A ROBUST AUDIO WATERMARKING SCHEME BASED ON LIFTING WAVELET TRANSFORM

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Abstract:
Digital watermarking is a technique to protect ownership and copyright protect of multimedia content. Many effective water marking algorithms have been proposed and implemented for audio signals. This is due to the fact, the human audio system is far more complex and sensitive than the human visual system. In this paper, an audio watermarking algorithm based on Lifting Wavelet Transform (LWT) is proposed. The LWT transformation decomposes the audio signal into detailed and approximation coefficients, enabling developers to locate the most appropriate location to embed the data. The watermarking data is embedded into the approximation coefficients of audio file by using an embedding technique. Inorder to obtain the robustness and imperceptibility and the results will be compared with the DWT transformed audio watermarking signals.

Keywords: LWT, Robustness, Copyright protection, Imperceptibility.

I.INTRODUCTION:

Watermarking is the one of the popular technique to protect the owner copyrights of the multimedia content (audio, image, video or text). Digital audio watermarking is a technique in that, owner information or any digital content as a watermark is embedded without losing the quality of the audio content. Audio watermarking have several advantages, such as content authentication, data hiding, fingerprinting, copy protection, broadcast monitoring and copyright protection. According to International Federation of Phonographic Industry (IFPI), an effective audio watermarking scheme should meet the following requirements: (a) Imperceptibility: embedded watermark should not degrade the quality of the original audio signal, (b)Robustness: the embedded watermark data should not be removed or eliminated by third party using common signal processing operations such as, resampling, requantization, cropping and so on. (c) Security: without knowing the secret key, nobody cannot detect the watermark.

According to state of art, digital audio watermarking algorithms are classified into two categories, Time domain and Frequency domain. Time domain audio watermarking algorithms are very easy to implement but poor in robustness. Frequency techniques are hard to implement but robustness is good in comparison with time domain techniques.

II. FUNDAMENTAL THEORY

Lifting Wavelet Transform:

Lifting Wavelet Transform was started as a method to improve a given discrete wavelet transforms to obtain specific properties. Later it became an efficient algorithm to calculate any wavelet transform as a sequence of simple lifting steps. It allows a faster implementation of the wavelet transform. It requires half number of computations as compare to traditional convolution based discrete wavelet transform. This is very attractive for real time low power
applications. it is easier to understand and implement. It can be used for irregular sampling.

Singular value decomposition (SVD):

The singular value decomposition (SVD) matrix is very useful for analyse matrices. It is an algorithm developed for a variety of applications. In SVD transform, let us consider a matrix M is decomposed into three matrices,

$$M = USV^T$$

Where,S is the diagonal matrix that is

$$\begin{pmatrix} \rho_1 & 0 & . & 0 & 0 \\ 0 & \rho_2 & : & 0 & 0 \\ . & . & & . & . \\ 0 & 0 & . & \rho_{n-1} & 0 \\ 0 & 0 & . & 0 & \rho_n \end{pmatrix}$$

And U and V are orthogonal Matrices, Thus

$$U^T U = V^T V = I$$

The diagonal elements of $(S_1, S_2, \ldots, S_n, S_{n-1})$ are called singular values(SVs) ,The SVD as some interesting properties:

1) M and its transpose (M^T)have same non zero values.
2) $M$and its flipped version round rows and columns have same non zero values.
3) Changing SVs slightly doesn't affect the quality of the signal much.
4) The SVs are invariant under common signal processing operations.

Cartesian-polar transforms:

Consider the polar coordinate system$(r,\ \theta)$, Where r represents the distance from the origin and θ represents the angle between a line of reference and the line through the origin and the point. The Cartesian coordinate to the polar coordinate is given by the following equation:

$$r = \sqrt{x^2 + y^2}, \qquad \theta = tan^{-1}\left(\frac{y}{x}\right)$$

Where (x, y) is a point in Cartesian coordinate system. The polar coordinate to the Cartesian coordinate is given by the following equation:

$$x = rcos\theta, \qquad y = rsin\theta$$

III. WATERMARKING SCHEME:

In this section, summary of both watermark embedding and detection process is presented. Suppose x is an original audio signal, the length of the audio signal is L. As follow:

$$x = \{x(n)|0 \leq n < L\}$$

Where $x(n)$ is the value on the magnitude of n^{th}.

watermark pre-processing:

Water mark should be pre-processed first in order to improve robustness and increase the confidentiality. In our watermark embedding scheme, the binary image is scramble from W to W_1 by using Arnold transform, where

$$W_1 = \{w_1(i,j)|0 \le i \le M, 0 \le j \le N\}$$

Then, it is transformed in to one-dimensional sequence of ones and zeros as follows:

$$W_m = \{w_m = w_1(i,j)|0 \le i \le M, 0 \le j \le N, m = (i-1) \times N + j, w_m \in \{0,1\}\}$$

Synchronization Code:

The synchronization code is used to locate the position of hidden informative bits, thus resisting the cropping and shifting attacks.In this scheme, the host signal is partitioned into two portions for synchronization code insertion and watermark embedding. This algorithm uses a chaotic sequence as the synchronization code in front of the watermark to locate the position where the watermark is to be embedded. The logistic chaotic sequence with initial value in the interval [0,1] is selected to generate the synchronization code and denoted as:

$$y(n+1) = g\big(y(n)\big) = \delta \times y(n)(1 - y(n))$$

Where$3.57 < \delta < 4$, $y(n)$ is mapped into the synchronization sequence $\{syn(n)|n = 1,2 \dots, L_{syn}\}$with the following rule:

$$syn(n) = \begin{cases} 1 & if\ y(n) > T \\ 0 & otherwise \end{cases}$$

Where $L_{syn}$the synchronization code length and T is pre-defined threshold, sufficient value of T is 0.5.

Watermark embedding process:

The embedding process is shown in fig.

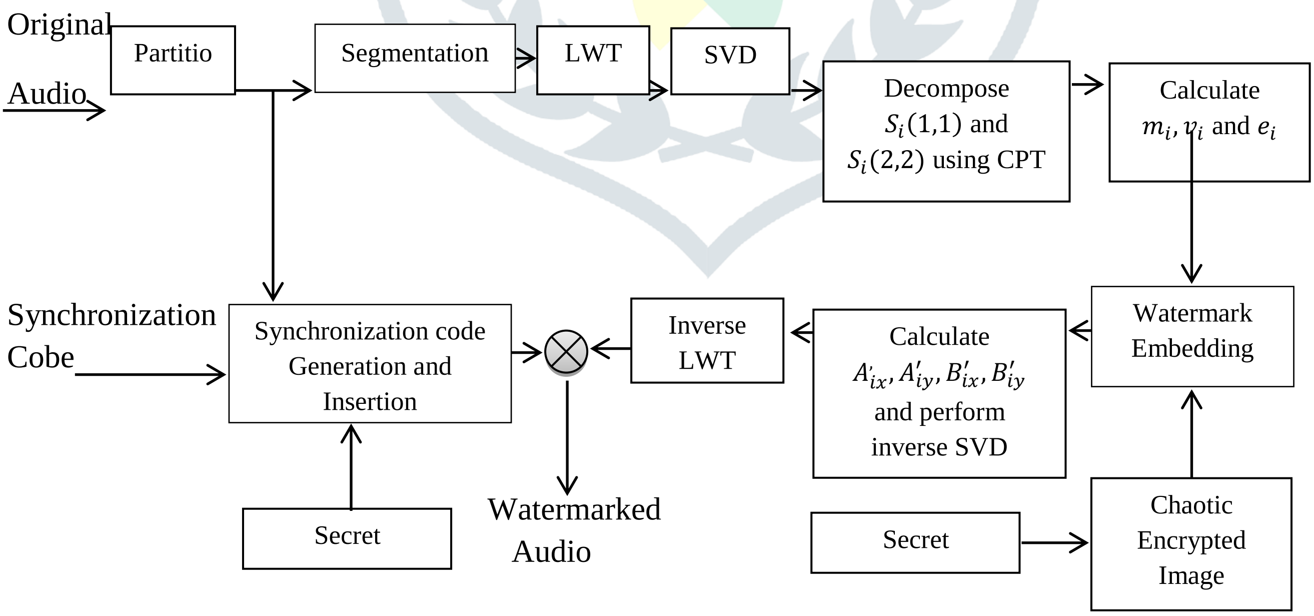

Fig. 1: Overview of watermark embedding process.

Watermark detection process:

The detection process is shown in fig.

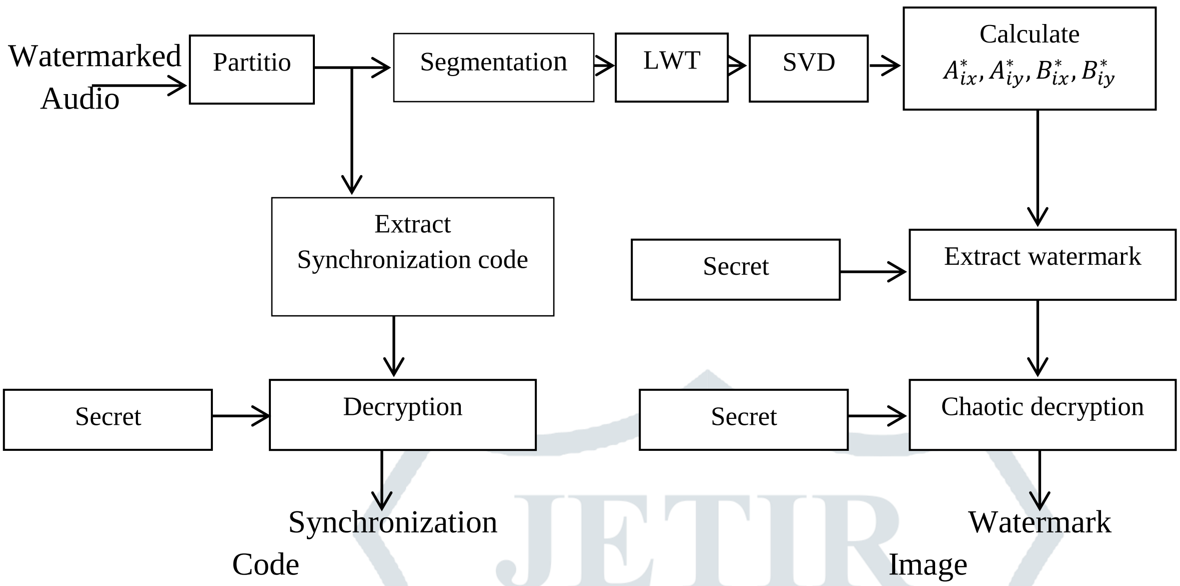

Fig. 2: Overview of watermark extraction process.

IV. SIMULATION RESULTS:

In this section, several experiments were conducted to demonstrate the performance of the proposed watermarking scheme. The performance of the proposed scheme is evaluated in terms of imperceptibility, robustness, and data payload. In this study, we selected four different types of 16 bit mono audio signals (Pop, Blue, Rock, Folkcountry and Funksoulrnb) sampled at 44.1 kHz. Each audio file contains 443,520 samples (duration 10 s). By using a frame size of 64. We embed one binary watermark bit in each frame of audio signal. Thus, the length of the watermark sequence is. A binary logo image and the corresponding encrypted image using chaotic encryption of size $M \times M = 96 \times 96 = 9216$ are shown in Fig. 3. From each frame of audio signal, we select first thirty six low frequency FFT coefficients ($l = 36$). After rearranging the selected FFT coefficients of each frame in a 6×6 matrix ($N = 6$), SVD is applied to each of these matrices.

(A) (B)

Fig.: (A) Binary watermark image, (B)Encrypted image

Imperceptibility test:

Imperceptibility is related to quality of audio signal. In order to evaluate quality of audio signal SNR equation is used:

$$SNR(x, x^*) = 10\log_{10}\left(\frac{\sum_{i=0}^{length-1} x^2(i)}{\sum_{i=0}^{length-1}[x(i) - x^*(i)]^2}\right)$$

Where $x(n)$and $x^*(n)$ are two original and watermarked audio signal, respectively. According to the IFPI (International Federation of the Phonographic Industry) recommendation (Bhat et al. 2010), audio watermarking should be imperceptible when SNR is over 20 dB. After embedding a watermark, the SNR values of all selected audio signals using the proposed scheme are above 20 dB, conforming to the IFPI standard, as shown in Table 1.

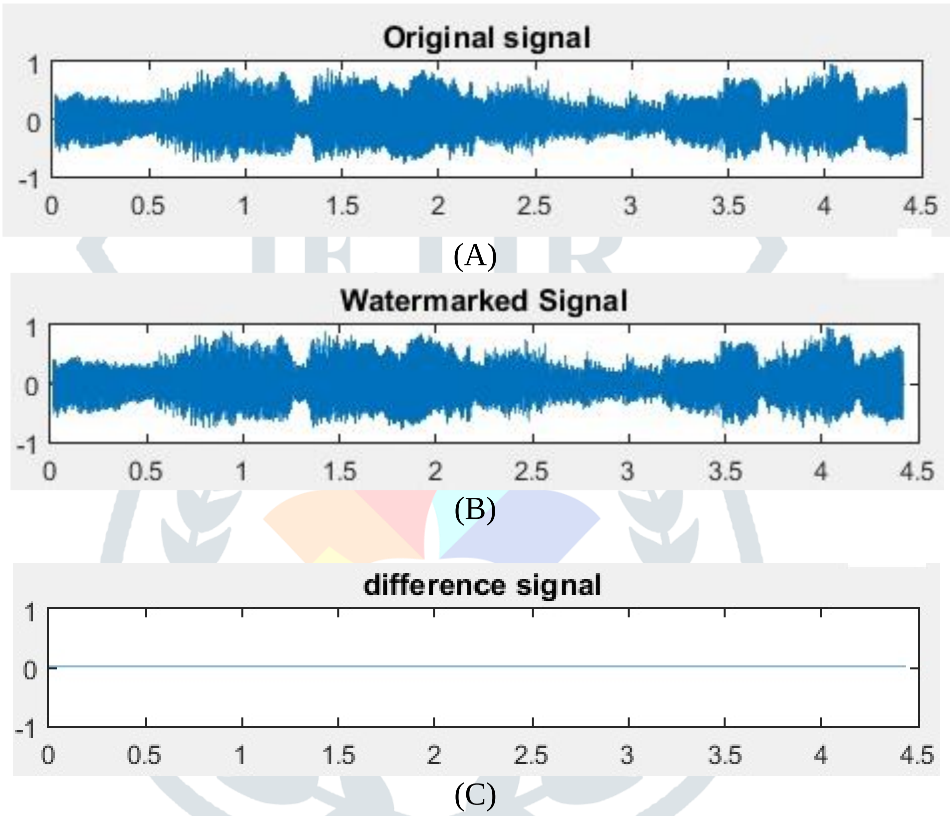

Fig 1: Imperceptibility of watermarked audio signal: (A) Original audio signal 'pop', (B) Watermarked audio signal 'pop', (C) Difference between original and watermarked audio signal

Type of signal	SNR		
	FFT	DWT	LWT
Pop	51.4675	54.5704	56.1785
Rock	52.7716	54.3659	56.3597
Folkcountry	52.3313	54.4046	56.0804
Blue	50.3467	53.6865	55.1941
Average	51.7340	54.2568	55.9531

Table 1: Imperceptibility test of this scheme for different watermarked sounds

Figure 1 show the time domain representation of the original audio signal with a watermarked audio signal in which the watermark is imperceptible using the proposed scheme.

Robustness test:

To compare the similarities between the original watermark $\boldsymbol{W}$and the extracted watermark $\boldsymbol{W}^*$normalized correlation (NC) coefficient is used, which is computed as:

$$NC(W^*, W) = \frac{\sum_{i=0}^{M-1}\sum_{j=0}^{M-1} w(i,j) * w^*(i,j)}{\sqrt{\sum_{i=0}^{M-1}\sum_{j=0}^{M-1} w^2(i,j)}\sqrt{\sum_{i=0}^{M-1}\sum_{j=0}^{M-1} w^{*^2}(i,j)}}$$

where$i$ and j are the indices of the binary watermark image. The correlation between $W$and $W^*$is very high if $NC(W, W^*)$is close to 1. On the other hand, the correlation between $W$and $W^*$is very low if $NC(W, W^*)$is close to zero.

The BER is used to measure the robustness of a watermarking scheme and is computed as:

$$BER = \frac{\sum_{i=0}^{M-1}\sum_{j=0}^{N-1} w(i,j) \oplus w^*(i,j)}{M * M} * 100\%$$

Type of signal	NC	BER	Data PayLoad
Pop	1	0	1.0128e+03
Rock	1	0	1.0128e+03
Folkcountry	1	0	1.0128e+03
Blue	1	0	1.0128e+03

Table 2: Extracted watermark image with NC and BER for different audio signal

Table 2 shows the robustness results of the proposed scheme in terms of NC and BER against several attacks for the audio signal ‘Pop’. It is obvious that NC values after attacks are very high while the BER values are very low. The minimum NC value and the maximum BER value are 0.9665 and 7.5470, respectively. The extracted watermark images are visually similar to the original watermark, which further verify the good performance of the proposed scheme against different attacks.

The robustness results for the audio signal ‘Rock’, ‘Folkcountry’, ‘Blue’and ‘Funksoulrnb’, respectively are summarized in Table 3. Weobserved that the NC and BER values range from 0.7871 to1 and 0 to 24.4672, respectively, demonstrating the high robustnessof our proposed scheme against different attacks.This is because watermark bits are embedded into the CPTcomponents of the highest singular values of the low frequencyFFT coefficients of each audio frame.

Table2 shows the robustness results of the proposed scheme in terms of NC and BER for the different audio signals ‘pop’,’rock’,’folkcountry’,’blue’.we observe the NC and BER values 1 and 0.

Security:

Robustness against attack is very important for a secured watermarking scheme. To enhance the security, the proposed method utilizes chaotic encryption. Since the proposed watermark embedding and detection processes depend on the secret keys $K_1$and K_2, it is impossible to maliciously detect the watermark without these keys.

Data payload:

The data payload defined as the number of bits that can be embedded into the original audio signal within a unit of time. It can be measured in bits per second (bps). Data payload ***DP*** is defined as:

$$DP = \frac{N_w}{Time}$$

Where $Time$ is duration of host audio signal in seconds and N_w is the no of watermark bits that can be embedded into the host audio signal. In our scheme, N_w= 9344 bits is embedded in 10 sec host audio signal, thus the pay load of our method is 934.4bps. This is relatively high payload as typical payload is 20-50bps.

V. CONCLUSION

An audio watermarking scheme based Lifting Wavelet Transform(LWT) based Singular Value Decomposition(SVD) and Cartesian-Polar Transform(CPT) is simulated .Lifting Wavelet Transform is used in the place of FFT and DWT in the same algorithm and same performance is compared .The performance analysis indicates that robustness and imperceptibility of the LWT algorithm outperforms the algorithm based on FFT and DWT.

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