INCREASED ROBUSTNESS OF LSB AUDIO STEGANOGRAPHY BY REDUCED DISTORTION LSB CODING

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ABSTRACT

In this paper, we present a novel high bit rate LSB audio watermarking method that reduces embedding distortion of the host audio. Using the proposed two-step algorithm, watermark bits are embedded into higher LSB layers, resulting in increased robustness against noise addition. In addition, listening tests showed that perceptual quality of watermarked audio is higher in the case of the proposed method than in the standard LSB method. Conventionally, a perceptual limit of three bits per sample is imposed to the basic LSB audio steganography method. The proposed algorithm makes use of minimum error replacement method for LSB adjustment and modified error diffusion method for decreasing SNR value. Subjective listening test proved that high perceptual transparency is accomplished even if four LSBs of host audio signal are used for data hiding. Using the proposed two-step algorithm, hidden bits are embedded into the higher LSB layers, resulting in increased robustness against noise addition. In addition, listening tests showed that perceptual quality of watermarked audio is higher in the case of rite proposed method than the perceptual quality obtained using the standard LSB method. We present a novel high bit rate LSB audio watermarking method. The basic idea of the proposed LSB algorithm is watermark embedding that causes minimal embedding distortion of the host audio.

Keywords: *Audio steganography, LSB coding, data hiding Category: D.4.6, H.5.1, Flipping, Robustness.*

1. INTRODUCTION

Multimedia data hiding techniques have developed a strong basis for steganography area with a growing number of applications like digital rights management, covert communications, hiding executables for access control, annotation etc. In all application scenarios given above, multimedia steganography techniques have to satisfy two basic requirements. The first requirement is perceptual transparency, i.e. cover object (object not containing any additional data) and stego object (object containing secret message) must be perceptually indiscernible . The second constraint is high data rate of the embedded data. All the stego applications, besides requiring a high bit rate of the embedded data, have need of algorithms that detect and decode hidden bits without access to the original multimedia sequence (blind detection algorithm). While the robustness against intentional attack is not required, a certain level of robustness of hidden data against common signal processing as noise addition or MPEG compression may be necessary. LSB coding is one of the earliest techniques studied in the information hiding and watermarking area of digital audio (as well as other media types). The main advantage of the LSB coding method is a very high watermark channel bit rate and a low computational complexity of the algorithm, while the main disadvantage is considerably low robustness against signal processing modifications. The perceptual quality of watermarked audio is higher in the case of the proposed method than in the standard LSB method is implementing in the Electronic Copyright Management System.

2. STANDARD LSB METHOD

Data hiding in the least significant bits (LSBs) of audio samples in the time domain is one of the simplest algorithms with very high data rate of

additional information. The LSB watermark encoder usually selects a subset of all available host audio samples chosen by a secret key. The substitution operation on the LSBs is performed on this subset, where the bits to be hidden substitute the original bit values. Extraction process simply retrieves the watermark by reading the value of these bits from the audio stego object. Therefore, the decoder needs all the samples of the stego audio that were used during the embedding process. The random selection of the samples used for embedding introduces low power additive white Gaussian noise (AWGN). It is well known from the psychoacoustics literature that the human auditory system (HAS) is highly sensitive to AWGN. That fact limits the number of LSBs that can be imperceptibly modified during watermark embedding. The main advantage of the LSB coding method is a very high watermark channel bit rate; use of only one LSB of the host audio sample gives capacity of 44.1 kbps (sampling rate 44 kHz, all samples used for data hiding) and a low computational complexity. The obvious disadvantage is considerably low robustness, due to fact that simple random changes of the LSBs destroy the coded watermark. As the number of used LSBs during LSB coding increases or, equivalently, depth of the modified LSB layer becomes larger, probability of making the embedded message statistically detectable increases and perceptual transparency of stego objects is decreased. Therefore, there is a limit for the depth of the used LSB layer in each sample of host audio that can be used for data hiding. Subjective listening test showed that, in average, the maximum LSB depth that can be used for LSB based watermarking without causing noticeable perceptual distortion is the fourth LSB layer when 16 bits per sample audio sequences are used. The tests were performed with a large collection of audio samples and individuals with different background and musical experience. None of the tested audio sequences had perceptual artifacts when the fourth LSB has been used for data hiding, although in certain music styles, the limit is even higher than the fourth LSB layer. Robustness of the watermark, embedded using the LSB coding method, increases with increase of the LSB depth used for data hiding. Therefore, improvement of watermark robustness obtained by increase of depth of the used LSB layer is limited by perceptual transparency bound, which is the fourth LSB layer for the standard LSB coding algorithm.

3. PROPOSED LSB METHOD

We developed a novel method that is able to shift the limit for transparent data hiding in audio from the fourth LSB layer to the sixth LSB layer, using a two-step approach. In the first step, a watermark bit is embedded into the ith LSB layer of the host audio using a novel LSB coding method. In the second step, the impulse noise caused by watermark embedding is shaped in order to change its white 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 i^{th} LSB layer is used for embedding the error caused by watermarking is 2^i-1 quantization steps (QS) (amplitude range is [-32768 to 32767]). The embedding error is positive if the original bit was 0 and watermark bit is 1 and vice versa. The key 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 equal to the watermark bit, the other bits can be flipped in order to minimize the embedding error. For example, if the original sample value was $0...01000_2=8_{10}$, and the watermark bit is zero is to be embedded into 4th LSB layer, instead of value $0...00000_2=0_{10}$, that would the standard algorithm produce, the proposed algorithm produces sample that has value $0...00111_2=7_{10}$, which is far more closer to the original one. However, the extraction algorithm remains the same, it simply retrieves the watermark bit by reading the bit value from the predefined LSB layer in the watermarked audio sample. In the embedding algorithm, the $(i+1)^{th}$ LSB layer (bit ai) is first modified by insertion of the present message bit. Then, the algorithm given below is run. In case that the bit ai need not be modified at all due to being already at a correct value, no action is taken with that signal sample. Underlined bits (*ai*) represent bits of watermarked audio.

Algorithm:

if host sample a>0
      if bit 0 is to be embedded
      if a_{i-1} =0 then $\underline{a}_{i-1}$ $\underline{a}_{i-2}$ …. $\underline{a}_0$ = 11…1
      if a_{i-1} =1 then $\underline{a}_{i-1}$ $\underline{a}_{i-2}$ …. $\underline{a}_0$ = 00…0 and
         if a_{i+1}=0 then $\underline{a}_{i+1}$ = 1
           else if a_{i+2}=0 then $\underline{a}_{i+2}$=1
           …..
           else if a_{15} =0 then $\underline{a}_{15}$ = 1
      else if bit 1 is to be embedded

if a_{i-1} =1 then $\underline{a}_{i-1}$ $\underline{a}_{i-2}$ …. $\underline{a}_0$ = 00…0
if a_{i-1} =0 then $\underline{a}_{i-1}$ $\underline{a}_{i-2}$ …. $\underline{a}_0$ = 11…1 and
   if a_{i+1}=1 then $\underline{a}_{i+1}$ = 0
     else if a_{i+2}=1 then $\underline{a}_{i+2}$=0
     …..
     else if a_{15} =1 then $\underline{a}_{15}$ = 0
else if bit 1 is to be embedded

if host sample a<0
   if bit 0 is to be embedded
   if a_{i-1} =0 then $\underline{a}_{i-1}$ $\underline{a}_{i-2}$ …. $\underline{a}_0$ = 11…1
   if a_{i-1} =1 then $\underline{a}_{i-1}$ $\underline{a}_{i-2}$ …. $\underline{a}_0$ = 00…0 and
     if $\underline{a}_{i+1}$=0 then $\underline{a}_{i+1}$ = 1
       else if a_{i+2}=0 then $\underline{a}_{i+2}$=1
       …..
       else if a_{15} =0 then $\underline{a}_{15}$ = 1

     else if bit 1 is to be embedded
if a_{i-1} =1 then $\underline{a}_{i-1}$ $\underline{a}_{i-2}$ …. $\underline{a}_0$ = 00…0
if a_{i-1} =0 then $\underline{a}_{i-1}$ $\underline{a}_{i-2}$ …. $\underline{a}_0$ = 11…1 and
   if a_{i+1}=1 then $\underline{a}_{i+1}$ = 0
     else if a_{i+2}=1 then $\underline{a}_{i+2}$=0
     …..
     else if a_{15} =1 then $\underline{a}_{15}$ = 0
else if bit 1 is to be embedded

The embedding characteristic of the proposed LSB coding algorithm is given in the Figure 1, for the case when watermark bit is equal to zero, and in Figure 2, for the case when the watermark bit equals one. Figures depict an example of the embedding characteristics where the 4th LSB layer is used for watermarking; the values obtained by the proposed LSB method are represented as the line. It is clear that the proposed method introduces smaller error during watermark embedding. If the 4th LSB layer is used, the absolute error value ranges from 1 to 4 QS, while the standard method (dash-dot line) in the same conditions causes constant absolute error of 8 QS. The average power of introduced noise is therefore 9.31 dB smaller if the proposed LSB coding method is used. In addition to decreasing objective quality measure, expressed as signal to noise ratio (SNR) value, proposed method introduces, in the second step of embedding, noise shaping in order to increase perceptual transparency of the method. A similar concept, called error diffusion method is commonly used in conversion of true color images to palette-based color images. In our algorithm, embedding error is spread to the four consecutive samples, as samples that are predecessors of the current sample cannot be altered because information bits have already been embedded into their LSBs. Let e(n)

denote the embedding error of the sample a(n). For the case of embedding into the 4th LSB layer, the next four consecutive samples of the host audio are modified according to these expressions:

$$a(n+1)=a(n+1)+[e(n)] \quad a(n+3)=a(n+3)+[e(n)]/3$$
$$a(n+2)=a(n+1)+[e(n)]/2 \quad a(n+4)=a(n+4)+[e(n)]/4$$

where $\lfloor A \rfloor$ denotes floor operation that rounds A to the nearest integer less than or equal to A.
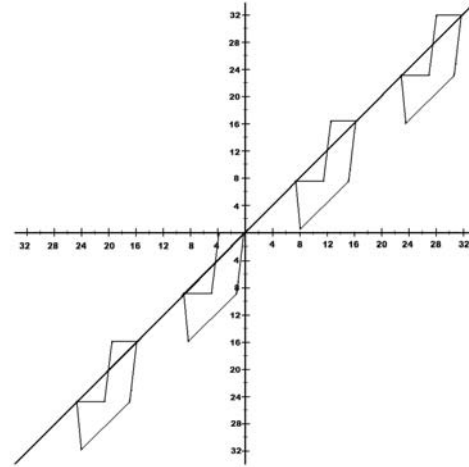

Figure 1. Embedding characteristics of the standard vs. proposed LSB coding algorithms (bit 0 embedded).
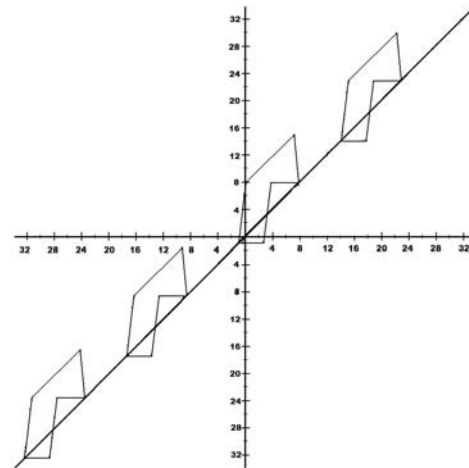

Figure 2. Embedding characteristics of the standard vs. proposed LSB coding algorithms (bit 1 embedded).

Error diffusion shapes input impulse noise, introduced by LSB embedding, by smearing it and changing its distribution to a perceptually better tuned one. Effect is most emphasized during silent

periods of audio signal and in fragments with low dynamics e.g. broad minimums or maximums. The both embedding steps jointly increase the subjective quality of audio stego object. Therefore, we expect that, using the proposed two-step algorithm, we can increase the depth of watermark embedding further than the 4th LSB layer and accordingly increase algorithm's robustness towards noise addition.

4. EXPERIMENTAL RESULTS

Proposed LSB watermarking algorithm was tested on 10 audio sequences from different music styles (pop, rock, techno, jazz). The audio excerpts were selected so that they represent a broad range of music genres, i.e. audio clips with different dynamic and spectral characteristics. All music pieces have been watermarked using the proposed and standard LSB watermarking algorithm. Clips were 44.1 kHz sampled mono audio files, represented by 16 bits per sample. Duration of the samples ranged from 10 to 15 seconds. As defined in, signal to noise ratio for the embedded watermark is computed as:

$$SNR = 10.\log10\{\Sigma_n\, x^2\,(n)\,/\,\Sigma_n\,[x^2(n) - y^2\,(n)]\}$$

where x(n) represents a sample of input audio sequence and y(n) stands for a sample of audio with modified LSBs. SNR values for the standard method (embedding performed in the 4th LSB layer) and the proposed method (embedding performed in the 4th, 5th and 6th LSB layer) are given in Figure 3. It can be seen from the Figure 3, that the novel algorithm outperforms standard LSB insertion algorithm. Two methods obtain similar SNR values when the embedding is done in the 6th LSB layer using the proposed method and in the 4th LSB layer in the case of the standard method. Subjective quality evaluation of the watermarking method was performed by listening tests involving ten persons. Three of them had basic or medium level music education or are active musicians
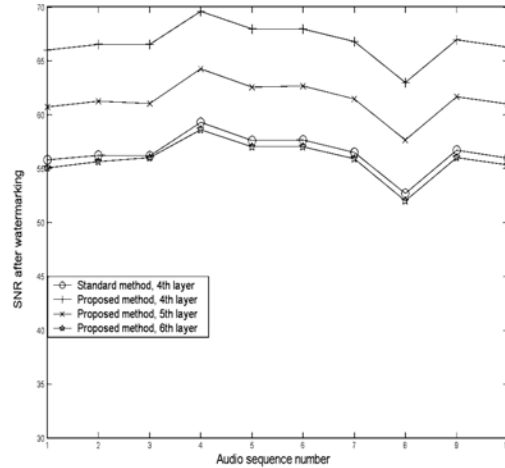

Figure 3. SNR values of 10 test audio sequences for standard and proposed LSB watermarking

In the first part of the test, participants listened to the original and the watermarked audio sequences and were asked to report dissimilarities between the two signals, using a 5-point impairment scale: (5: imperceptible, 4: perceptible but not annoying, 3: slightly annoying, 2: annoying and 1: very annoying). Table 1 presents results of the first test, with the average mean opinion score (MOS) for three of the 10 tested audio excerpts. In the second part, test participants were repeatedly presented with unwatermarked and watermarked audio clips in random order and were asked to determine which one is the watermarked one (blind audio watermarking test).

Experimental results are presented also in Table 1. Values near to 50% show that the two audio clips (original audio sequence and watermarked audio signal) cannot be discriminated by people that participated in the listening tests. Results of subjective tests showed that perceptual quality of watermarked audio, if embedding is done using the novel algorithm, is higher in comparison to standard LSB embedding method.

	Country	Violin	Pop
Discrimination values (%)			
Standard method (4th LSB)	52	49	51
Standard method (5th LSB)	59	40	57
New method (6th LSB)	51	50	51
New method (7th LSB)	55	45	55
Mean opinion score (MOS)			
Standard method (4th LSB)	5.0	4.9	5.0
Standard method (5th LSB)	4.6	4.5	4.7
New method (6th LSB)	5.0	5.0	5.0
New method (7th LSB)	4.6	4.6	4.6

Table 1. Mean opinion scores and discrimination values

Discrimination values and mean opinion scores in the case of proposed algorithm embedding in the 6th LSB layer are practically the same as in the case of the standard algorithm embedding in the 4th LSB layer. This confirms that described algorithm succeeds in increasing the depth of the embedding layer from 4th to 6th LSB layer without affecting the perceptual transparency of the watermarked audio signal. Therefore, a significant improvement in robustness against signal processing manipulation can be obtained, as the hidden bits can be embedded two LSB layers deeper than in the standard LSB method. In order to compare the robustness of the proposed algorithm and the standard one, additive white Gaussian noise was added to the samples of watermarked audio and bit error rate (BER) measured. The values given in Figure 4 are for 44.1 kbps embedding rate and calculated as number of flipped hidden bits over the total number of received bits. The improvement in robustness against additive noise is obvious, as the proposed algorithm obtains significantly lower bit error rates than the standard algorithm if the same noise variance is added to the watermarked audio. As described above, the proposed algorithm flips bits in more than one bit layers of the watermarked audio during the embedding procedure. This property may increase the resistance against steganalysis that identifies the used LSB layer by analyzing the noise properties of each bit layer. Figure 6 and 7 show histogram of the number of modified bit layers in a 1.5 sec audio sample (66150 bits in total) for the standard and proposed LSB algorithm, respectively.
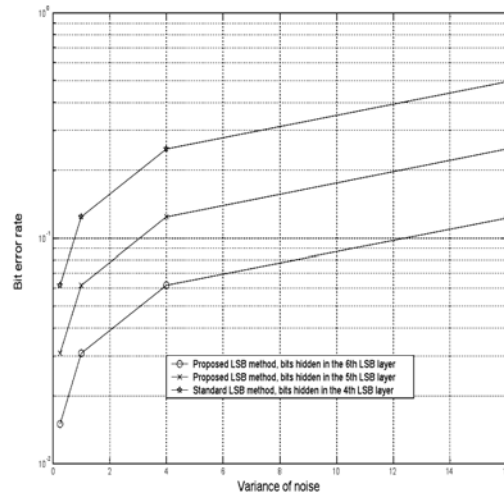

Figure 4. Robustness of the algorithms in presence of additive white Gaussian noise audio during the embedding procedure.

This property may increase the resistance against steganalysis that identifies the used LSB layer by analyzing the noise properties of each bit layer. Figures 5 and 6 show histogram of the number of modified bit layers in a 1.5 sec audio sample (66150 bits in total) for the standard and proposed LSB algorithm, respectively.
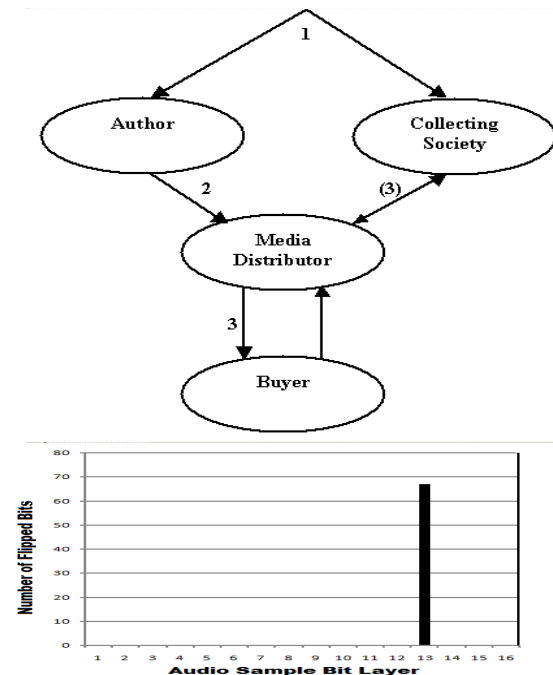

Figure 5. Number of flipped bits per bit layer for the standard LSB algorithm (data hiding is done in 4th LSB layer)

It is clear that number of flipped bits per bit layers is distributed over all bit layers in the proposed algorithm, while the standard algorithm flips bits only in one bit layer. In the case of standard LSB algorithm, LSB steganography techniques can easily detect the bit layer where the data hiding was performed. It is a much more challenging task in the case of the proposed algorithm, because there is a significant number of bits flipped in seven bit layers and the adversary cannot identify exactly which bit layer is used for the data hiding.
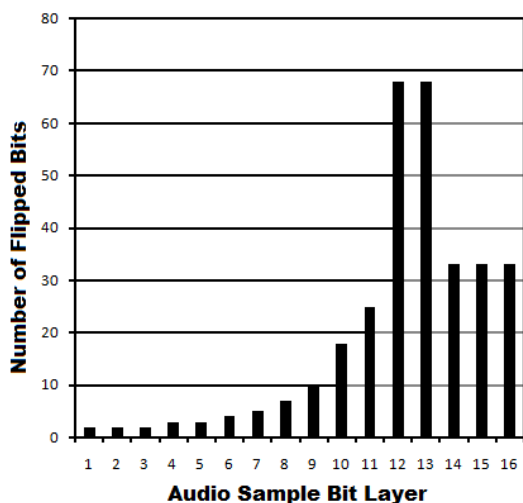

Figure 6. Number of flipped bits per bit layer for the proposed algorithm (data hiding is done in 4th LSB layer)

5. IMPLEMENTATION DETAILS

This project consists of implementing the Electronic Copyright Management System. In ECMS there are four modules.

Fig 7. Implementing the Electronic Copyright Management System.

In Author Module Creation Unique Number is embedded into the Image using author private key. In the embedding of CUN it uses asymmetric watermarking algorithm. Distributor PIN is also embedded into the image using private key Asymmetric encryption algorithm.

Collection Society is the trusted third party that will ensure that the protected document traded correctly. It involves transaction between buyer & media distributor.

In collection Society module, Buyer PIN is embedded into the image using CS private key in Asymmetric encryption. It also computes Hash value of the image which should be sent to buyer. It is used as authentication purpose. This hash value is also appended into the image and the encrypted image is transferred to the buyer using LAN or Email networks.

In Buyer module, Buyer decrypts the encrypted digest using CS public key and the digest value is computed. Hash value is recomputed from the decrypted digest and the hash value is compared. If these values are same then it ensures no transmission loss. From third encrypted watermark buyer decrypt the Buyer PIN from it and ensures it legal ownership.

Control Authority is used for Illegal usage detection phase. It compares CUN with buyer watermark and distributor watermark and detect the legal or illegal ownership.

6. CONCLUSION

We presented a reduced distortion bit-modification algorithm for LSB audio steganography. The key idea of the algorithm is watermark bit embedding that causes minimal embedding distortion of the host audio. Listening tests showed that described algorithm succeeds in increasing the depth of the embedding layer from 4th to 6th LSB layer without affecting the perceptual transparency of the watermarked audio signal. The improvement in robustness in presence of additive noise is obvious, as the proposed algorithm obtains significantly lower bit error rates than the standard algorithm. The steganalysis of the proposed algorithm is more challenging as well, because there is a significant number of bits flipped in a number in bit layers and the adversary cannot identify exactly which bit layer is used for the data hiding.

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