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Slantlet Transform Based Digital Audio Watermarking for Copyright Protection of Audio Content

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Abstract*: Digital audio watermarking (DAW) serves an important role in the protection of copyrights of digital audio content. Many DAW schemes are available in the literature; achieving a balance among the imperceptibility, robustness and payload is always substantial. For this, a blind DAW based on Slantlet Transform (ST) is proposed in this paper. Pre-processed watermark is embedded in transformed audio coefficients. The proposed (ST DAW) method is evaluated on four classes of audio signals viz., pop, rock, jazz and folk country. Experimentation is carried out to validate its performance in terms of imperceptibility, robustness and payload. Comparative analysis of the ST DAW method with state-of-art methods proves its superiority.

***Index Terms*: Audio Watermarking, Gaussian chaotic map, Payload, Quantization Index Modulation, Slantlet Transform.**

I. INTRODUCTION

Digital audio watermarking became an imperative solution for audio copyright protection. These watermarking methods are generally categorized into spatial and transform domain methods. The transformed audio watermarking methods performance is good comparatively spatial domain methods in terms of robustness. In transform domain methods, a watermark is embedded in transformed audio coefficients. Those transforms may be DCT, DWT, FFT, LWT or FWHT.

Authors [1] and [2] proposed multi-transformed watermarking based on DWT and DCT on audio signals. Synchronous pattern is inserted in 11th approximation sub-band and watermark in the form of bits are inserted in 1st to 9th detail sub-bands. DCT is applied for these detail sub-bands and then scrambled watermark is embedded through Rational Dither Modulation (RDM) and Windowed Vector Modulation. The authors reported that, this method achieves payload of 86bps. In [3], authors presented a blind DAW scheme using DCT, SVD and exponential logarithm operations. Here, highest singular values of the exponential coefficients which are obtained from the DCT sub-band with highest power of each audio frame is considered to insert the watermark. This resulted in an imperceptibility of 33.47dB and payload of 172.39bps. Authors [4] proposed two algorithms for improving the DWT-RDM audio watermarking performance. Both schemes operate on a 4th-level approximation sub-band of DWT. The first scheme includes the usage of a two-tap FIR low-pass filter to degrade the noise it is induced by inserting the watermark. The second scheme is developed to address the vector modulation distortion. Both enhanced schemes achieves payload of 689bps. A blind audio watermarking with multiple transforms is presented in [5]. Audio is divided into segments and binary watermark is embedded into transform domain coefficients which are obtained by applying DWT and DCT. Authors claimed that this method reaches SNR of 23.49 dB and payload of 172.27 bps. A high capacity DAW method based on FFT is presented in [6]. Here, Fibonacci number sequence is used to embed the watermark into the selected FFT coefficients. This scheme achieved a capacity of 700 bps to 3 kbps. In this proposed work, slantlet transform is used and this transform has more time-localization when compared to DWT. Earlier this transform is used in image compression [7] and denoising [8], now this is extended to watermarking [9]. In this paper, slantlet transform is used for audio watermarking to convert time samples to transformed coefficients. Pre-processed watermark is embedded in these transformed coefficients based on QIM mechanism. This paper is organized as follows: Methods used in this paper to develop the audio watermarking is discussed in section 2. Watermark embedding and extraction procedure is discussed in section 3 and section 4 respectively. Results and discussions are detailed in section 5. Conclusion remarks are given in section 6.

II. METHODOLOGY

Slantlet transform is used in this proposed method to convert the original audio into transform coefficients, QIM is used to achieve blind nature of watermarking and the Gaussian chaotic map is used for pre-processing of binary watermark. These detailed discussions are presented in this section.

A. Slantlet Transform

DWT is the more useful transform and it has more applications due to the property of time localization.

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Similar transform to get more time-localization is slantlet transform. Two level synthesis filter bank for slantlet transform and its analysis filter bank is shown in Fig. 1.

Because of the shorter length of the filters, slantlet transform filter banks are less frequency selective than the traditional DWT. One more advantage of slantlet transform is its filters are piecewise linear [10]. Compression and reconstruction of an input signal using slantlet transform is shown in Fig. 1. Here, E1(Z), E2(Z), E3(Z) and E4(Z) are analysis filters used to compress the given signal and E1'(Z), E2'(Z), E3'(Z) and E4'(Z) are synthesis filters used to reconstruct the signal.

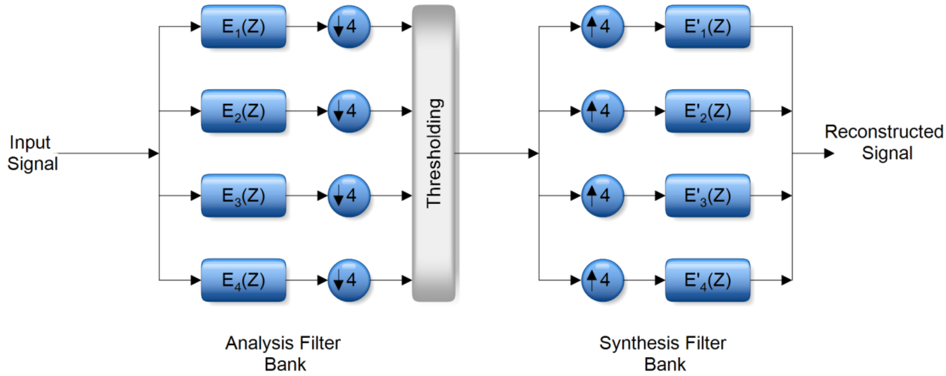

Fig. 1: Slantlet transform Analysis and Synthesis filters

A. Quantization Index Modulation (QIM)

To achieve blind property of watermarking method, QIM is the most powerful mechanism in transform domain audio watermarking methods [11]. In QIM mechanism, embedding of binary bits uses the (1).

$$e(i) = \begin{cases} round\left[\frac{x(i)}{Q}\right]Q, & if\ E_i = 0 \\ (floor\left[\frac{x(i)}{Q}\right]Q) + \frac{Q}{2}, & if\ E_i = 1 \end{cases} \quad (1)$$

where, $x(i)$ is original signal, E_i is the binary bits and Q is the embedding strength.

Equation (2) is used to extract the binary bits of the watermarked signal is as follows:

$$E_n^{'} = \begin{cases} 1 & if\ \frac{Q}{4} \le \mathrm{mod}(y^{'}(n), Q) < \frac{3Q}{4} \\ 0 & otherwise \end{cases} \quad (2)$$

where, $E^{'}(n)$ is extracted binary bits and $y^{'}(n)$ is watermarked signal coefficients.

B. Gaussain Chaotic Map

Chaotic maps are simple and more useful techniques for scrambling the image to enhance the security in audio watermarking methods. In this proposed method, Gaussian chaotic map is used and scrambled by using the process as follows:

$$P_{n+1} = e^{(-\alpha P_n^2)} + \beta \quad (3)$$

Where P_n is the initial value in the range of 0 to 1. α and β are the real parameters.

$$G_n = \begin{cases} 1 & if\ P_n > \frac{1}{4} \\ 0 & otherwise \end{cases} \quad (4)$$

$M \times M$ size watermark image is converted to one-dimensional vector W_n and this is encrypted with G_n with the help of (5).

$$E_n = G_n \oplus W_n \quad (5)$$

III. WATERMARK EMBEDDING PROCESS

In this embedding process, the scrambled binary watermark is concealed in transform domain audio signal using QIM mechanism. The detailed process is given below:

Step 1: Divide the original audio into segments.
Step 2: Select one segment randomly to embed the watermark and this has acted as a secret key while extraction the watermark.
Step 3: Convert one dimensional selected segment into two dimensional segment of size N X N.
Step 4: Apply slantlet transform to that matrix.
Step5: Binary watermark of size N X N is scrambled (pre-processed) using a Gaussian map.
Step 6: This pre-processed watermark is embedded into transformed audio segment using QIM mechanism.
Step 7: Apply inverse slantlet transform and combine all segments to get a watermarked audio signal.

IV. WATERMARK EXTRACTION PROCESS

A watermark is extracted from the watermarked audio without involvement of original audio and this detailed extraction process steps are given below:

Step 1: Divide the watermarked digital audio into segments.
Step 2: Select one segment based on the secret key that is used in the embedding steps.
Step 3: Convert this 1-D audio segment to 2-D matrix.
Step 4: Apply slantlet transform

Step 5: Extract scrambled binary bits from the transformed coefficients using the QIM mechanism without using an original audio signal.
Step 6: Apply inverse Gaussian chaotic map to get the extracted binary watermark.

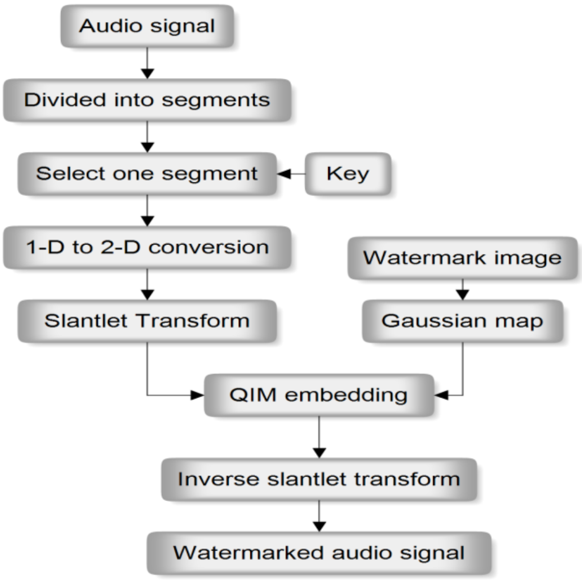

Fig. 2: Watermark embedding flowchart

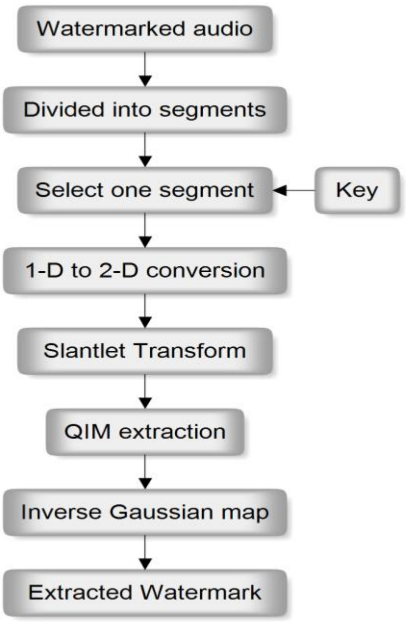

Fig. 3: Watermark extraction flow chart

V. EXPERIMENT RESULTS AND DISCUSSION

This proposed method is tested for four classes of audio i.e., POP, ROCK, JAZZ and FOLKCOUNTRY audio signal each has 44,100Hz sampling frequency, 16-bit and 10 Sec is shown in figure. A watermark of 256 X 256 size binary image is considered in this proposed work.

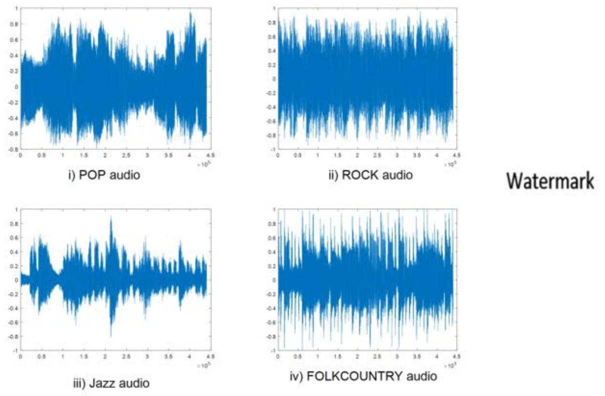

Fig. 4: Four classes of audio signals (i-iv) and original watermark

Imperceptibility is measured in terms of SNR and robustness is measured in terms of BER and NC for this proposed method. Average SNR for four classes of audio signals got 39.055 dB and it is greater than 20 dB as per IFPI recommendations shown in Table 1.

Table 1: SNR of four classes of audio

Audio signal	SNR in dB
POP	39.9349
ROCK	41.2428
JAZZ	34.9468
FOLKCOUNTRY	40.0956
Average	39.055025

Ability to extract the watermark even after applying the signal processing operations on watermarked audio is called robustness of that method. Robustness is measured for the proposed method by applying different signal processing operations i.e., resample (44.1kHz-22.05kHz-44.1kHz), requantization (16 bit-8 bit-16 bit), random noise(40 dB), filter (cutoff frequency = 20kHz), cropping (starting, middle and ending), MP3 compression (256 kbps), Additive White Gaussian Noise (60 dB and 50 dB), signal addition (added 2000 samples of original audio), signal subtraction (2000 samples of original audio is subtracted). BER and NC values of extracted watermark for four classes of audio signals under signal processing operations are mentioned in Table 2.

Table 2 indicates that the proposed method achieved better BER and NC values for all the signal operations except for resampling attack. Number of bits that can be embedded into an audio signal without disturbing the audio quality is referred as payload. This proposed method achieved a payload of 6.5 kbps and it is much greater than the 20 bps specified by IFPI.

The proposed method performance in terms of SNR and payload is compared with existing works and given in Table 3.

Table 2: BER and NC values of the extracted watermark for four classes of audio under signal processing operations

	POP		ROCK		JAZZ		FOLKCOUNTRY	
	BER	NC	BER	NC	BER	NC	BER	NC
Without attack	0	1	0	1	0	1	0	1
Resample	0.2513	0.8582	0.4187	0.7493	0.2704	0.8465	0.3458	0.7986
Requantization	0.0232	0.9877	0.0235	0.9876	0.0236	0.9875	0.0231	0.9878
Noise	0	1	0	1	0	1	0	1
Filter20	0.07	0.9626	0.1907	0.8944	0.0501	0.9733	0.1123	0.9393
croping1	0	1	0	1	0	1	0	1
croping2	0	1	0	1	0	1	0	1
croping3	0	1	0	1	0	1	0	1
mp256	0.1309	0.9288	0.129	0.9299	0.1469	0.9197	0.1337	0.9273
awgn60	0	1	0	1	0	1	0	1
awgn50	0.1153	0.9376	0.1159	0.9372	0.1138	0.9384	0.1145	0.938
signal add	0	1	0	1	0	1	0	1
signal sub	0	1	0	1	0	1	0	1

Table 3: Comparison of Proposed method with state-of-art works in terms of SNR and payload

	Proposed Method	[5]	[4]	[2]	[6]
SNR in dB	39.0550	23.49	20.260	19.918	35-61
Payload in bps	6.5K	172.27	689.06	86	3K

VI. CONCLUSION

DAW plays an important role in copyright protection of audio content. In this paper, slantlet transform based blind DAW scheme is proposed. Pre-processed watermark is embedded into transformed audio using QIM. The performance of the ST DWT is measured in terms of imperceptibility, robustness and payload for four different classes of audio signals. Comparative analysis of the proposed method with existing methods validates its superiority in terms of SNR and payload.

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