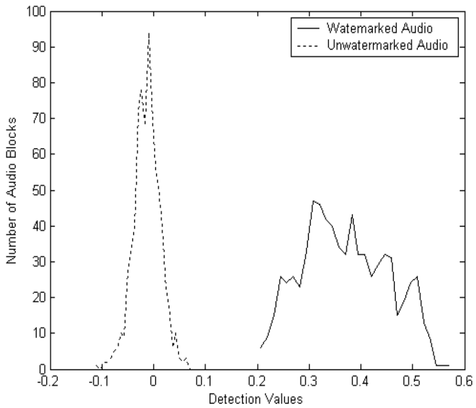


Fig. 6: probability distribution of detection values for watermarked and unwatermarked audio samples.

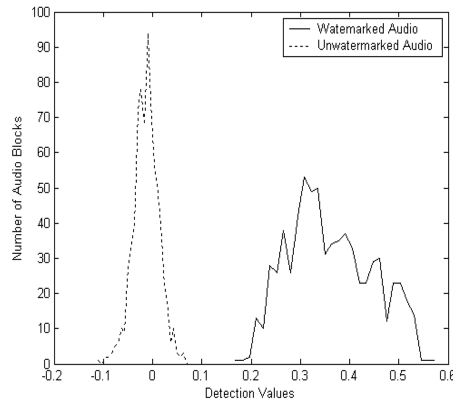


Fig. 7: probability distribution of detection values for watermarked and unwatermarked audio samples after applying low pass filter attack.

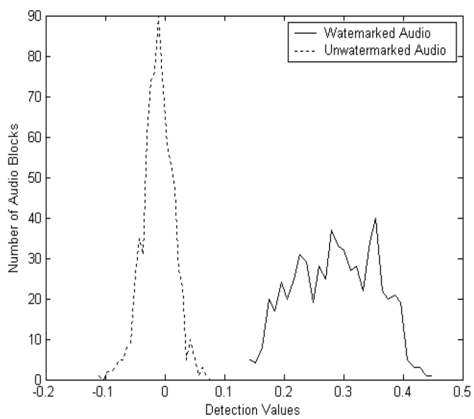


Fig. 8: probability distribution of detection values for watermarked and unwatermarked audio samples after applying high pass filter attack.

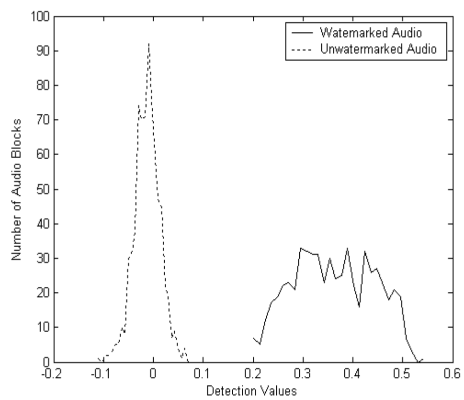


Fig. 9: probability distribution of detection values for watermarked and unwatermarked audio samples after adding dynamic random noise.

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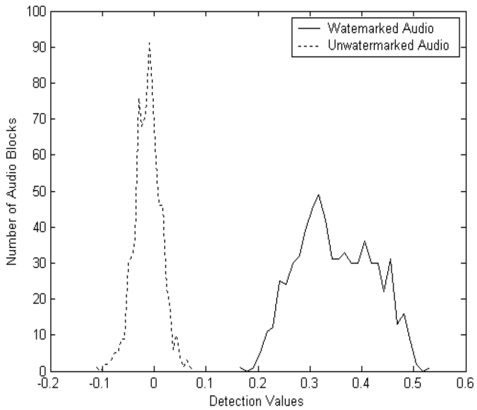

Fig. 10: probability distribution of detection values for watermarked and unwatermarked audio samples after applying compressor attack.

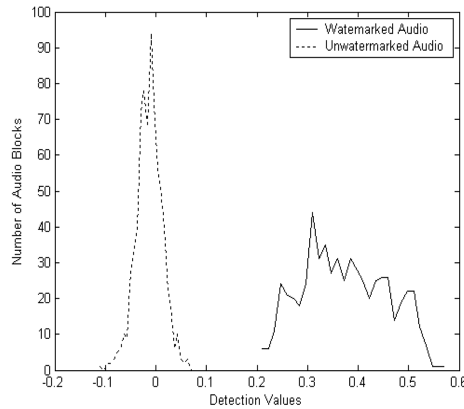

Fig. 11: probability distribution of detection values for watermarked and unwatermarked audio samples after applying zerocrossing attack.

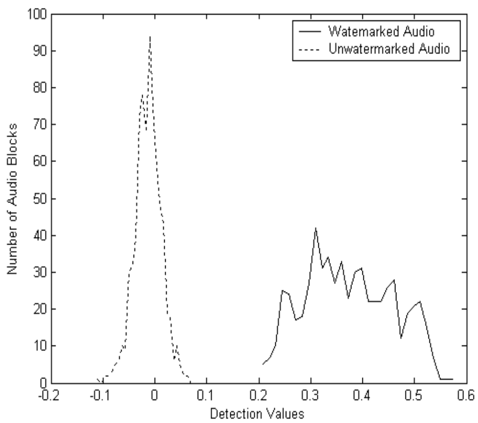

Fig. 12: probability distribution of detection values for watermarked and unwatermarked audio samples after applying exchange samples attack

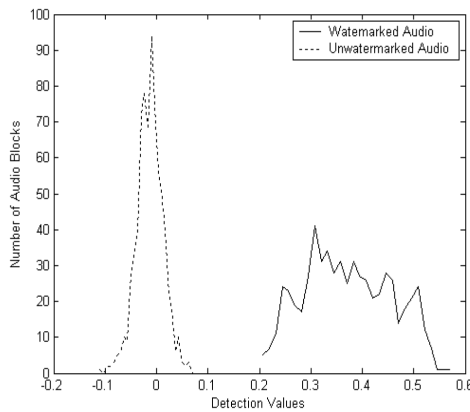

Fig. 13: probability distribution of detection values for watermarked and unwatermarked audio samples after applying amplifying attack.

5. Conclusion

This paper presents a new audio watermarking algorithm, which uses spread spectrum technology in embedding a watermark in wavelet domain. Two techniques are proposed to be incorporated in this algorithm, WMPG and BSP. The first technique uses very simple operations to generate robust and inaudible watermark patterns and, unlike other watermarking algorithms, does not require using the psychoacoustic masking model to render the watermark inaudible (e.g. [4]).The BSP technique shows very good results in improving the performance of both embedding and detection algorithms (high level of fidelity of watermarked audio and more reliable watermark detection). From another side, the payload of the proposed algorithm is decreased. That is because of ignoring the periods of silence and weak audio signals during watermark embedding.The proposed algorithm exploits different techniques to overcome the difficulties of audio watermarking. Another wavelet-based watermarking algorithm by Kim et al embeds watermark signal by modifying audio signal regardless the distortions such process may result in.

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خوارزمية حديثة لتضمين العلامات المائية في الإشارات الصوتية باستخدام تقنية نشر الطيف ومعتمدة على التحويل المويجي

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<u>الخلاصة</u>

يقترح هذا البحث خوارزمية لإضافة العلامات المائية للإشارات الصوتية بالاعتماد على التحويل المويجي وتقنية نشر الطيف. تستخدم هذه الخوارزمية تقنية جديدة لتحديد فترات أو مقاطع من الإشارة الصوتية لتضمين إشارة العلامة المائية حيث يجب إن تستوفي هذه الإشارة الصوتية التي تم تحديدها على مميزات محددة تدعم متانة العلامة المائية وإخفاء أثارها المسموعة. إما عملية تضمين العلامة المائية فهي تستخدم تقنية جديدة مقترحة تولد العلامة المائية بالاعتماد على مواصفات الإشارة الصوتية المــراد إضافة العلامة المائية لها. وتتم إضافة العلامات المائية ضمن المجال المويجي. تتضمن المعالجات المستخدمة لتوليــد العلامــة المائية تقيسا و ترشيحا للإشارة لإزالة الضوضاء الذي يمكن إن تسببه عملية التضمين. وفي مرحلة استخراج العلامات المائيــة يجب علينا أولا تحديد مواقع العلامات المائية باستخدام نفس التقنية المستخدمة في مرحلة التضمين.بعدها يعاد توليد العلامــات المائية من الإشارة الصوتية المضمنة دون الحاجة إلى توفر إشارة الصوت الأصلية. ويتم تحديد إذا ما كانت الإشارة الــصوتية موجودة أو لا من خلال قياس الارتباط. تم اختبار الخوارزمية المقترحة لتحديد مستوى المتانة التي توفرها الخوارزمية المقترحة حيث توضح هذه الورقة البحثية نتائج الاختبارات وكيف تجنبت الخوارزمية الاختبارات لإزالة العلامة المائية.

<u>الكلمات المفتاحية</u>: العلامات المائية في الإشارات الصوتية, إخفاء المعلومات, التحويل المويجي, التزامن.