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Copyright Protection Using Audio Watermarking

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Abstract

In this paper, digital audio watermarking of an audio signal is performed in frequency domain using DCT. This process exploits the irrelevant properties of the human auditory system for effectively embedding the watermark. We are using a randomly generated sequence/pattern as watermark. The system is also tested for a grayscale image watermark. Relying on exhaustive simulations we test the robustness of the process for additive noise, re-quantization, re-sampling, filtering and cropping. The analysis of the performance reveals that the watermarking scheme maintains high audio quality.

Index Terms**—***audio watermarking, watermarking, Discrete Cosine Transform, copyright protection.*

I. Introduction

Copyright infringement has increased in the past few years due to piracy. Digital Audio Watermarking has received lot of attention in music industry to provide efficient solution for copyright protection of digital media by embedding a digital signature called watermark in the original audio signal [1]-[5],[9]. Major requirement of digital audio watermarking is that the watermarked data must be inaudible to maintain SNR and robust to signal distortion/attacks. Finally, watermark must be easy to embed and extract to prove the ownership copyright.

II. Discrete Cosine Transform (DCT)

Similar to DFT, DCT is also a transformation technique wherein a signal is transformed from spatial to frequency domain. The purpose of DCT is to decompose the signal into high and low frequency components. The common definition of a 1D sequence of length *N* [6] is

$$y(k) = w(k) \sum_{n=0}^{N-1} x(n) cos \frac{\pi(2n-1)(k-1)}{2N} \quad (1)$$

For k=0,1,2,…,(N-1). Similarly, the IDCT is defined as

$$x(n) = \sum_{n=0}^{N-1} w(k) y(k) cos \frac{\pi(2n-1)(k-1)}{2N} \quad (2)$$

For n=0,1,2,…,(N-1). In both the above equations (1) and (2) *w(k)* is defined as

$$w(k) = \begin{cases} \frac{1}{\sqrt{N}}, & \textbf{if } k = 0 \\ \sqrt{\frac{2}{N}}, & \textbf{if } 1 \leq k \leq N-1 \end{cases} \quad (3)$$

The first DCT coefficient gives the average value of the transformed sequence, generally referred to as the DC coefficient and all the other coefficients are referred to as the AC coefficients.

The advantage of DCT is that we need only real valued cosine function. It reduces the computational effort and memory requirement.

III. Quantization Index Modulation (QIM)

This class of embedding process provides the optimal trade-offs for rate-distortion-robustness and is vastly used in watermark embedding. This technique requires the embedding induced distortion to be small and uses a scaling parameter, α to adjust the quantized levels [7]. The value of this parameter determines the amount of distortion. By increasing the value of α, robustness of the watermarking increases with the expense of audio quality and vice-versa. A fraction (1 - α) of the quantization error is fed back to compensate for the distortion. One criterion for choosing α is to maximize the SNR at the quantizer. Typically, α is in the range of $0 \leq \alpha \leq 1$. This criterion is effective especially when the communication channel is additive Gaussian noise channel.

$$\mathbf{s}(x, m) = \mathbf{q}(x; m, \Delta/\alpha) + (1-\alpha)[x - \mathbf{q}(x; m, \Delta/\alpha)] \quad (4)$$

where, q(x;m,Δ/α) is the m-th quantizer whose reconstruction points have been scaled by α so that two reconstruction points are separated by a distance Δ/α after scaling instead of Δ. The first term in (4) represents normal QIM embedding. We refer to the second term as the distortion-compensated QIM term.

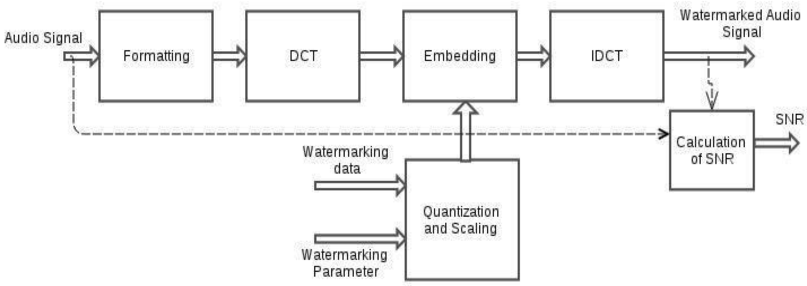

Fig.1: Watermark Embedding Process

IV. WATERMARK EMBEDDING AND EXTRACTION

The algorithm implemented here is based on applying the DCT on the audio signal in which a watermark is to be embedded. The algorithm consists of two parts; watermarking embedding process and watermarking extraction process

A. *Watermark Embedding Process*

The embedding process is as shown in Fig.1. The process is detailed as follows:

1. The input stereo audio signal is converted into a mono audio signal.
2. This signal is then decomposed into basis functions using DCT.
3. The data to be watermarked is quantized and scaled in accordance with watermarking parameter and then embedded into the DCT operated audio signal using QIM as mentioned above in (4) .

   The embedding of watermark is done in low-frequency component of the DCT to obtain robustness against some attacks like low-pass filtering, re-sampling, re-quantization, and additive noise [3]-[4].
4. The watermarked signal is reconstructed using Inverse-DCT and the SNR is computed at this point as an indication for audio quality.

B. *Watermark Extraction Process*

The embedding process is as shown in Fig.1. The process is detailed as follows:

1. The DCT of the attacked watermarked audio signal is computed.
2. During decoding of an audio signal several procedures such as re-sampling, filtering, re-quantization may be performed. This will affect the effective extraction of the watermark. At this stage we compute the SNR to indicate how robust the mechanism is towards these attacks.
3. The watermarked data is extracted by using original audio signal, watermarking parameter and transformed watermarked signal.
4. To verify the quality of the extracted watermarked data, it is compared with the original watermark data. This is done to check the reliability of the algorithm.

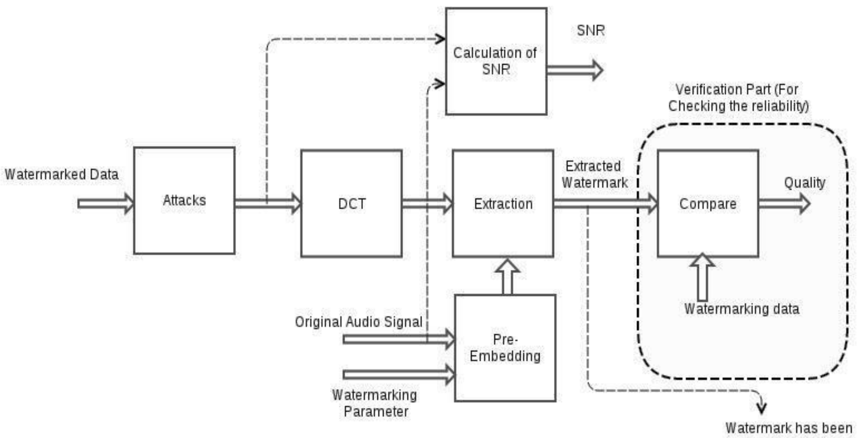

Fig.2: Watermark Extraction Process

V. Simulation Results and Discussion

In this section the performance of the watermarking system is checked for a mono audio signal sampled at 44.1 kHz. Since watermark data can be any image, to make the testing more generalized, the system is tested for a random watermark sequence.

Using subjective listening tests by human acoustic perception or objective evaluation tests by measuring the SNR and ODG, we can perform the perceptual quality assessment. In this work we use the objective evaluation test. According to the norms specified by IFPI the SNR of the signal must be above 20dB [3].

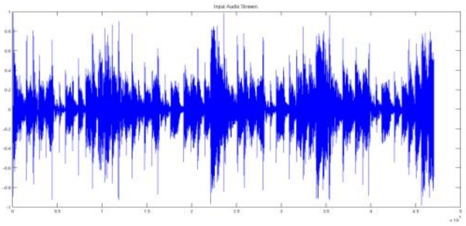

Fig.3a: Original Audio Signal

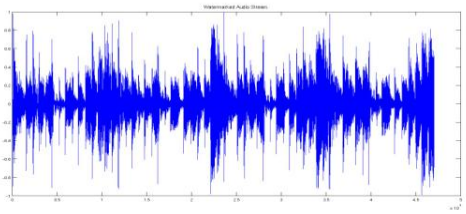

Fig.3b: Watermarked Audio Signal

Figures Fig.3a and Fig.3b shows the original and watermarked audio signal.

A. *Robustness Test*

To assess the robustness, following attacks are performed:

- **Re-sampling** – The watermarked signal which was originally sampled at 44.1 kHz is re-sampled at 22.05 kHz and restored back by sampling again at 44.1 kHz.

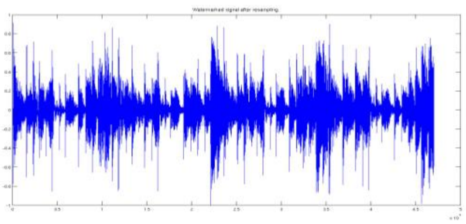

Fig.4a: Watermarked signal after re-sampling

- **Additive noise** – Random noise is added to the watermarked audio signal.

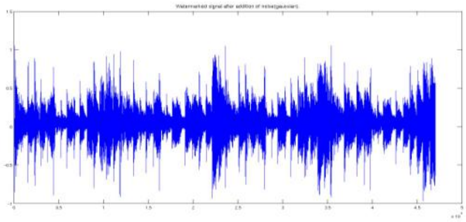

Fig.4b: Watermarked signal after addition of noise

- **Filtering** – Filter the watermarked audio signal using Dirichlet window for 4 kHz cut-off.

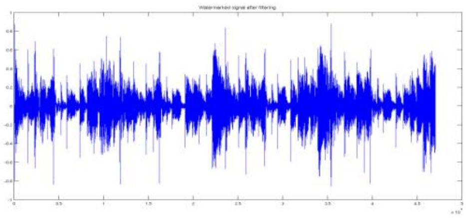

Fig.4c: Watermarked signal after filtering

- **Re-quantization** – The watermarked signal is re-quantized down to 8-bits/sample and then back to 16-bits/sample.

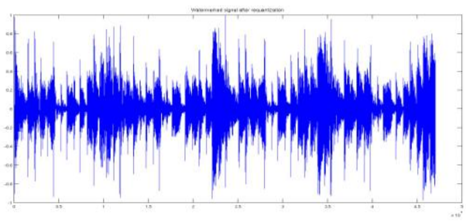

Fig.4d: Watermarked signal after re-quantization

- **Cropping** – 10% segments at the beginning and 20% segments at the end are cropped from the watermarked audio signal.

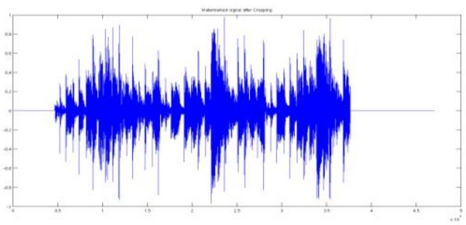

Fig 5: Watermarked signal after cropping

B. Mean Squared Error (MSE)

The mean squared error between the original signal and the watermarked signal is calculated for QIM based watermarking method as follows:

$$MSE = \frac{1}{N}\sum_{n=0}^{N-1}[y(n) - x(n)]^2 \qquad (5)$$

where, *y(n)* is the watermarked signal, *x(n)* is the original signal and *N* is the total number of data points.

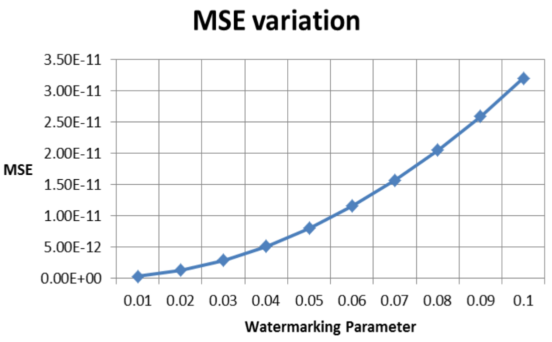

Fig.6: MSE variation

The value of watermarking parameter, α is decided considering the SNR as a major criterion. The MSE is inversely related to SNR. Higher the MSE lower the SNR. From the graph shown in Fig.6, the value ($\alpha \rightarrow 0$) gives best SNR but system will not be robust. The value ($\alpha \rightarrow 1$) gives best robustness but worst SNR performance. Thus, there is a trade-off between SNR and robustness. For optimal performance we have chosen $\alpha = 0.05$.

C. Imperceptibility Test

In order to evaluate the quality of the watermarked audio signal with or without attacks the following equation is used to compute SNR:

$$SNR = 10log\frac{\sum_{n=0}^{N-1}[x(n)]^2}{\sum_{n=0}^{N-1}[x(n) - y(n)]^2} \qquad (6)$$

where, *x(n)* is the original audio signal and *y(n)* can be the watermarked signal with or without attack.

When the watermark is embedded the SNR of the audio signal using this method is above 25dB which is sufficient to prove the imperceptibility of the system which satisfies the IFPI specification as well.

Table 1: Performance analysis

Attack type	SNR (in dB)
No Attack	25.015
Re-sampling	24.816
Additive Noise	18.484
Filtering	19.473
Re-quantization	23.259
Cropping	5.10

VI. EXAMPLE

An image is taken as a watermarking signal instead of a random sequence. We have considered all the attacks mentioned earlier to indicate the effectiveness of the algorithm. The DCT of the audio signal is also shown wherein the frequency content of the audio signal is clearly seen.

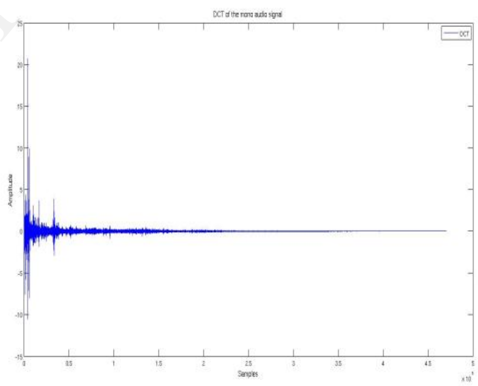

Fig 7: DCT coefficients of the input audio signal

As expected, the first few coefficients contains the low frequency components which contains the major part of the information and the watermark is added in this portion [10]-[12].

Fig.8: Comparison showing watermarking image and extracted watermark image with all attacks performed

It is evident from the Fig.8 that, with all the attacks present the system is able to extract the watermark image effectively. This shows the robustness of the system.

VII. Conclusion and Future Scope

The QIM-based audio watermarking technique implemented in this paper is robust and it can be used effectively for copyright protection. This technique is comparatively less complex than the existing techniques.

This technique can be made more practical by excluding the original audio signal from extraction process. It can be further improved by using complex transformation techniques such as DWT, EMD, etc.

This approach can be extended to image and video watermarking.

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References

[1] J. Cox and M. L. Miller, "The first 50 years of electronic watermarking," J. Appl. Signal Process., vol. 2, pp. 126–132, 2002.

[2] M. D. Swanson, B. Zhu, and A. H. Tewfik, "Robust audio watermarking using perceptual masking," Signal Process., vol. 66, no. 3, pp. 337–355, 1998.

[3] S. Wu, J. Huang, D. Huang, and Y. Q. Shi, "Efficiently self-synchronized audio watermarking for assured audio data transmission," IEEE Trans. Broadcasting, vol. 51, no. 1, pp. 69–76, Mar. 2005.

[4] V. Bhat, K. I. Sengupta, and A. Das, "An adaptive audio watermarking based on the singular value decomposition in the wavelet domain," Digital Signal Process. vol. 2010, no. 20, pp. 1547–1558, 2010.

[5] D. Kiroveski and S. Malvar, "Robust spread-spectrum audio watermarking," in Proc. ICASSP, 2001, pp. 1345–1348.

[6] R. M. Zhao, H. Lian, H. W. Pang, B.N. Hu, "A Watermarking Algorithm by Modifying AC Coefficies in DCT Domain," International Symposium on Information Science and Engieering, pp. 159-162. IEEE, 2008.

[7] B. Chen and G. W. Wornell, "Quantization index modulation methods for digital watermarking and information embedding of multimedia," J. VLSI Signal Process. Syst., vol. 27, pp. 7–33, 2001.

[8] J. X. Liu, Z. M. Lu; J. S. Pan "A Robust Audio Watermarking Algorithm Based on DCT and Vector Quantization," Eighth International Conference on Intelligent Systems Design and Applications, 2008.

[9] M. Arnold, "Audio watermarking: features, applications and algorithms," in Proc. of IEEE International Conference on Multimedia and Expo, vol. 2, pp. 1013-1016, 2000.

[10] Boney, L., Tewfik, A. H., and Hamdy, K. N., "Digital watermarks for audio signal," International Conference on Multimedia Computing and Systems, pp. 473-480, 1996.

[11] I. J. Cox, J. Kilian, F. T. Leighton, and T. Shamoon. "Secure spread spectrum watermarking for multimedia. IEEE Trans. Vol. 6, No. 12, Dec 1997.

[12] P. K. Dhar, M. I. Khan, C. H. Kim, and J. M. Kim, "An Efficient Audio Watermarking Algorithm in Frequency Domain for Copyright Protection, " CCIS, 2010, Vol 122, pp. 104-113, Springer

[13] L. Yaroslavsky, Y. Wang, "DFT, DCT, MDCT, DST and signal Fourier spectrum analysis," 10th European Signal Processing Conference, pp. 1065-1068, Tampere, Finland 2000.

[14] D. Salomon. G. Motta and D. Bryant, "Data Compression: The Complete Reference," Fourth Edition. Springer-Verlag 2007.

[15] Yanzliong Yo and Xinwei Wang. 2002. "A Robust Analysis for. Watermarking Algorithms in DCT." Computer Engineering, Vo1.29. pp 81 -83.