LOSSLESS AUDIO WATERMARKING BASED ON THE ALPHA STATISTIC MODULATION

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ABSTRACT

In this paper, we propose a high capacity, self-synchronized, lossless audio watermarking algorithm based on the alpha ('α') statistic modulation. Here 'α' is related to the correlation among any given sequence i.e audio samples and it is modulated according to the watermark bit stream. The embedding scheme is tested in both the time domain and DWT domain. Though the time domain embedding reduces the computational time in searching the synchronization codes, the time-frequency localization capability of DWT provides good trade off between the computational complexity and robustness of synchronization codes. In case of DWT, 'α' related to the 2nd level DWT coarse wavelet components is used for embedding the watermark. The offset value used for embedding is made adaptive to the required SNR for the final watermarked audio signal. After extraction of the embedded watermark using a watermark key, original audio can be recovered with minimal distortion. The watermarking method presented here does not require the use of the original signal for watermark detection. Also high embedding capacity is achieved by using small sized audio frames. Experimental results reveal that the proposed watermarking scheme maintains high audio quality and is simultaneously highly robust to pirate attacks, including MP3 compression, cropping, filtering, re-sampling, and re-quantization.

KEYWORDS

Audio watermarking, Digital Rights Management, Lossless Watermark, Self Synchronization, DWT, Robust, Blind.

1. INTRODUCTION

The fast growth of digital audio technology enables the ease of reproducing and retransmitting digital audio. Hence there is a need for the protection and enforcement of intellectual property rights for digital media. Digital watermarking is one of the promising ways to meet this requirement. The primary objective of digital watermarking is to hide the copyright information (e.g. owners/company name, logo etc.) into a multimedia object, without disturbing the perceivable quality of the content [1]. Watermarking of audio signals is more challenging compared to the watermarking of images or video sequences due to wider dynamic range of the human auditory system (HAS) in comparison with the human visual system (HVS). Two properties of the HAS dominantly used in audio watermarking algorithms are frequency masking and temporal masking [1]. According to the International Federation of the Phonographic Industry (IFPI), SNR of watermarked audio signal should be always greater than 20 dB. The

embedded watermarks should not be removed or degraded using common audio processing techniques. The watermark embedding process should be faster, so that integrated watermarking functionality can be enabled in the delivery of an audio over a network. Also it should support fast watermark detection in order to authenticate audio objects, delivered over the networks. According to the IFPI, there should be more than 20 bits per second (bps) data payload for watermark [1-3]. These requirements present great challenges.

Existing audio watermarking techniques are broadly categorized into time domain and transform domain techniques. Time domain techniques [4-8, 13] are simple to realize, but they are less robust compared to transform domain techniques. In transform domain techniques, the host signal is transformed into the transform domain and the watermark is embedded into the transform domain coefficients [3,8,10-12]. Among different transform domain techniques, wavelet domain watermarking is found to be the most effective and efficient as far as robustness is concerned [11, 12]. Further, audio watermarking techniques are also classified into reversible or irreversible techniques based on the possibility of recovery of original audio after extraction of watermark from the watermarked audio. Although reversible watermarking has been widely researched [5-7], the research has mainly been focused on imperceptible reversible image watermarking only. Reversible watermarking enables the embedding of useful information in a host signal without any loss of host information. In [7], the watermark is embedded into selected FFT coefficients' magnitudes of the cover audio using frequency hopping method. However, the embedded watermark is perceptible unless it is removed and original audio is extracted using watermarking secret key. Also most of these reversible techniques are semi-fragile or less robust in nature [5-7].

The main motivation behind our work is to achieve reversibility along with the increased robustness. The work aims at developing a robust reversible audio watermarking scheme that can resist various audio signal processing attacks effectively. Hence as an extension to our previous work [9], here we propose α statistic modulation based lossless robust watermarking method in both Time and DWT domain. In case of time domain, the watermark is embedded into the audio samples directly while in case of DWT, the watermark is embedded into the 2nd level DWT approximate components. In both the cases, non-silent frames are used for embedding in order to take advantage of the perceptual properties of the Human Auditory System (HAS) and to improve the transparency of the digital watermark. Also the proposed scheme modifies the audio samples only if watermark bit=1 is to be embedded. This achieves high imperceptibility. The watermark can be removed using a watermarking key (WK) and original audio data can be retrieved with only minimal remaining distortion. Also in order to resist synchronization attack effectively, watermark is embedded along with synchronization codes [3]. The proposed watermarking method does not require the use of the original signal for watermark detection. Thus, the copyright owner does not have to perform data search in his archives prior to detection. Such searches are very time consuming and render watermarking useless for audio monitoring in digital broadcasting. The binary watermark logo image is first permuted using Arnold Transform in order to increase the secrecy of embedded watermark. To achieve better error detection and correction capability, the combined watermark bit-stream is replicated according to the redundancy factor (r) specified.

The outline of the paper is as follows. Section 2 provides introduction to the α statistic modulation based embedding and extraction technique; Section 3 and Section 4 discuss proposed algorithm in time domain and DWT domain respectively. Experimental results are compared with the results of previous works in Section 5; followed by the conclusions in Section 6.

2. ALPHA '*α*' STATISTIC MODULATION

For any given sequence X of length L, two different sub-sets A and B can be formed as shown in Figure 1, i.e., subset A ={$a_1,a_2,\ldots a_{L/2}$} containing of all odd samples marked by '+', the other subset B={$b_1,b_2,\ldots b_{L/2}$} containing of all even samples marked by '-'.
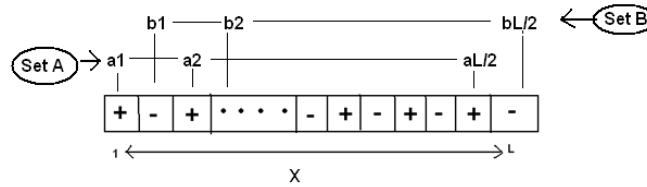

Figure 1. '*α*' Difference pair pattern

Each sub-set has L/2 coefficients. For any given sequence X, the robust '*α*' statistic is defined as an arithmetic average of differences [13] of sample pairs and is given as,

$$\alpha = \frac{1}{L/2}\sum_{i=1}^{L/2}(a_i - b_i)$$

(1)

The distribution of α statistic for an audio frame is as shown in the Figure 2. Most values of '*α*' are very close to zero.
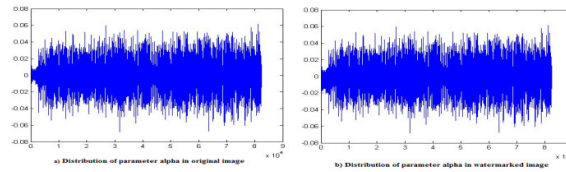

Figure 2. Distribution of α Parameter

As the value '*α*' is based on the statistics of all sample values in the selected audio frame, it has certain robustness against attacks (such as mp3 compression and other slight alteration) so that we can select this '*α*' as a robust quantity for embedding information bit.

2.1 Watermark Bit Embedding Strategy using '*α*'

The threshold value 't' is chosen according to the distribution of '*α*' in the original audio signal.

Case 1: If the difference value '*α*' is located within a defined threshold then,
a) If bit='1' is to be embedded, shift '*α*' to the right side beyond a threshold, by adding a fixed step offset 'S' from each sample value within subset A and subtracting same 'S' value from each sample value within subset B.
b) If bit= '0' is to be embedded, the frame is kept intact. i.e. the samples are modified only if bit=1 is to be embedded. Thus high imperceptibility is achieved, as the probability of being any watermark bit=1 is always 0.5.

Case 2: If the difference value '*α*' is located outside the threshold
No bit is embedded and '*α*' is shifted farther away beyond the threshold. While adding or subtracting 'S', care is taken such that the values should not lead to overflow/underflow problem.

2.2 Watermark Detection using '*α*'

At receiver side, the statistic '*α*' is calculated for embedded audio samples using equation (1).

Case 1: If 'α' is located outside the threshold, then bit '1' is extracted and the embedded audio samples can be restored back to their original values by subtracting 'S' from each sample value within subset A and adding same 'S' value to each sample value within subset B.

Case 2: If the difference value 'α' is within the threshold, then bit '0' is extracted. Once the watermark bit is extracted, the watermarked audio can be restored back to the original audio with minimum distortion. Thus the technique is lossless and reversible.

3. PROPOSED SCHEME IN TIME DOMAIN

The proposed scheme consists of watermark processing stage and audio processing stage. The detailed procedure in case of watermark embedding process is as follows,

3.1 Watermark Embedding Process

Step 1: In watermark processing stage, the binary logo image is first permuted using Arnold Transform in order to enhance the security of the embedded watermark. Arnold Transform is the image transformation technique where the pixels of the image are scrambled [8]. Due to the periodicity of the transform, the image can be recovered after transformation. If watermark image is of size N x N and (x, y) is the coordinate of the image pixel, then (x', y') is the coordinate after applying the transform and given by,

$$\begin{bmatrix} x' \\ y' \end{bmatrix} = \begin{bmatrix} 1 & 1 \\ 1 & 2 \end{bmatrix}\begin{bmatrix} x \\ y \end{bmatrix}(\mod N)$$

(2)

We can decide on the number of times the transform is repeatedly applied on an image in order to generate different results. Due to the Arnold Transform periodicity, the original image can be recovered. Figure 3 shows the effect of selecting different number of iterations (f) for the Arnold Transform.

(a)Original Image   (b)Permuted Image   (c)Permuted Image
                     (After 2 iterations)   (After 10 iterations)

Figure 3. Application of Arnold Transform

Scrambling the watermark before embedding guarantees the embedded watermark will be robust against attacks like clipping, resampling etc. So even if an attacker detects the watermark, he cannot recover the original watermark without the knowledge of scrambling algorithm and the parameters used for the scrambling.

Let $W = \{w(i, j), 0 \le i \le m, 0 \le j \le n\}$ denote permuted binary watermark of size m x n representing the watermark logo where $w(i, j) \in \{0,1\}$ are the respective pixel intensities of the watermark image at coordinate (i, j).

Step 2: The resulting permuted binary image is converted into one dimensional array containing series of 0's and 1's (bits). To enhance the error detection capability of the system, this bit stream is replicated as specified by the redundancy factor (r).

Step 3: The proposed scheme embeds 16 bit Barker code as synchronization code to locate the position of hidden informative bits, thus resisting the cropping and shifting attacks [3]. Barker codes are used for frame synchronization in digital communication systems. They have low correlation side lobes. A correlation side lobe is the correlation of a codeword with a time-shifted version of itself [3]. The correlation side lobe C_k for a k-symbol shift of an N-bit code sequence $\{x_j\}$ is given by,

$$C_k = \sum_{j=1}^{N-k} x_j x_{j+k}$$

(3)

where x_j is an individual code symbol taking values +1 or -1 for $j = 1,2,..,N$ and the adjacent symbols are assumed to be zero.

Step 4: In audio processing stage, the original host audio is first segmented into non-overlapping audio frames of size M samples. For convenience, we take M as an even number. If $X = (x(1), x(2),...,x(N))$ denotes the original audio signal having size N where, $x(i) \in (-1.0,+1.0)$ are respective sample values normalized in the given range.
Then the audio segments are given as,

$$Y_k = (X_{M(k-1)+1}, X_{M(k-1)+2},..., X_{Mk})$$

(4)

where $k = 1,2,..,N_s$ and N_s =total number of audio segments=N/M after appropriate padding.

The embedding capacity also depends on this N_s , as each frame can embed one bit of watermark

Step 5: As distortion becomes noticeable in silent parts of the signal, skip all silent audio frames that are present in the audio signal. Hence only non-silent frames are considered for further embedding process.

Step 6: Decide the offset value used for the embedding stage as S=S0 initially. This S is made adaptive to the minimum SNR value required to be maintained for final watermarked signal. This also retains transparency of the embedded watermark.

Step 7: Split selected audio frame into two different sub-sets A and B as shown in Figure 1 and calculate the statistic 'α' using equation (1). Embed the watermark bits by modulating 'α' using bit embedding strategy defined earlier.

Step 8: Reconstruct watermarked audio by combining all frames sequentially, after embedding. The SNR (Signal to Noise Ratio) of this watermarked audio with respect to the original audio signal is calculated and compared with the expected value of the SNR and Offset value 'S' is changed adaptively. The experimentation shows that after embedding the 1024 bits of watermark bits the stego audio gives PSNR value more than 40dB.

Step 9: The Unique Watermarking Key (WK) is generated and distributed along with the watermarked audio signal. Different parameters used during embedding process like Energy Threshold Factor 'E$_{th}$', Redundancy factor 'r' , offset value 'S' chosen, Arnold Transform frequency parameter etc. are used for generating WK. Only authenticated user holding valid WK can remove watermark from the watermarked audio and can recover original audio with minimal distortion.

3.2 Watermark Detection Process

The watermark extraction algorithm consists of all the audio processing steps that are carried out at the time of embedding the audio frames. All required parameters are extracted from WK by the receiver. First stego audio signal is segmented into non-overlapping frames of size M samples each. Then select all non-silent audio frames.

The statistic 'α' is calculated for each selected audio frame using equation (1) and watermark bit is extracted comparing 'α' with the threshold. Once all bits are extracted, the watermark logo image can be reconstructed by first detecting the synchronization codes. The original audio is not required in the extracting process and thus the proposed algorithm is blind. Also once the watermarked is extracted, the watermarked audio can be restored back to original audio with minimum distortion. Thus the technique is lossless and reversible. Also as the receiver has an access to the original watermark information, that can be used to increase the quality of restored audio further. The distortion caused is not perceptually audible as only non-silent frames are modified.

4. PROPOSED SCHEME IN DWT DOMAIN

Figure 4 shows both proposed scheme for watermark embedding and extraction process in case of DWT domain. All the steps up to Step 6 will be same as that of previous scheme.

According to the multi-resolution characteristic of DWT, select the proper wavelet family to decompose selected frame to the 2nd level. Here, each frame is decomposed into two parts, the high frequency/Detail components and the low frequency/coarse components by passing it through a series of high pass filters and low pass filters. Let [CA1, CD1] denotes the 1st level DWT coarse and detail components while [CA2, CD2] denotes 2nd level DWT coarse and detail components of a selected audio frame. In order to make the watermarked signal inaudible, we embed the watermark into low frequency part of the non-silent audio frames containing some energy, thus taking advantage of frequency mask effect of HAS [10].

To take the advantage of low frequency DWT components which has a higher energy value and robustness against various signal processing, CA2 is chosen for embedding. Calculate the statistic 'α' on CA2 sub-band using equation (1). Embed the watermark bits by modulating 'α' using bit embedding strategy defined earlier. Here as 'α' is based on the 2nd level DWT coarse coefficients; it has more robustness against different attacks compared to that of time domain embedding scheme.

After embedding, watermarked audio is reconstructed by applying Inverse Discrete Wavelet Transform (IDWT) and combining all frames sequentially. The Unique Watermarking Key (WK) is generated as step 10.
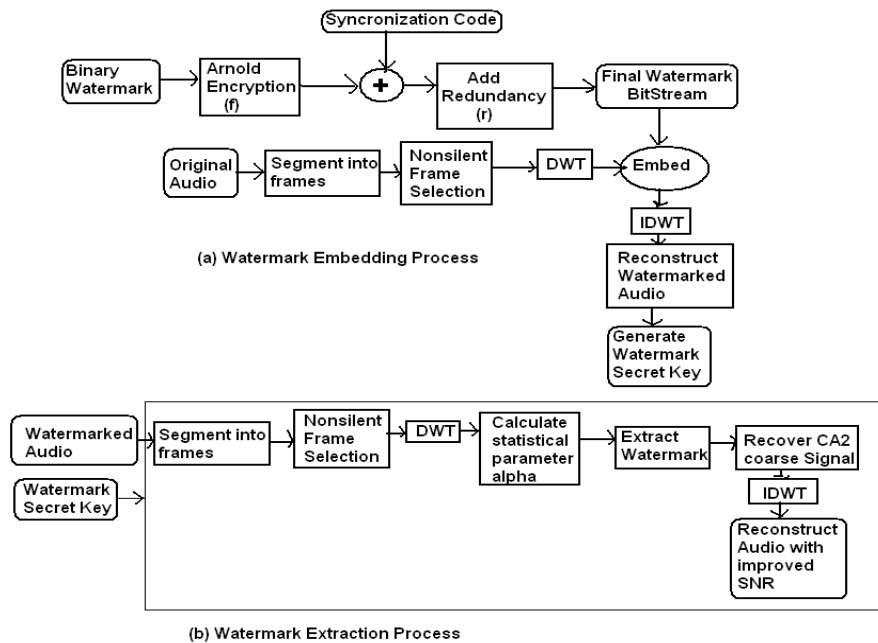
Figure 4. Proposed audio watermark Embedding and Extraction Technique

4.1 Watermark Detection Process

As shown in Fig 4 (b), the extraction algorithm consists of all the audio processing steps that are carried out at the time of embedding the audio frames. All required parameters are extracted from WK by the receiver. First stego audio signal is segmented into non-overlapping frames of size M samples each. Then select all non-silent audio frames.

The difference value 'α' is calculated for CA2 sub-band of each selected audio frame in DWT domain using equation (1). If the difference value 'α' is outside the threshold, then bit '1' is extracted and the CA2 coefficient values of that frame can be restored back to its original value. If the difference value 'α' is within the threshold, then bit '0' is extracted. In this way, we can extract the watermark bits along with the synchronization codes. Once all the bits are extracted, the watermark logo image can be reconstructed and the watermarked audio can be restored back to original audio with minimum distortion. Thus reversibility along with increased robustness is achieved in DWT domain. Experimental results show that a maximum offset value S='0.055' can be used without affecting the perceptual audio quality of the host signals. For the stereo audio signals, dual-channel signals are available for watermarking, while in case of a mono audio signals; only one single-channel signal is available for watermarking.

5. EXPERIMENTAL RESULTS

5.1 Experimental setup

To assess the performance of the proposed audio watermarking scheme, several experiments are carried out on different types of 30 mono audio signals of length 20 seconds each. These CD Quality audio signals are sampled at sampling rate 44.1 KHz with 16 bit resolution. These audio signals are categorized into following categories; the rock music (denoted by A1) that has very high signal energy, classical music (denoted by A2) and speech signal (denoted by A3) that has moderate signal energy as shown in Figure 5.
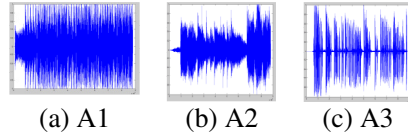
(a) A1   (b) A2   (c) A3

Figure 5. Original audio signals (a) Rock music (b) classical music and (c) speech signal

The ownership information is represented by a 32x32 binary logo image (w) as shown in Figure 6. The logo image is first permuted using Arnold Transform and converted into a one dimensional bit stream of 1's and 0's. After adding synchronization codes, it is replicated using redundancy factor r=3.

Figure 6. 32x32 binary logo watermark images used

The data payload (D) refers to the number of bits that are embedded into the audio signal within a unit of time. It is measured in the unit of bps (bits per second). If the sampling rate of an audio signal is F(Hz) and the frame size is Fs (i.e. number of samples per frame), then the data payload of the proposed algorithm is given as,

$$D = \frac{F}{Fs}(bps)$$

(5)

In case of DWT domain, for a frame containing 128 samples, the estimated data payload is 344 bps. Without applying any redundancy, it needs an audio section about 3.02 seconds in order to embed a 16 bit synchronization code (1111100110101110) along with a 32x32 binary watermark. With r=3, it needs an audio section about 9.06 seconds. To achieve good robustness against different attacks, the value of offset value/embedding strength (S) can be increased upto 0.055 maximum for all audio files. The Haar wavelet basis is used. The smaller level DWT will influence the robustness of the watermark; and the larger one will cause large calculation, so 2nd level DWT is performed in this test.

5.2 Perceptual Quality Measures

To measure imperceptibility, we use signal-to-noise ratio (SNR) as an objective measure and a listening test as a subjective measure. SNR is based on the difference between the undistorted original audio signal and the distorted watermarked audio signal. If A corresponds to the original audio signal, and A' corresponds to the watermarked audio signal then, SNR is given as,

$$SNR = 10 \log_{10}\left(\frac{\sum_n A_n^2}{\sum_n (A_n - A'_n)^2}\right) dB$$

(6)

where n=total length of audio signal. Table 1 lists the corresponding average SNR values of watermarked audio. After extraction of watermark bits, the watermarked audio is restored back with improved SNR values.

Table 1. SNR and MOS values for audio signals

Audio File	Watermarked Audio SNR(dB)	Restored Audio using Extracted Watermark SNR(dB)	Restored Audio using Original Watermark SNR(dB)
A1	34.21	42.71	78.18
A2	33.14	46.47	80.44
A3	30.14	44.21	78.50

5.3 Effect of Offset value and Frame Size on SNR

We have checked the effect of varying offset value S along with different Frame Size Fs on the quality of final watermarked audio. Empirically it is found that as S increases, SNR decreases. As Fs increases, SNR decreases. But the robustness increases with both S and Fs. So a good trade off among different parameters is needed in order to achieve good transparency of embedded watermark along with sufficient robustness against common intentional attacks.

5.4 Robustness Test

Both Normalized Correlation (NC) and Bit Error Rate (BER) between the original watermark and the extracted watermark are used as an objective measure for the robustness and calculated using following equations;

$$NC = \frac{\sum_{i=0}^{m}\sum_{j=0}^{n} w(i,j)w'(i,j)}{\sqrt{\sum_{i=0}^{m}\sum_{j=0}^{n}|w(i,j)|^2}\sqrt{\sum_{i=0}^{m}\sum_{j=0}^{n}|w'(i,j)|^2}} \qquad (7)$$

where $w(i,j)$ is an original watermark and $w'(i,j)$ is the extracted watermark.

$$BER = \frac{Number\_of\_bits\_in\_error}{Total\_number\_of\_bits} * 100\% \qquad (8)$$

The performance of both schemes is compared. The proposed algorithm gives moderate SNR values along with good amount of embedding capacity and lower bit error rates as shown in Table 2. The NC values are always above 0.9 for most of the common audio processing attacks.

Table 2. Comparison DWT and Time Domain Schemes

Algorithm	NC Under MP3 (128kbps)	BER (%) Under MP3 (32kbps)	SNR (dB)
Our Scheme in DWT Domain	0.9(approx)	<4.26%	> 40 always
Our Scheme in Time Domain [9]	0.7(approx)	<25%	

The performance of the proposed algorithm is also compared with the algorithms proposed by Kamalika Datta and Indranil Sengupta, 2010 [11] and Chen, Huang, Chen and Wu [12]. The results summarized in Table 3. Computational Time increases with the level of DWT used.

Table 3. Algorithm Comparison

Algorithm	Reversible Technique	DWT level	Synchro-nization Codes	SNR (dB) (Approx)	NC for MP3 (32kbps) (Approx)	Payload (bps)
Our	Yes	2nd	Yes	>40	0.9	344
[11]	No	3rd	No	30-40	0.9	200
[12]	No	7th	No	30	Good, Value Not mentioned	115

The watermarked audio signal is subjected to several standard audio processing attacks, in order to assess the robustness of the proposed watermarking scheme. The results are summarized in Table 4. From the results, it can be seen that the proposed audio watermarking scheme is robust to most of the common audio processing attacks. In case of stereo audio signal, same watermark information or the meta data used during the embedding process (redundancy factor, length of

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watermark, Arnold Transform Frequency, value of S etc.), can be embedded in the second audio channel. This can also help the receiver to retrieve and verify the watermark blindly.

Table 4. NC and BER values for various attacks

Attack	Robustness	
	BER (%)	NC (%)
No attack	0.0	1.00
White noise (awgn)	9.2	0.92
Cropping (20%)	13.52	0.65
Re-sampling (22.05 kHz)	4.79	0.89
Re-sampling (11.025 kHz)	9.25	0.76
Re-sampling (8.00 kHz)	12.18	0.64
Re-quantization (8 bit)	0.0	1.00
Re-quantization (24 bit)	0.0	1.00
LPF (6 order Butterworth, 22.05 kHz)	9.11	0.82
MP3 compression with the rate of 48 kbps	13.30	0.75
MP3 compression with the rate of 32 kbps	16.44	0.69
Amplitude (increased by 0.2)	4.00	0.91
Amplitude (increased by 0.4)	5.18	0.89

6. CONCLUSIONS

The main motivation behind our work is to achieve reversibility along with the increased robustness. In this correspondence, we propose a blind, high capacity, self-synchronized, lossless audio watermarking algorithm based on the alpha ('α') statistic modulation. It is found that the time-frequency localization capability of DWT provides good trade off between the computational complexity and robustness of synchronization codes compared to time domain. After extraction of the embedded watermark using a watermark key, original audio can be recovered with minimal distortion. Another advantage of the scheme is that the audio samples are modified only if watermark bit=1 is to be embedded. Thus high imperceptibility is achieved, as the probability of being any watermark bit=1 is always 0.5. Experimental results show that the embedded watermark is perceptually transparent and for most of the attacks, the normalized correlation coefficient is more than 0.9. Further research will focus on achieving more robustness towards time scale modification attacks.

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