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Data Embedding in Audio Signal by a Novel Bit Marking Method

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Abstract

Embedding data in audio signals is a technique to hide a certain amount of information in the digital audio signal to preserve confidentiality and authenticate information. The bit replacing methods are now mainly used in the techniques of data embedding in the audio signal. Although the capacity of data embedding is large, but embedding error (EE) is high, thus reducing the possibility of transparence of embedded signal. In this paper, we propose a novel method of bit marking (BM) to embed data into the audio signals. To reduce the embedding error and increase signal- to-noise ratio (SNR), we first conducted comparing each segment of data to the bit segments in the audio samples, then find out the bits of audio signals which coincide with embedded data. After that, we use the low-significant bit of audio samples to mark the positions of the bit segments found. The number of marking bits is reduced by the minimization algorithm, and optimization of the parameters of the system. The experiment results show that the quality of the proposed system is better than the least significant bit (LSB) and the embedding large sample (ELS) methods. Therefore, the security of the embedded data is also significantly improved.

Keywords: *Bit Marking (BM), Embedding Error (EE), Embedded signal, Least Significant Bit (LSB), Embedding Large Sample (ELS).*

1. Introduction

The social development has increased the demand for information exchange. Information flowing on transmission systems is diverse, in which the audio signal is one of the most popular. A practical requirements set out today is, in many cases, the audio signal transmission needs associated with copyright to justify when disputes arise. However, the data used to determine the copyright must be hidden in the audio signal, do not change the format, size and hardly affect the quality of the audio signal.

Embedded signal is often used to solve the problems mentioned above. In addition, the embedded signal is also used to solve confidential communication problems applied in the field of security and defense. The secret message embedded in the audio signal will be sent from the transmitter to the receiver by the available way of transmission. The receiver will receive the secret message by using a special de-embedding algorithm agreed with the transmitter.

Audio signals often have a high sampling frequency (from 8 kHz to 44 kHz), so the amount of embedded data can be very large [2][11]. The sound characteristics like: amplitude, frequency, phase, echo effects can be exploited to perform the embedding techniques [3][6][8][10][12]. However, due to the sensitivity of the human auditory system, the embedding data in the audio signal is easily detected [2][9]. The biggest challenge posed to the data embedding techniques in audio signal is reducing EE but increasing SNR [1][5][7][8].

Due to the advantages of the embedded signal and from the characteristics of audio signals, there are many researches to bring out the embedding method to solve the above-mentioned challenges such as: (i) LSB coding [4][8][14]; (ii) Parity coding [8][14]; (iii) Phase coding [4][8][9][14]; (iv) embedding techniques based on Echo [8][14]; (v) spreading spectrum techniques [2][14]. In [13], we have proposed ELS method, with which the data is divided into small segments, and then embedded into large amplitude audio samples. Experiment results show that the method has significantly improved the quality of the system.

However, according to our survey, although the above-mentioned methods have analyzed and assessed the impact of the embedded data on the audio signal, most of the work has focused on the techniques substituting data bits to the position of the least significant bit of the audio signal. Therefore, the analyzing and finding out embedding methods which limit the use of bit substituting techniques is an open practical topic today.

From the practical and scientific significance above, in this paper, we analyze and propose the BM method to implement embedding data into audio signals. The paper has the following contributions:

- Propose BM method used $\mho$ [bit] to mark the Ω [bit] embedded data with $\mho < \Omega$;
- Propose algorithm minimizing the redundant bits (RB), the algorithm minimizing the marking bits (MB) and the SNR optimization algorithm. Test results of the proposed algorithms show that the quality of the embedded signals is better than the ones using the LSB embedding method [4][8][14] and ELS method [13].

The rest of the paper is organized as follows: The survey system model is presented in Section 2. Section 3 presents the algorithms to optimize the parameters of the model. Experimenting and analyzing results are clarified in Section 4. Summary, conclusions and recommendations for further research are presented in Section 5.

We agreed to use some symbols in this paper as follows: lowercase and italics perform variables; uppercase performs audio signal, embedded data; the notations written below represent audio samples or bit segments of embedded data.

2. System model

2.1. The proposed model

Figure 1 illustrates the process of embedding and de-embedding with the original audio signal by BM method. The audio signal in the transmitter is given by:

$$\mathrm{C} = \left\{\mathrm{C}_j;\ j = 1, 2, ..., n\right\}, \tag{1}$$

where C_j is audio sample coded by BP $[\mathrm{bit}]$ (BP: Bits PerSample), the sampling frequency SR $[\mathrm{Hz}]$ (SR: Sample Rate). The embedded data is given by the expression:

$$\mathrm{B} = \left\{\mathrm{B}_i;\ i = 1, 2, ..., m\right\}, \tag{2}$$

where B_i is a 8-bit string encoded from characters in the text. To implement embedding, C is divided into small segments (C_∂):

$$\mathrm{C} = \left\{\mathrm{C}_\partial;\ \partial = 1, 2, ... , p\right\} \text{ and } \mathrm{C}_\partial = \left\{\mathrm{C}_j;\ j = 1, 2, ... , \zeta\right\}. \tag{3}$$

The segment C_∂ has the length ζ of audio samples, in each sample C_∂, the segment C_v bit with the length of Ω $[\mathrm{bit}]$ with $v = 1, 2, ..., \zeta$ will be taken out. Embedded data is divided into segments $\left(\mathrm{B}_\theta\right)$:

$$\mathrm{B} = \left\{\mathrm{B}_\theta;\ \theta = 1, 2, ... , q\right\}. \tag{4}$$

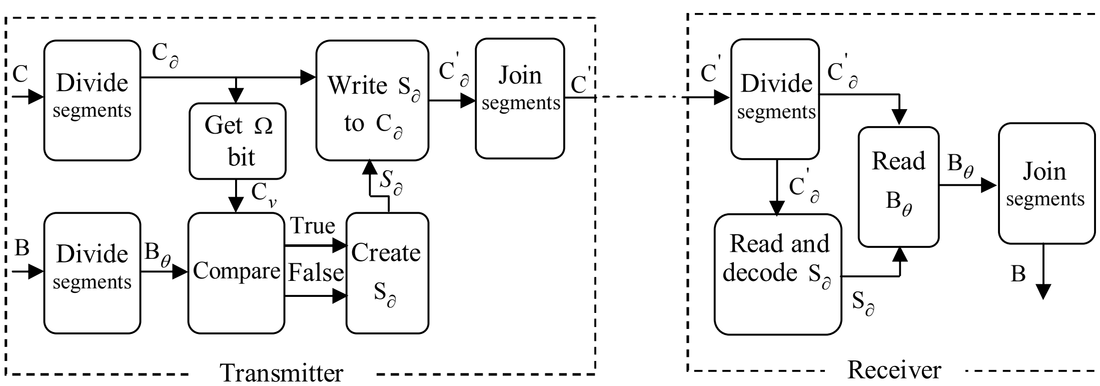


Figure 1. The process of embedding and de-embedding with the proposed method

The segment B_θ also has Ω [bit] length. The process of embedding is done by comparing the B_θ bit string with the C_v bit string of ζ samples in segment C_∂. If the comparing result is True ($B_\theta \equiv C_v$ in C_∂), S_∂ string will be created including $\mho$ [bit] to mark the position of sample which has $C_v \equiv B_\theta$ in segment C_∂, overwrite S_∂ on C_∂, repeating the comparing process with $\theta = \theta + 1$ and $\partial = \partial + 1$. If there is no segment $C_v \equiv B_\theta$ in segment C_∂, a zero, $(S_\partial = 0)$ will be used to mark the unembedding segment C_∂, overwrite S_∂ on C_∂, repeating the comparing process with $\theta = \theta + 1$. At the end of this embedding process, we have:

$$C' = \{C'_j; j = 1, 2, ..., n\} = \{C'_\partial; \partial = 1, 2, ..., p\}, \tag{5}$$

where C' is the embedded audio signal, C'_j is the audio sample of C'.

In the process of the de-embedding, the receiver handled inversely to the transmitter to get the embedded data B.

2.2. Maximum embedded capacity (D_{max})

Assuming all C_∂ are embedded, the embedded capacity will achieve maximum value

$$D_{max} = p \times \Omega \text{ [bit]}. \tag{6}$$

2.3. Real embedded capacity (D_{tt})

D_{tt} is calculated by the formula:

$$D_{tt} = r \times \Omega \text{ [bit]}, \tag{7}$$

where r is the number of times of comparisons between B_θ and C_v, which has True result. $r \le p$ from that $D_{tt} \le D_{max}$.

2.4. Embedding Error (EE)

In the general case, combine (1) with (5), we have:

$$EE = |C' - C| \tag{8}$$

$$\text{or } EE = \sum_{j=1}^{n} |C'_j - C_j| = \sum_{j=1}^{n} |EE_j|, \tag{9}$$

$$\text{where } EE_j = C'_j - C_j, \tag{10}$$

EE_j in Equ.(10) is called embedding error of the secondary audio sample j. Unlike the LSB embedding method, embedding errors in the BM method are not caused by the bit of B, but by overwriting the bit of S on C.

$$S = \{S_\partial; \partial = 1, 2, ..., p\}, \tag{11}$$

or calculated by the formula:

$$S = (r \times \mho) + (p - r) \text{ [bit]}, \tag{12}$$

where $(p - r)$ is the number of times of comparisons between B_θ and C_v, which has False result. (S_∂ has the value of 1 bit). In reality, a number of S bit overlaps with a number corresponding bit of C so $EE \le S$. Therefore:

$$EE \le (r \times \mho) + (p - r) \text{ [bit]}. \tag{13}$$

2.5. Signal- to-noise ratio (SNR)

According to [1][5][13] in general case, the SNR is calculated by the following formula:

$$\mathrm{SNR} = \frac{\sum_{j=1}^{n} \left|\mathrm{C}_j\right|^2}{\sum_{j=1}^{n} \left|\mathrm{EE}_j\right|^2}. \tag{14}$$

Combine with (9) and (10) we have:

$$\mathrm{SNR} = \frac{\sum_{j=1}^{n} \left|\mathrm{C}_j\right|^2}{\sum_{j=1}^{n} \left|\mathrm{C}'_j - \mathrm{C}_j\right|^2}, \tag{15}$$

$$\mathrm{SNR} = 10\log_{10}\left[\frac{\sum_{j=1}^{n} \left|\mathrm{C}_j\right|^2}{\sum_{j=1}^{n} \left|\mathrm{EE}_j\right|^2}\right]\left[\mathrm{dB}\right] = 10\log_{10}\left[\frac{\sum_{j=1}^{n} \left|\mathrm{C}_j\right|^2}{\sum_{j=1}^{n} \left|\mathrm{C}'_j - \mathrm{C}_j\right|^2}\right]\left[\mathrm{dB}\right]. \tag{16}$$

3. Optimization of model parameters

3.1. RB minimizing algorithm

In the proposed model, the parameters ζ, Ω, $\mho$ are closely related to each other. In this paper, we propose algorithms maximizing the value of the parameters ζ, Ω, $\mho$ in oder that EE, SNR is the best. From (4) and (7) suggests that to embed all of B_θ, it must be that: $r = q$. From (13), to reduce EE, the value of one or both of the following expressions must decrease:

$$(p - r), \tag{17}$$

$$(r \times \mho). \tag{18}$$

The expression (17) shows how many segments C_∂ do not contain data B_θ or in other words, (17) indicates the number of bits that were overwritten on C without containing information identifying of the location of B. We call these redundant bits (RB). Due to $r = q$, in order to reduce (17), the only way is reducing the value of p. It is done by stopping embedding when $r = q$ (stop creating S_∂ chain when finishing embedding components B_q). Then (11)(12)(13) will be written:

$$\mathrm{S} = \left\{S_\partial; \partial = 1, 2, ..., p'\right\}, \tag{19}$$

$$\mathrm{S} = (r \times \mho) + (p' - r) \ [\mathrm{bit}], \tag{20}$$

$$\mathrm{EE} \le (r \times \mho) + (p' - r) \ [\mathrm{bit}], \tag{21}$$

where p' is the order of the last segment C_∂ embedded $\left(r \le p' \le p\right)$.

3.2. MB minimizing algorithm

Expression (18) shows the total number of bits needed to mark the location of the segments B_θ, we call these marking bits (MB). Due to $\mho$ is the number of bits S_∂ corresponding to the case $\mathrm{C}_\mathrm{v} \equiv \mathrm{B}_\theta$ in which the the first bit always has value of 1 to inform the receiver of embedded segments C_∂. (3) shows that $\mho$ must be large enough to encode ζ of C_j samples in segments C_∂, so $\mho$ must satisfy the conditions:

$$2^{\mho - 1} = \zeta. \tag{22}$$

The number of bit reduced in each time embedding is calculated by the formula:

$$\beta = \Omega - \mho \;[\mathrm{bit}], \tag{23}$$

EE is small when $\beta > 0$, so

$$\mho < \Omega . \tag{24}$$

Therefore, the value of Ω must be chosen first, then combine (24) with (22) to find the value of $\mho$ and ζ. Ω is the number of B_θ bit, as well as the number of C_v bit. $C_v \subset C_j$ so:

$$2 \le \Omega \le \mathrm{BP} . \tag{25}$$

The value of Ω is inversely proportional to the probability of True results in comparisons B_θ with C_v in C_∂. If Ω is too big, the result of the expression $\left(p' - r\right)$ increases, which according to (20) may lead to the increase in EE. Ω is also proportional to the amplitude and frequency of C, that is if C has large C_j, the higher the differences of value C_j, the more $\left(p' - r\right)$ decreases. With the audio signals which have $\mathrm{BP} = 16\;[\mathrm{bit}]$, slow tempo (voice, light music ...) is $5 \le \Omega \le 7$, fast-paced, powerful tempo (rock, rap ...) is $8 \le \Omega \le 10$. In addition, you can use the trial embedded method with the value $\Omega = 5, 6, .., 10\;[\mathrm{bit}]$ to find the value of p'_Ω. Then select the optimum Ω, corresponding to the case which p'_Ω is the smallest.

3.3. SNR optimizing algorithm

In segment C_∂, according to (3)(8)(14) and (15) we have

$$C'_\partial = \left\{C'_j;\; j = 1, 2, ..., \zeta\right\}, \tag{26}$$

$$\mathrm{EE}_\partial = \left|C'_\partial - C_\partial\right| = \sum_{j=1}^{\zeta}\left|C'_j - C_j\right| = \sum_{j=1}^{\zeta}\left|\mathrm{EE}_j\right|, \tag{27}$$

$$\mathrm{SNR}_\partial = \frac{\sum_{j=1}^{\zeta}\left|C_j\right|^2}{\sum_{j=1}^{\zeta}\left|\mathrm{EE}_j\right|^2} = \frac{\sum_{j=1}^{\zeta}\left|C_j\right|^2}{\sum_{j=1}^{\zeta}\left|C'_j - C_j\right|^2} . \tag{28}$$

In [13], one of the methods to increase SNR of C' is dividing B before embedding. Applying to this case, we have

$$S_\partial = \left\{S_\eta;\; \eta = 1, 2, ..., h\right\}, \tag{29}$$

where h is the number of bit segments divided $1 \le h \le \mho$. The length of S_η is:

$$l_{S_\eta} = \frac{\mho}{h}\;[\mathrm{bit}]. \tag{30}$$

Also, in order to reduce the influence of S_η on C_j we choose the method which S_η is overwritten on the positions of the least significant bits of C_j.

(30) shows that if $h = \mho$, $l_{S_\eta} = 1\;[\mathrm{bit}]$, which means $\mho$ bit of S_∂ is overwritten on the place of the least significant bit of h of C_j sample in C_∂ (The rest C_j in segment C_∂ remains unchanged). Then, (27) will be rewritten as follows:

- With the segment C_∂ which has embedded data:

$$\mathrm{EE}_{\partial 1} = \left|C'_\partial - C_\partial\right| = \sum_{j=1}^{h}\left|C'_j - C_j\right| + \sum_{j=h+1}^{\zeta}\left|C'_j - C_j\right| \tag{31}$$

$$\mathrm{EE}_{\partial 1} = \sum_{j=1}^{h}\left|\left[\left(C_j \pm S_\eta\right) - C_j\right]\right| + \sum_{j=h+1}^{\zeta}\left|\left[\left(C_j\right) - C_j\right]\right| \tag{32}$$

$$\mathrm{EE}_{\partial 1} = \sum_{j=1}^{h} \mathrm{S}_{\eta} + 0 = h = \mho, \tag{33}$$

S_{η} may coincide with overwritten bits, so

$$\mathrm{EE}_{\partial 1} \le \mho. \tag{34}$$

- With the segment C_{∂} which has no embedded data, $h = \mho = 1$, using the same way, we have

$$\mathrm{EE}_{\partial 0} \le 1. \tag{35}$$

Thus, the total embedded errors calculated according to amplitude will be:

$$\mathrm{EE} \le (r \times \mathrm{EE}_{\partial 1}) + \left[(p' - r) \times \mathrm{EE}_{\partial 0}\right]. \tag{36}$$

Combining with (34) and (35), we have

$$\mathrm{EE} \le (r \times \mho) + (p' - r). \tag{37}$$

Comparing (37) with (21) shows that

$$\mathrm{EE} = \mathrm{EE}\,[\mathrm{bit}] \le (r \times \mho) + (p' - r). \tag{38}$$

So embedded errors calculated by amplitude are equal to the ones calculated by bits. Now, because only $\left[(r \times \mho) + (p' - r)\right]$ sample C_j is changed one least significant bit and $\left[\mathrm{C}'_j - \mathrm{C}_j\right] \le 1$ so (15) will be rewritten as follows:

$$\mathrm{SNR} = \frac{\sum_{j=1}^{n} |\mathrm{C}_j|^2}{\sum_{j=1}^{[(r \times \mho) + (p' - r)]} |\mathrm{C}'_j - \mathrm{C}_j|^2 + \sum_{j=[(r \times \mho) + (p' - r)]+1}^{n} |\mathrm{C}'_j - \mathrm{C}_j|^2} \tag{39}$$

$$\mathrm{SNR} \ge \frac{\sum_{j=1}^{n} |\mathrm{C}_j|^2}{\sum_{j=1}^{[(r \times \mho) + (p' - r)]} |1|^2 + \sum_{j=[(r \times \mho) + (p' - r)]+1}^{n} |0|^2} \tag{40}$$

$$\mathrm{SNR} \ge \frac{\sum_{j=1}^{n} |\mathrm{C}_j|^2}{(r \times \mho) + (p' - r)} \tag{41}$$

$$\text{or } \mathrm{SNR} \ge 10 \log_{10} \frac{\sum_{j=1}^{n} |\mathrm{C}_j|^2}{(r \times \mho) + (p' - r)} \; [\mathrm{dB}]. \tag{42}$$

Equation (41) shows that SNR under the proposed model will be greater than or equal to the sum of squares of the audio signal power divided by the total number of bits overwritten on the audio signal.

4. Experiment results

4.1. Experiment Scenario

To prove the effectiveness of the proposed method, we conducted experiments embedding 50 audio files $\{\mathrm{C}_k; k = 1, 2, .., 50\}$ with different capacities, wave format, 2-channel (stereo), $\mathrm{SR} = 44100\,[\mathrm{Hz}]$, $\mathrm{BP} = 16\,[\mathrm{bit}]$; embedded data is 10 text files $\{\mathrm{B}_v; v = 1, 2, .., 10\}$ with different capacities. Testing proposed methods with model as described in Section 2, the parameters are selected as follows: $\Omega = 8\,[\mathrm{bit}], \mho = 6\,[\mathrm{bit}], \zeta = 32\,[\mathrm{sample}]$ and using (38)(41) to calculate $\mathrm{EE\_Poroposed}_v$, $\mathrm{SNR\_Poroposed}_v$ with $(v = 1, 2, .., 10)$. To proceed to the processing in real time, the algorithms of the proposed model are programming in VisualBasic (Figure 2). Get data for comparison by performing LSB embedding [4][8][14], ELS [13] and use (9)(16) to calculate the values of EE_LSB_v, SNR_LSB_v, EE_ELS_v, SNR_ELS_v with $(v = 1, 2, .., 10)$.

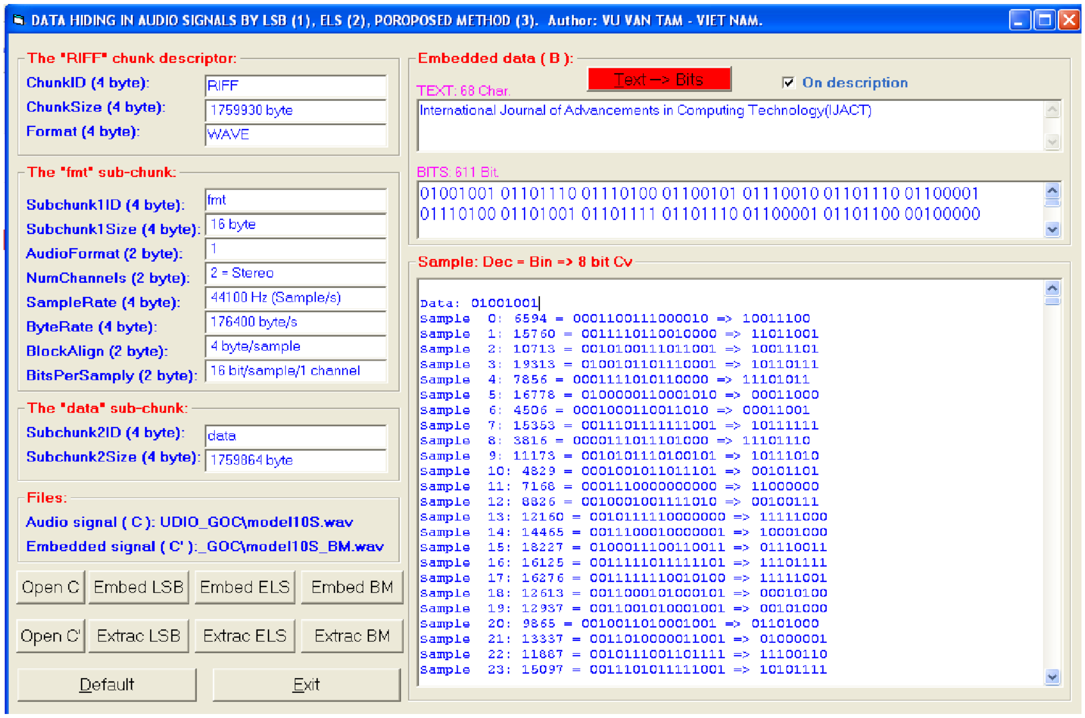


Figure 2. The interface of the embedding and decoding the embedding programs

The calculating and comparing charting results are performed in MATLAB environment. The scenario test is shown in the flow chart below:

$C = \{C_k; k = 1, 2, .., 50\}$; //50 file audio

$B = \{B_v; v = 1, 2, .., 10\}$; // 10 file text

For method = LSB, ELS, Proposed // Methods = LSB => ELS => Proposed

{ For k = 1 to 50 //file audio $C = C_1, C_2, .., C_{50}$

{ For v = 1 to 10 //file text $B = B_1, B_2, .., B_{10}$

$$\{ C'\_method_{kv} = B_v \overset{\text{Embedding method}}{\otimes} C_k;$$

$$\xrightarrow{(9),\text{if method = Proposed then } (38)} EE\_method_{kv} \xrightarrow{(16),\text{if method = Proposed then } (41)} SNR\_method_{kv}; \}$$

}

For v = 1 to 10 //Calculate and display results

{ $EE\_method_v = 0$; $SNR\_method_v = 0$;

For k = 1 to 50

{ $EE\_method_v = EE\_method_v + EE\_method_{kv}$; $SNR\_method_v = SNR\_method_v + SNR\_method_{kv}$; }

$EE\_method_v = EE\_method_v / 50$; $SNR\_method_v = SNR\_method_v / 50$;

Plot (EE_method_v, B_v); //Figure 4

Plot (SNR_method_v, B_v); //Figure 5

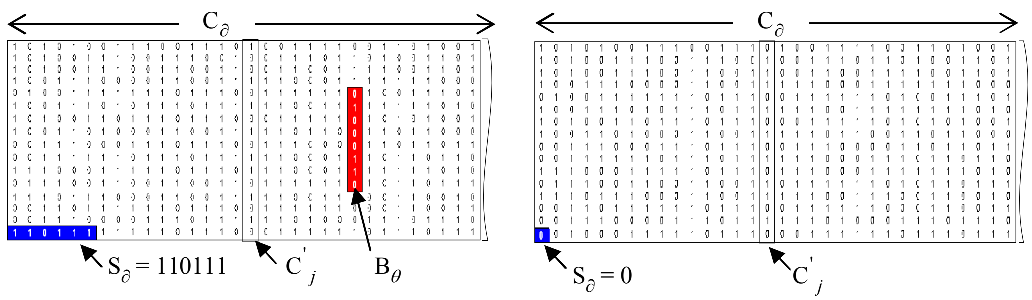
}

$\text{Plot}\left(\frac{j}{\text{SR}}\left|\text{C}'\_\text{method}_{11} - \text{C}_1\right|\right)$; //Figure 6

}

4.2. Analyzing the results

1) Figure 3 describes the process of embedding B_θ in S_∂ section in the testing model. Figure 3.a describes C_∂ embedded sections with $\text{B}_\theta \subset \text{C}_{23}$ corresponding to $\text{S}_\partial = 110111$, It shows that the data embedded in C is S (the address of B). Therefore, the security of B is very high, even when we carried out the attacks by comparing C with C'. In particular, the security will be even better when combined with a number of techniques such as: encoding S_∂, changing the location recording S_∂, splitting discontinuous segments C_∂.



a) Segment S_∂ with embedded data **b)** Segment S_∂ without embedded data

Figure 3. The description of embedding B_θ into segment S_∂

2) Figure 4 and Figure 5 are the graphs of performing quality $\text{EE}, \text{SNR}\,[\text{dB}]$, which is calculated by (9)(38), with the LSB, ELS embedding methods and (16)(41) with the proposed embedding methods, of according to size B. The results show that the proposed method significantly improves the quality $\text{EE}, \text{SNR}\,[\text{dB}]$ of the system. This has been achieved with the use of such algorithms as: RB Minimizing algorithm, making $p' \ll p$; MB minimization algorithm, according to (23) the number of bit reduced in each time embedding in test model is $\beta = 2$ [bit]; SNR optimization algorithm write the bits of S_∂ on the position of the least significant bit of the 6 audio samples, which reduces the influence to minimum of S_∂ on the audio signal. The improved quality $\text{EE}, \text{SNR}\,[\text{dB}]$ of BM method has proved the superiority of the model and algorithms proposed above.

3) Unlike the LSB and ELS embedding methods, the proposed method performed embedding only when $\text{C}_\text{v} \equiv \text{B}_\theta$. Therefore, the quality $\text{EE}, \text{SNR}\,[\text{dB}]$ depends on fast or slow tempo of audio signals and values of Ω. So in fact, the trial embedding should be done first to find the optimal value of Ω.

4) Figure 6 describes the amplitude, the distribution of EE according to signal models in LSB, ELS and proposed methods calculated by (10). It shows that the proposed method with $\text{EE}_j = 0$ or 1, and also EE is contributed all over C'. Therefore, the sound quality remains nearly unchanged and is not changed abruptly. So, C' is almost 'transparent' to ordinary users. This is achieved by using SNR optimization algorithms, as a result EE is also optimized. As EE is small, C' is almost the same as C. Consequently, the transmission of C' would reduce the doubt or attack from the enemy. Therefore, the safety of B would be increased.

5) The proposed method has complex handling rate, so if implementing embedding and decoding real-time embedding need to have good hardware. Because we only do the embedding when $\text{C}_\nu \equiv \text{B}_\theta$, the capacity of the proposed method is limited. However, this limitation can be overcome by selecting audio files with high amplitude.

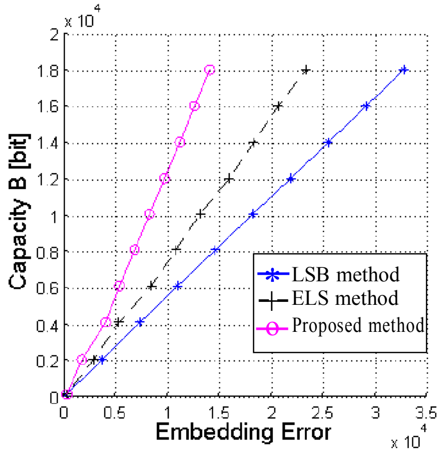


Figure 4. Quality EE

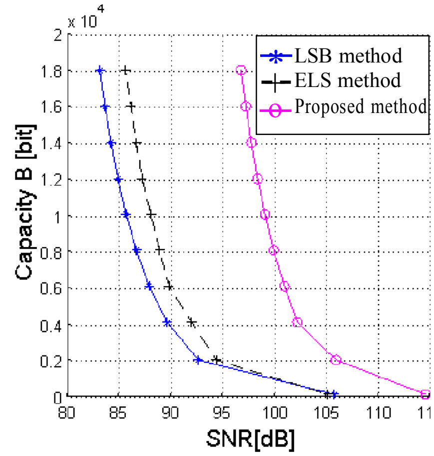
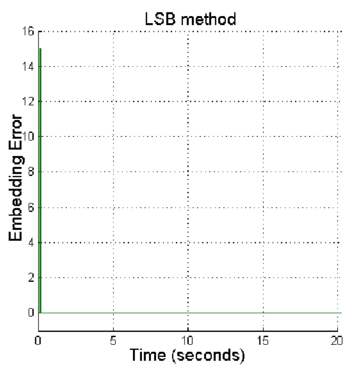
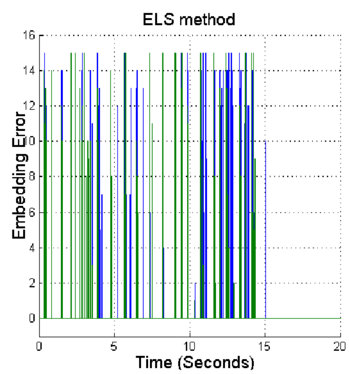


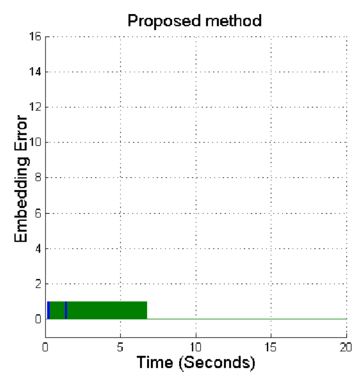
Figure 5. Quality SNR



a) LSB Method



b) ELS method



c) Proposed method

Figure 6. The amplitude and contribution of EE

5. Conclusions

In this paper, we have proposed a novel bit marking method for improvement of the embedding data into the audio signals. The completed scheme is the combination of the proposed bit marking method along with RB minimization algorithm, MB minimization algorithm, and SNR optimization. The experimental results have proven that the quality of the proposed system is better than the previous LSB [4][8][14] and ELS [13] methods. The SNR, the distribution of embedding errors, and the security of the embedded data are significantly improved. Because the embedding error is small, there is no much different between the original and embedded audios that lead to reducing of doubts and attacks from the enemy. Our proposed method has not only contributed to embed the data in audio signals more thoroughly, but also opens up a new direction for solving the embedding data into other signals.

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