Digital watermarking in A udio signals for covert data Transmission

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Abstract:

With increasing the usage of digital multimedia, the protection of intellectual property rights problem has become a very important issue. Everyday, thousands of multimedia files are being uploaded and downloaded. Therefore, multimedia copyrights become an important issue to protect the intellectual property for the authors of these files. So, this research presents an audio watermarking method that embed a watermark bit in the four LSB layers of audio signal; the algorithm is applied to a stereo audio signal. Embedding watermark in these layers of a sample is equivalent to adding a noise to original signal. The noise level due to embedding watermark is significantly reduced if it is compared with standard LSB watermarking scheme and the quality of watermarked signals is also improved. So, the idea of a proposed algorithm is to obtain a minimal error due to embedding a watermark in an audio signal. Robustness under some attacks is also performed such as resampling and adding noise.

1. Introduction:

Powerful and low cost computers allow people to easily create and copy multimedia content, and the Internet has made it possible to distribute this information at very low cost. However, these enabling technologies also make it easy to illegally copy, modify, and redistribute multimedia data without regard for copyright ownership.

Digital watermarking is seen as a partial solution to the problem of protecting digital media, for it allows content creators to embed sideband data into a host signal, such as author or copyright information. Many techniques have been proposed for watermarking audio, image, and video[1].

Audio watermarking is a method to enforce the intellectual property rights and to protect the audio from tampering which is a technology of embedding hidden information into digital audio signals with minimal effect on the perceptual quality of original host audio signals [1,2,3]. If the watermarked signals are copied, then the hidden information is also carried in the copies. For audio watermarking, the embedded hidden information, called watermark, is usually coded in binary format. One important application of watermarking is to prove the copyright or ownership of watermarked signals. Besides the proof of copyright, audio watermarking also has other applications such as customer tracing, data authentication, and steganography [4,5]. The embedded watermark should be robust and readily extracted from watermarked audio signals even after incidental or intentional attacks, such as noise addition, resampling, filtering or MP3 compression. There are several approaches to embed watermarks in audio signals. For the blind audio watermarking approach, the embedded watermarks can be extracted from watermarked signals without original host signals. For the non-blind audio watermarking approach, the original host signals are required to extract the watermarks [6,7]. In this research, a blind audio watermarking method is proposed and will be explained in details in the next sections.

2. A Digital Representation:

There are two critical parameters to most digital audio representations: sample quantization method and temporal sampling rate. The digital sound is obtained from the analog sound by converting it to digital domain. This process implies two sub processes: sampling and quantization. Sampling is the process in which the analogue values are only captured at regular time intervals. Quantization converts each input value into one of a discrete value. The most popular format for representing samples of high-quality digital audio is a 16-bit linear quantization e.g., Windows Audio-Visual (WAV) and Audio Interchange File Format (AIFF) [8,9,10]. Another popular format for lower quality audio is the logarithmically scaled 8-bit m-law. These quantization methods introduce some signal distortion, somewhat more evident in the case of 8-bit m-law. Popular temporal sampling rates for audio include 8 kHz (kilohertz), 9.6 kHz, 10 kHz, 12 kHz, 16 kHz, 22.05 kHz, and 44.1 kHz. Sampling rate impacts data hiding in that it puts an upper bound on the usable portion of the frequency spectrum (if a signal is sampled at ~8 kHz, it is not desirable to introduce modifications that have frequency components above ~4 kHz). For most data hiding techniques developed, usable data space increases at least linearly with increased sampling rate[8,11,12].

3. Proposed Algorithm:

This research introduces a method to embed a watermark bit in the four LSB layers of audio signal. Then, the impulse noise caused by watermark embedding is shaped in order to change its noise properties. The standard LSB coding method simply replaces the original host audio bit in the ith layer (i=1,..., 16) with the bit from the watermark bit stream. In the case when the original and watermark bit are different and ith LSB layer is used for embedding the error caused by watermarking is 2^{i-1}. So, for example, when i=4 that implies the error caused by watermarking equals to 8. The embedding error is positive if the original bit was 0 and watermark bit is 1 and vice versa. The idea of the proposed LSB algorithm is watermark bit embedding that causes minimal embedding distortion of the host audio. It is clear that, if only one of 16 bits in a sample is fixed and equalled to the watermark bit, the other bits can be flipped in order to reduce the embedding error of watermarking.

Proposed Algorithm:

However, in the embedding algorithm, the (i+1)th layer is first modified by insertion the present message bit. Where:

m: is a bit of the watermark,
x: is the bit layer of original signal,
& i= 0……15.

The following steps illustrate how the algorithm is work:

If (m equal to x_i) → no change is happened.
If (m = 0 & x_i =1) → $x_i = 0$
if ($x_{i-1} = 0$) → $x_{i-1}\, x_{i-2} \ldots x_0$ = ones
if ($x_{i-1} = 1$) → $x_{i-1}\, x_{i-2} \ldots x_0$ = zeros &
if ($x_{i+1} = 0$) → $x_{i+1} = 1$
else if ($x_{i+2} = 0$) → $x_{i+2} = 1$
…
else if ($x_{15} = 0$) → $x_{15} = 1$
Else If (m=1 & $x_i = 0$) → $x_i = 1$
if ($x_{i-1} = 1$) → $x_{i-1}x_{i-2} \ldots x_0$ = zeros
if ($x_{i-1} = 0$) → $x_{i-1}x_{i-2} \ldots x_0$ = ones &
if ($x_{i+1} = 1$) → $x_{i+1} = 0$
else if ($x_{i+2} = 1$) → $x_{i+2} = 0$
…
else if ($x_{15} = 1$) → $x_{15} = 0$

For example:

(1) If the original sample value is: 0001**0**111 = 23
and the watermark bit is **1** is to be embedded into 4th layer
The sample will be: 0001**1**111 = 31
after flipping the error, the sample will be: 0001**1**000 =24
which is more closer to the original one.
(2) If the original sample value is: 0000**1**111 = 15
And the watermark bit is **0** is to be embedded into 4th layer
The sample will be: 0000**0**111 = 7
After flipping the error, the sample will be: 0001**0**000 =16
which is more closer to the original one.

4. Experimental Results:

The algorithm is applied to a stereo audio signal and embedding watermark in the four bit layers of the sample. This is equivalent to adding noise to original signal. For measuring the quality of audio signal, we performed the peak signal-to-noise ratio (PSNR) to the watermarked audio signal due to embedding watermark bits in different bit layers. The proposed watermarking scheme is applied to stereo music signal of 16 bits per sample. A watermark bit stream is embedded in 1^{st} layer to 4^{th} layer of the samples of stereo signals. The mono signal watermarking algorithm is also applied to one single-channel of the music signal. The same watermark bit stream is embedded in 1^{st} layer to 4^{th} layer of each mono signal samples and generate 4 different watermarked mono signals. The peak signal-to-noise ratio (PSNR) of watermarked signals due to embedding watermark bits is calculated according to equation (1) as follows [13]:

$$PSNR = 10 \log_{10} (2^{L-1})^2 / MSE \ \ldots\ldots.(1)$$

Where L represents the number of bits per sample, The PSNRs of stereo & mono watermarked signals are shown in Table (1), while figure (1) states the difference between the mono & stereo watermarked signals as curves.

Table.1 PSNR of embedding a watermark in audio signals

Watermarked Bit Layer	**PSNR of watermarked signals**	
	Mono	**Stereo**
Lay.1	64.45	73.53
Lay.2	63.13	69.98
Lay.3	58.79	65.27
Lay.4	41.91	60.89

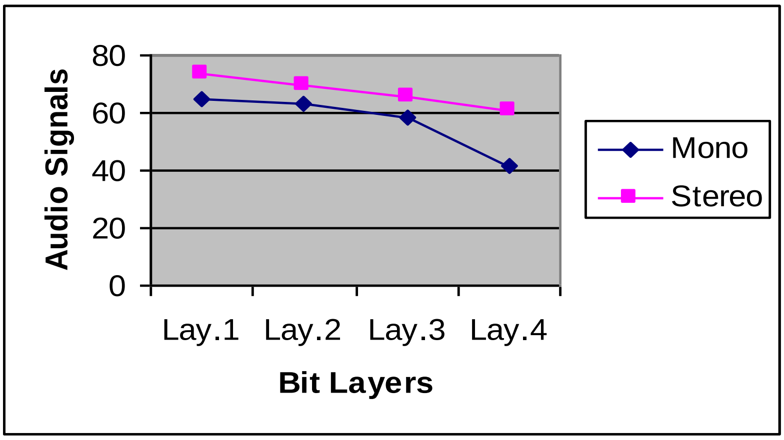

Figure.1 PSNR of mono & stereo watermarked audio signals

When the proposed algorithm is used, additional noise level due to embedding watermark is significantly reduced if it is compared with standard LSB watermarking scheme and the quality of watermarked signals is also improved. It can be noticed that the values of PSNR of each embedded bit layer for the proposed watermarking method is higher than that for mono signal watermarking method. For mono signal watermarking method, the PSNR of 1st embedded bit layer is 64.45 dB, while the PSNR for stereo signal of 1st embedded bit layer is 73.53, and the PSNR of 2nd embedded bit layer is 63.13 dB, while the PSNR for stereo signal of 2nd embedded bit layer is 69.98 and so on. Which means the rate of the error in stereo becomes little than in mono signal and the noise is not noticeable in stereo likes in mono.

To test the robustness of the algorithm, some attacks are performed as follows:

1. Resampling: the proposed audio watermarking method was tested the robustness under attack of resampling. The watermarked stereo signals are first resampled at two different frequencies and then resampled back at 44.1 kHz which is the original sampling frequency. After resampling, the length of audio signal and the number of samples are not changed. Table.2 & figure.2 show the exaction rates under attack of resampling:

Table.2 Extraction Rate under attack of resampling

Watermarked Bit Layers	Resampling Frequency	
	22.05 kHz	88.2 kHz
Lay.1	48.59 %	52.12 %
Lay.2	49.98 %	52.99 %
Lay.3	51.09 %	60.73 %
Lay.4	52.78 %	63.67 %

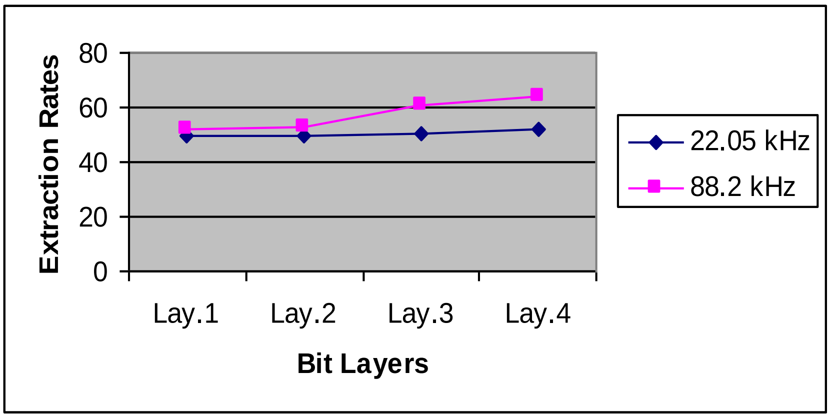

Figure.2 Extraction Rate of watermark bits

When resampling frequency is 22.05 kHz the extraction rates are about 50% while, when resampling frequency is 88.2 kHz, the extraction rates increased to reach above of 60% in 3rd & 4th bit layers. So, it can be said that the proposed method has more robustness when sampling frequency is an integer multiple of original sampling frequency.

2. Adding the noise: to test the robustness under attack of noise, it was added a

certain magnitude of noise to 4 watermarked stereo and mono signals, and obtain watermarked signals with resulting PSNRs of 40dB each of them. The stream of watermark bits can be extracted by reading the bits in the embedded bit layers. The percentages of correct extracted bits are calculated for each of these watermarked signals. Table.3 & figure.3 stated extraction rates of watermark bits from noised signals.

Table.3 Extraction Rate under attack of noise

Bit Layer with 40dB PSNR of watermarked signals	**Extraction Rate of watermark**	
	Mono	**Stereo**
Lay.1	49.45 %	51.28 %
Lay.2	60.91 %	67.74 %
Lay.3	64.84 %	73.54 %
Lay.4	70.96 %	82.08 %

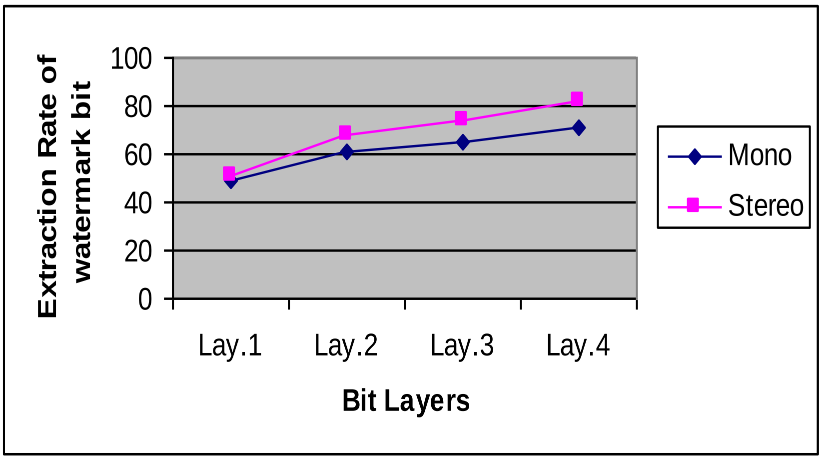

Figure.3 Extraction Rate of watermark bits

When adding a noise with resulting PSNRs of watermarked signals are 40dB, the extraction rate for proposed stereo signal algorithm is higher than for mono signal watermarking algorithm. So, the extraction rates are increased through the 4^{th} bit layers. For example, the rate of extraction for 4^{th} bit layer in stereo is 82.08% higher than the extraction rate 70.96 % for mono watermarked signals.

5. Conclusions:

In this research, an audio watermarking algorithm is proposed which embeds watermark bits in stereo signals. When it is used, additional noise level due to embedding watermark is significantly reduced if it is compared with standard LSB watermarking scheme and the quality of watermarked signals is also improved which can be noticed that through the values of PSNR of each embedded bit layer for the proposed watermarking method is higher than that for mono signal watermarking method. Also, some attacks are performed to test the robustness of the system likes resampling frequency, which found when resampling frequency with 22.05 kHz, the extraction rates are about 50% while, when resampling frequency is 88.2 kHz, the extraction rates increased to reach above of 60% in 3rd & 4th bit layers that means the algorithm is more robustness when the sampling frequency is higher than it is used. Adding noise is another attack is performed which found that the algorithm is robust with this level of noise, a higher level of noise may destroy the watermarked signals so the extraction rate is increased through the four bit layers that implies to the proposed stereo signal algorithm is higher than for mono signal .

References

[1] Harsh Verma, Ramanpreet Kaur and Raman kumar, " Random Sample Audio Watermarking Algorithm for Compressed Wave Files" , IJCSNS International Journal of Computer Science and Network Security, VOL.9 No.11, November 2009.

[2] S.W. Foo, H.S. Muh, and N.M. Mei "Audio watermarking using time frequency compression expansion" , IEEE International Symposium on Circuits and Systems, 2004.

[3] N. Cvejic and T. Seppänen, " Audio watermarking: requirement, algorithms, and benchmarking ". Digital watermarking for digital media, Information Science Publishing, Pennsylvania, 2005.

[4] J. Foote, J. Adco, and A. Girgensohn, " Time base modulation: a new approach to watermarking audio " [Electronic Version], Palo Alto Laboratory. California. 2003.

[5] M.A. Suhail, " Digital watermarking for protection of intellectual property ". Multimedia security: steganography and digital watermarking techniques for protection of intellectual property, Idea Group Publishing. Pennsylvania, 2006.

[6] P. Bassia, W.T. Pitas, " Robust Audio Watermarking In Time Domain", Patras, Greece, pp 25-28, EUSIPCO 1998.

[7] H. zer, B.Sankur, and N.Memon, " An SVD-Based Audio Watermarking Technique ", in proceedings of the 7th workshop on Multimedia and security '05, ACM Press, August 2006.

[8] Pradeep Kumar Singh et. al., " Enhancement of LSB based Steganography for Hiding Image in Audio " , IJCSE International Journal on Computer Science and Engineering, Vol. 02, No. 05, 2010.

[9] M. L. Mat Kiah, B. B. Zaidan, A. Zaidan, A. Mohammed Ahmed and Sameer Hasan Al-bakri, " A review of audio based steganography and digital Watermarking" , International Journal of the Physical Sciences Vol. 6(16), August, 2011.

[10] Alsalami MAT, Al-Akaidi MM. " Digital Audio Watermarking: Survey ", 17th European Simulation Multi-conference, UK, 2003.

[11] Hmood AK, Jalab HA, Kasirun ZM, Zaidan BB, Zaidan A. , "On the Capacity and Security of Steganography Approaches: An Overview ",J. Appl. Sci., 10(16), 2010.

[12] Lin Y, Abdulla WH, " Perceptual evaluation of audio watermarking using objective quality measures", IEEE Int. Conf. Acoust. Speech Signal Process, 2008.

[13] Kiryung Lee, Dong Sik Kim, Taejeong Kim and Kyung Ae Moon, "EM Estimation of

Scale Factor for Quantization-Based Audio Watermarking ’’, Lecture Notes in Computer Science, 2004.

العلامة المائية الرقمية في ملفات الصوت لنقل البيانات

الخلاصه:

أصبحت حماية حقوق الملكيه الفكريه ضروره ملحه مع زيادة أستخدام الوسائط المتعدده الرقميه ، حيث هنالك في كل يوم الالاف من ملفات الوسائط المتعدده يتم تحميلها من والى شبكة الانترنت. لذلك أصبح من المهم حماية حقوق الملكيه الفكرية لأصحاب تلك الملفات.

يقدم البحث طريقه لأخفاء العلامة المائيه في ملفات الصوت في أدنى أربع بتات للأشاره الصوتيه من نوع (Stereo). أن هذه العمليه تكافئ أضافة ضوضاء(noise) للأشاره الاصليه حيث أن الضوضاء الناتجه من هذه العمليه سوف تقل بالتأكيد أذا ما تم مقارنتها بطريقة البت الأدنى (LSB) الأصليه لذلك فأن هدف هذه الطريقه هو الحصول على أقل نسبة خطأ ناتجه من أخفاء الأشاره المائيه (watermark) في الأشاره الصوتيه. لقد تم أختبار الموثوقيه (Robustness) للخوارزميه المقترحه من خلال تطبيق بعض الهجمات (Attacks) مثل أعادة التقطيع (Resampling) وأضافة الضوضاء (Adding noise).