A New Method for Watermarking using Genetic Algorithms

Mehdi Sadeghzadeh, and Mahsa Taherbaghal

***Abstract*—** In this paper, an innovative watermarking scheme for audio signal based on genetic algorithms (GA) is proposed. Designing an optimal audio watermarking system is an open difficult issue since its two basic performance measures, i.e., imperceptibility and robustness. So, an optimal audio watermarking scheme needs to optimally balance both imperceptibility and robustness. In order to realize such an optimal watermarking system, we propose an optimal audio watermarking scheme using genetic optimization with variable-length mechanism in this paper. The presented genetic optimization procedure can automatically determine optimal embedding parameters for each audio frame of an audio signal. Specially, employed variable-length mechanism can effectively search most suitable positions for watermark embedding, including suitable audio frames and their AC coefficients. By dint of the genetic optimization with variable-length mechanism, proposed audio watermarking scheme can not only guarantee good quality of watermarked audio signal but also effectively improve its robustness. Experimental results show that proposed watermarking scheme has good imperceptibility and high capability against common signal processing and some de synchronization attacks.

***Keywords*—** Intelligence watermarking, Genetic algorithm

I. Introduction

DIGITAL watermarking techniques as an effective means of copyright protection for digital media has been become a hot technique in multimedia security domain now. For an audio watermarking system, there are several requirements, such as imperceptibility, robustness, watermark capacity, security, reliability, etc. From the view of signal processing, imperceptibility and robustness are its two basic requirements. So, from the view of embedding strategy, both embedding strength and embedding position are two primary ingredients that affect the performance of an audio watermarking system. From the view of embedding strategy, both embedding strength and embedding position are two primary ingredients that affect the performance of an audio watermarking system. Generally, watermarking method in transform domain (such as DCT, DWT, etc.) has better robustness than those in temporal domain. For DCT domain, many researchers have indicated that different frequency bands have the different effects for imperceptibility and robustness. The watermarking methods, which embed watermark into high-frequency coefficients, usually hold good auditory quality for watermarked audio signal but can result in lower robustness. Because low-frequency coefficients belong to important auditory component and carry most of energy of an audio signal, the embedded watermark can be preserved still under certain distortions. Therefore, embedding watermark into low-frequency coefficients can bring better robustness [1]. However, the embedding way should be completed carefully because too big embedding strength may influence imperceptibility of watermark or fidelity of audio signal. Generally, embedding watermark into middle-frequency coefficients can have better robustness in case of guaranteeing imperceptibility [2].

However, designing an optimal audio watermarking system is an open difficult issue because of its two conflicting performance measures, i.e., imperceptibility and robustness. How to improve the robustness of audio watermarking system in the case of guaranteeing audio quality is a problem faced by the most of watermarking algorithms. In addition to embedding position, embedding strength of watermark is also an important factor that affects both imperceptibility and robustness. The stronger the embedded energy is, the better the robustness is, but too much energy will degrade the quality of audio signal. In order to solve this contradiction, some researchers have discussed content-adaptive audio watermarking methods integrating with auditory masking, psychoacoustic model, etc. In [3], embedded energy was estimated under unitary transformation through determining the relation between signal–noise ratio and watermark's energy, and then embedding strength was determined. A content-adaptive watermarking method applying auditory masking principle was proposed in [4]. This method selected candidate audio segments according to temporal-domain auditory masking principle and further selected their middle–high-frequency DCT coefficients to embed the watermark according to frequency-domain auditory masking principle.

In existing audio watermarking methods, perceptually acceptable performance of watermarked audio signals is usually obtained according to predefined embedding rules or empirical perceptual models. Consequently, conventional audio watermarking methods cannot essentially arrive at inherent performance upper limit. Finding optimal balance between both imperceptibility and robustness by applying optimization algorithms is another way for solving above contradiction. Thus, genetic algorithms (GAs) can be used to construct optimal watermarking schemes. Currently, GAs has been used in watermarking technique, such as audio watermarking [5–8] and image watermarking [9,10].In these

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watermarking schemes mentioned above, GAs are used either to determine optimal watermarking parameters or to find suitable embedding positions (frequency band). However, as described above, two performance measures of a watermarking system are affected by both embedding strength and embedding position. So, these watermarking schemes which only determine optimal watermarking parameters or find suitable embedding positions may not optimally balance the contradiction between both imperceptibility and robustness. In this paper, we together take account of the two ingredients that affect watermarking performance in an audio watermarking system and propose an audio watermarking scheme using genetic optimization with variable-length mechanism …. In addition, since we need to use these optimal watermarking parameters and most suitable embedding positions when extracting watermark, the proposed watermarking method is a semi-blind watermarkingmethod .

This paper is organized as follows: Section 2presentsold research in this context. Section 3 presents a watermarking algorithm employed in this paper. Section 4, experimental results and discussion are shown. Section 5 presents the conclusions.

II. Watermarking Algorithm

In our audio watermarking algorithm, watermark is converted to bits sequence then encrypt with genetic algorithm. Then audio signal is segmented into audio frames and each frame is divided into two sub-frames. Then, synchronization code is embedded into first sub-frame of each audio frame, while watermark bits are embedded into DCT coefficients of its second sub frame, as shown in Fig. 1. When data are extracted, we firstly extract synchronization code from first part of audio frame. With the determination of the position of synchronization code, we extract hidden watermark bits from DCT coefficients in remainder part of the audio frame.

first audio frame		… …	*i*th audio frame		… …
Synchronization code	Watermark	… …	Synchronization code	Watermark	… …

Fig. 1 The construction of embedding data

A. Synchronization code embedding and extraction

In this paper, a 16-bit Barker code is chosen as synchronization code. Let $B = \{b1b2 \ ... \ b16 \mid bi \in \{0, 1\}, 1 \le i \le 16\}$ be synchronization code to be embedded. The synchronization code will be embedded into first subframe of each audio frame by mean quantization in temporal domain. Because mean of samples has a smaller variance than a single sample, the effect of severe audio distortion on single bit will be reduced greatly. Formally, suppose $A1$ is first sub-frame of an audio frame. To embed the synchronization code, $A1$ is segmented into 16 sections with length n. Let $A1i = \{ai(j) \mid 1 \le j \le n\}$ be ith section of $A1$. Its mean value is calculated as follows:

$$\bar{a}_i = \frac{1}{n}\sum_{j=1}^{n} a_i(j) \tag{1}$$

Let $\Delta 1$ be the quantization step. The quantization rule of mean value is given by

$$\bar{a}'_i = \begin{cases} \lfloor \bar{a}'_i/\Delta_1 \rfloor \times \Delta_1 + 3\Delta_1/4 \text{ , if } b_i = 1 \\ \lfloor \bar{a}'_i/\Delta_1 \rfloor \times \Delta_1 + \Delta_1/4 \text{ , if } b_i = 0 \end{cases} \tag{2}$$

where $1 \le i \le 16$, $\lfloor \ \rfloor$ indicates the floor function. The sample values are modified by

$$a'_i = a_i(j) + (\bar{a}'_i - a'_i), 1 \le j \le n, 1 \le i \le 16 \tag{3}$$

where $\bar{a}'_i$ and $a'_i(j)$ are mean value of ith section and sample values after quantization, respectively.

To resist de synchronization attacks, synchronization code extraction is carried out frame-by-frame. The idea of synchronization code extraction is as follows: For each audio frame, we extract a binary sequence from it by using synchronization code extraction algorithm and then compare the extracted sequence with original synchronization code bit-by-bit. If the number of different bits among them are less than or equal to a predefined threshold, the extracted binary sequence will be determined as synchronization code of the audio frame. Thus, starting position of second sub-frame (i.e., watermarked sub-frame) in the audio frame is also determined. The synchronization code extraction algorithm is described as follows: Suppose $\hat{A}_1$ is an audio frame to be extracted and is divided into 16 sections with length n in same manner described above. For ith section $\hat{A}_i = \{\hat{a}_i(j) \mid 1 \le j \le n\}$, we calculate its mean value as follows:

$$\hat{\bar{a}}_i = \frac{1}{n}\sum_{j=1}^{n} \hat{a}_i(j) \tag{4}$$

Then, we extract the synchronization code bits using the following rule:

$$\hat{b}_i = \begin{cases} 1, \text{if } \bar{\hat{a}}_i - \lfloor \bar{\hat{a}}_i/\Delta_1 \rfloor \times \Delta_1 \ge (\Delta_1/2) \\ 0, \text{if } \bar{\hat{a}}_i - \lfloor \bar{\hat{a}}_i/\Delta_1 \rfloor \times \Delta_1 < (\Delta_1/2) \end{cases} \tag{5}$$

Usually, in most of the quantization schemes, the quantization step Δ_1 is determined experimentally or by experience. Since audio signal may suffer different attacks, determining of Δ_1 can impact accurate rate of synchronization code extraction. So, the quantization step Δ_1 will be determined automatically by optimization procedure based on GA in this paper.

B. Watermark embedding and extractio n

The watermark information will be embedded into DCT coefficients of second sub-frame of each audio frame. After correctly segmenting audio signal into audio frames and their sub-frames, we perform DCT transform on second sub-frames of audio frames respectively. Formally, suppose $A2$ is second sub-frame of an audio frame and its DCT AC coefficient set is $C = \{ci \mid 1 \le i < L\}$, where L is the length of second sub-frame of audio frame. In this paper, watermark signal to be embedded is a binary image with size $N1 \times N2$. After transforming the binary image into a one-dimensional watermark sequence, we denote the watermark sequence by $W = w1w2 ... wN1 \times N2$.

Here, a quantization index modulation technique is employed to carry out the watermark embedding because of its good robustness and blind nature. The rule for watermark embedding is as follows:

$$C'_i = \begin{cases} \lfloor c_i/\Delta_2 \rfloor \times \Delta_2 + 3\Delta_2/4, \text{if } w_i = 1 \\ \lfloor c_i/\Delta_2 \rfloor \times \Delta_2 + \Delta_2/4, \text{if } w_i = 0 \end{cases} \quad (6)$$

where $ci(1 \leq \quad i < L)$ are DCT coefficients of original audio sub-frame and $c'_i(1 \leq i < L)$ are DCT coefficients of audio sub-frame after embedding. Δ_2 is a quantization step which plays the embedding strength role. Of course, it is desired that Δ_2 should be as large as possible under the constraint of imperceptibility. Since local features of audio frames may be different, we employ different quantization steps for different audio frames. This reflects that it actually is an adaptive quantization modulation technique. In this paper, GA is employed to optimally determine these quantization steps.

For watermark extraction, we divide a tested audio signal into audio frames in the same manner. For each audio frame, after determining synchronization code and starting position of second sub-frame using the above synchronization code extraction algorithm, we can determine a second sub-frame of the audio frame. Formally, suppose $\hat{A}_2$ is the second sub-frame of an audio frame and its DCT AC coefficient set is $\hat{C} = \{\hat{c}_i | 1 \leq i < L\}$. The rule for watermark extraction is as follows:

$$\hat{w}_i = \begin{cases} 1, \text{if } \hat{c}_i - \lfloor \hat{c}_i/\Delta_2 \rfloor \times \Delta_2 \geq \Delta_2/2 \\ 0, \text{if } \hat{c}_i - \lfloor \hat{c}_i/\Delta_2 \rfloor \times \Delta_2 < \Delta_2/2 \end{cases} \quad (7)$$

audio watermarking technique in DCT domain, previous researches indicate that watermark should be embedded into middle-frequency DCT coefficients in order to get better imperceptibility and robustness. Besides, we notice that some AC coefficients are not suitable for watermark embedding due to their small coefficients, and even some audio frames also are not suitable for watermark embedding since their energies are too low. In this paper, it is automatically determined by using genetic optimization with variable length mechanism which coefficients are chosen for watermark embedding.

C. Watermark encryption with genetic algorithm

The block diagram of the proposed method is shown in Fig 2. It consists of pseudorandom sequence generator, crossover operator, and encryption and decryption modules.

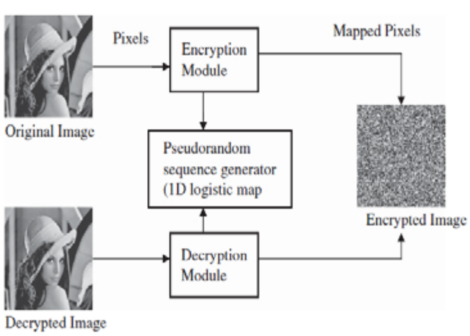

Fig **2** . The block diagram of the proposed method

The encrypting process emulates the working of the crossover operator using pseudorandom sequence. The steps for the data encryption as follows:

1. Generate the pseudorandom binary sequence using the chaos as *Yn* .
2. Convert the binary pseudorandom sequence into decimal pseudorandom sequence ranging from0 to7 as Z_n.
3. Read 16 consecutive bytes from the data file.
4. Initialize j =0
5. Initialize i =0
6. Modify the consecutive bytes using byte substitution for creating confusion, as per AES standard.
7. Take two consecutive bytes of the data stream as *A*1 *and A*2
8. Perform crossover on two consecutive bytes of the data stream as *B*1 *and B*2 by using the number *Zi* .
9. Encrypt data as *C*1 *and C*2, where
   a) $x_i = Z_i \oplus (Z_i \ll 4)$
   b) $x_{i+1} = Z_{i+1} \oplus (Z_{i+1} \ll 4)$
   c) $C_1 = B_1 \otimes X_i$
   d) $C_2 = B_2 \otimes X_{i+1}$
10. $i = i + 2$ and $j = j + 1$ repeat steps 6 to 9 until $i \leq 16$.
11. Repeat steps 5 to 10 until $j \leq 5$.
12. Again perform the byte substitution over the encrypted 16 consecutive bytes for further creating confusion.
13. Repeat steps 3 to 12 until end of the data.

III. Optimization Procedure

From the view of embedding strategy, two performance measures of an audio watermarking system, imperceptibility and robustness, are influenced by both embedding strengths and embedding positions. So, we should determine optimal embedding strengths and select the most suitable embedding positions for a watermarking system. As described above, some frequency coefficients or audio frames are not suitable for guaranteeing the robustness due to their too small coefficient or too low energy, so if a watermarking system can provide enough watermark capacity, these coefficients or frames should not be selected to embed watermark. In this paper, two tasks, determining optimal embedding strengths and selecting most suitable embedding positions, will carry out simultaneously by genetic optimization with variable-length mechanism. Specially, the variable-length mechanism can automatically select most suitable embedding positions for watermark embedding under the constraint of embedding capacity.

The block diagram of the proposed audio watermarking optimization procedure using variable-length genetic algorithm is shown in Fig. 3. The optimization procedure consists of two parts, i.e., part I and part II in Fig. 3. The embedding and extraction operations of watermark will be accomplished in part I, while part II will implement genetic operation with variable-length mechanism. For an original audio signal to be embedded, we can obtain its optimal parameters by the optimization procedure, including embedding strengths and embedding positions. According to the obtained optimal parameters, we can generate final embedding outcome by using watermark embedding algorithm again. The audio watermarking optimization procedure using variable-length genetic algorithm can be described below.

D. Chromosome encoding

In the proposed optimization procedure, population consists of *J* chromosomes, and these chromosomes are encoded by using a binary string encoding approach due to its simple realization. Since variable-length mechanism is employed, chromosomes in the population may have different lengths. Every chromosome expresses a group of embedding parameters, which correspond to a potential candidate of optimal watermarking parameters, so J chromosomes in the population actually are J such candidates. As described above, each audio frame includes two sub-frames, where the first sub-frame has an embedding parameter and the second sub-frame has a number of embedding parameters which correspond to its AC coefficients. Here, each embedding parameter uses 32-bit binary string.

Figure 4 shows an example of a chromosome encoding. Assume that an audio signal to be embedded has M audio frames, which are denoted by frame #1, frame #2, . . ., frame #M, respectively. Each audio frame has two sub-frames; for example, "sub-frame 1" and "sub-frame 2" are two sub-frames of jth audio frame in Fig. 4, and second sub-frame "sub-frame 2" has $L-1$ AC coefficients, $c\,j1$, $c\,j2$, . . . , $c\,j(L-1)$. Each chromosome in the population consists of a number of parameter blocks, where each parameter block expresses embedding parameters of some audio frame. For example, the chromosome in Fig. 4 has m parameter blocks, where ith parameter block corresponds to embedding parameters of jth audio frame. For reasons of the variablelength mechanism, we should notice two aspect as follows: On the one hand, all parameter blocks may correspond to some audio frames among all audio frames of an audio signal, i.e., $m \le M$ in Fig. 4. That is, some audio frames are used to embed watermark, while other audio frames are not embedded. On the other hand, when an audio frame is used to embed watermark, it's some AC coefficients are selected as candidates for watermark embedding, but watermark are not embedded into its other AC coefficients. For example, in Fig. 4, $\Delta 1$ corresponds to first sub-frame of jth audio frame, while $\Delta_{21},\Delta_{22},\ldots,\Delta_{2r}$ correspond to some AC coefficients of second sub-frame, $c\,j1$, $c\,j3$, . . . , $c\,jr$, respectively. So, some AC coefficients may not be used, such as $c\,j2$, $c\,jL$, and consequently $r \le L$.

In addition, since a chromosome only encodes the information about watermarking parameters, we design a position string for each chromosome in the population when realizing. The position string consists of some sub-strings, where each sub-string reflects the position index of corresponding watermarking parameter in the chromosome associated with it. It should be noticed that the position string does not perform genetic operations unlike the chromosome and also is a variable length. However, some sub-strings in a position string are added or removed due to inserting new parameter block or combination operation like in a chromosome.

E. Initialization

In initialization phase, two indexes, embedding amount N and watermark capacity $N0$, are used and meet that $N \le N0$. In this paper, watermark signal to be embedded is a binary image with size $N1 \times N2$, so $N = N1 \times N2$. If audio signal is divided into M audio frames and their second sub-frames are with length L, then $N0 = M \times (L-1)$. Every chromosome in the population is generated as follows: Firstly, two random numbers, $m \in [1,M]$ and $r \in [1, L]$, are generated such that $m \times r = N$. Secondly, m audio frames are selected randomly from all audio frames of an audio signal and the chromosome consists of m parameter blocks which correspond to these selected audio frames. For each selected audio frame, we randomly choose r AC coefficients from $L-1$ AC coefficients of its second sub-frame as embedding positions. Finally, each chromosome has $m \times (r+1)$ initial quantization steps, which are random numbers in $(0, 1)$. Each quantization step uses 32-bit binary encoding, so each initialized chromosome has $m \times (r+1) \times 32$ bits in all. Moreover, the position string associated with a initialized chromosome consists of the positions of corresponding quantization steps in the chromosome.

F. Accomplishing watermark embedding and compting PSNR values

As described above, each chromosome in the population corresponds to a potential group of optimal watermarking parameters. So J chromosomes correspond to J watermarking parameters. Assume that $A = \{a(i) | 1 \le i \le K\}$ is an audio signal considered for embedding watermark, where K is the length of A. Here, it should be noticed that when watermark capacity provided by a chromosome is greater than the number of watermark bits, we will employ repeatedly embedding strategy. According to each chromosome (a group of watermarking parameters), we can generate a watermarked audio signal by running watermark embedding algorithm. Consequently, we can obtain J watermarked outcomes, denoted by $\bar{A}_1, \bar{A}_2, \cdots, \bar{A}_J$. Formally, $\bar{A}_j = \{\bar{a}_j(i) | 1 \le i \le K\}, 1 \le j \le J$.

In order to evaluate the imperceptibility of a watermarked outcome, peak signal-to-noise ratio (PSNR) is used to measure the difference between an original audio signal and a watermarked audio signal. For jth watermarked outcome, its PSNR is calculated by

$$PSNR_j = 10\log_{10}\left[\frac{K \times \max_{1\le i\le K}|a(i)|^2}{\sum_{i=1}^{K}\left(a(i)-\bar{a}_j(j)\right)^2}\right] \tag{8}$$

Generally, a larger PSNR indicates that the watermarked audio signal more closely resembles the original audio signal, meaning that the watermarking algorithm makes the watermark more imperceptible.

G. Attacking watermarked audio signals

It is well-known that when a watermarked audio signal is used or transmitted, it may suffer from different signal processing operations or attacks. Therefore, we wish that the watermarked outcome should have high capability to resist attacks, that is, high robustness. In this paper, several attacking methods are considered to attack all watermarked audio signals, such as MP3 compression, low-pass filtering, median filtering, additive noise, re-quantizing, random cropping, amplitude variation, pitch shifting, time-scale modification, and jittering. Formally, assume that S attacking methods are considered. For each watermarked outcome $\bar{A}_j$,

by attacking it using these methods, we can obtain S attacked audio signals, denoted by $\hat{A}_{js}, s = 1,2,\cdots,S$. So, in each generation of the optimization procedure, $J \times S$ attacked audio signals are generated in all, i.e.,

$\hat{A}_{js}|s=1,2,\cdots,S\,, j=1,2,\cdots,J,$ where $\hat{A}_{js}=\{\hat{a}_{js}(i)|1\le i\le K\}, 1\le s\le S, 1\le j\le J.$

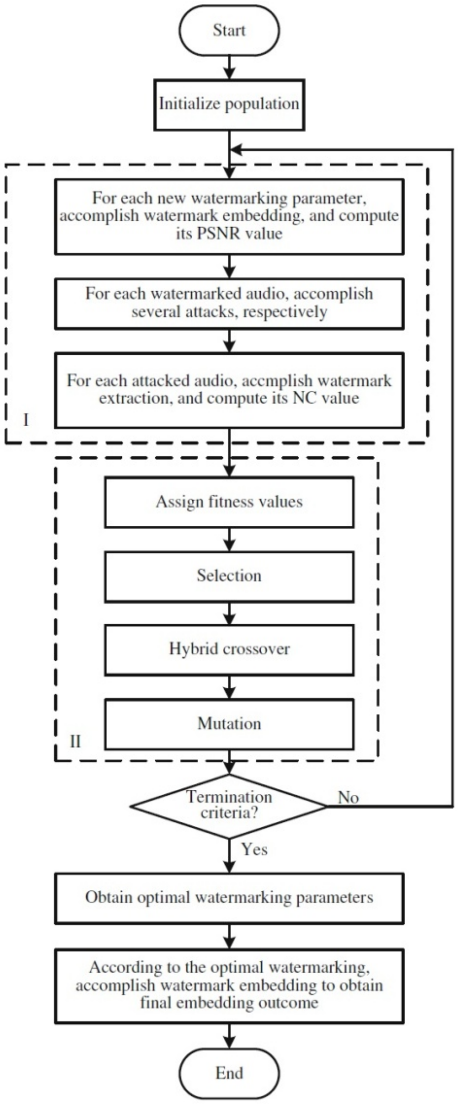

Fig. 3 Flowchart of the proposed audio watermarking optimization procedure using variable-length genetic algorithm

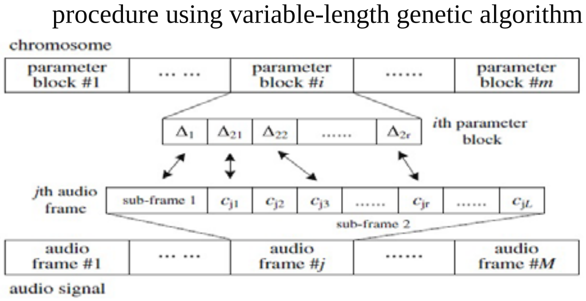

Fig. 4 Example of chromosome encoding

H. Accompolishing watermark extraction and computing NC values

In order to evaluate robustness against attacks, we should extract watermarks from attacked audio signals by using watermark extraction algorithm. For robustness capability, normalized correlation (NC) is employed to measure similarity between an original watermark and the corresponding extracted watermark. Generally, a lager NC reveals that the extracted watermark resembles the original watermark more closely.

Assume that $W=\{w(u,v)|1\le u\le N_1, 1\le v\le N_2\}$ is original watermark image. For each attacked audio signal $\hat{A}_{js}$, watermark image extracted from it by using watermark extraction algorithm is denoted by $\widehat{W}_{js}=\{\widehat{w}_{j,s}(u,v)|1\le uN_1, 1\le v\le N_2\}$, and its NC is defined as follows:

$$NC_{js}=\sum_{u=1}^{N_1}\sum_{v=1}^{N_2}\frac{[w(u,v)\times\widehat{w}_{js}(u,v)]}{[w(u,v)]^2} \qquad (9)$$

I. Assigning fitness values

For each chromosome in the population, after obtaining the PSNR of its watermarked outcome and the NC values of corresponding outcomes, we can calculate its fitness value. Formally, assume that *j*th chromosome has a watermarked outcome $\bar{A}_j$ whose PSNR value is PSNR*j* and *S* attacked outcomes $\{\hat{A}_{js}|s=1,2,\cdots,S\,,\ j=1,2,\cdots,J\}$ whose NC values are NC*js*, *s* = 1, 2, . . . , *S*, respectively. So, its fitness value is calculated by the following fitness function

$$Fitness_j=\mathrm{PSNR}_j+\sum_{s=1}^{S}\sigma_s\mathrm{NC}_{js} \qquad (10)$$

Where σs is a weighted factor and plays the role to balance both imperceptibility and robustness. Simply, we set the uniform σs, i.e., $\sigma s=\sigma$.

J. Selection

In this paper, roulette wheel selection is used as selection method of GA. In roulette wheel selection, the probability that a chromosome will be selected is proportional to its fitness.

K. Hybrid crossover

In this paper, a hybrid crossover method is employed for the crossover of chromosomes and substrings. Assume that *pc* is crossover rate. The hybrid crossover has three ways for generating offspring (see Fig. 5).

1. Perform only uniform crossover for substrings with a probability *pc* × *pc*.
2. Perform only two-point crossover for chromosomes with a probability 1 − *pc*.
3. Perform both uniform crossover for substrings and two-point crossover for chromosomes with a probability *pc* × (1 − *pc*).

The employed uniform crossover and two-point crossover are described below.

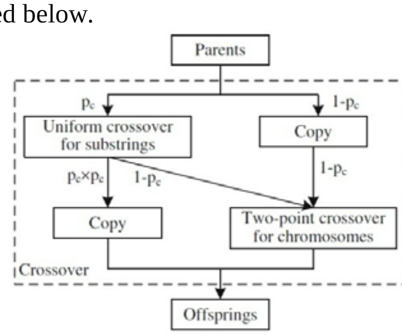

Fig. 5 Hybrid crossover for chromosomes and substrings

The uniform crossover for substrings is limited to the scope of a substring rather than the whole chromosome. Figure 6 shows an example of the crossover where the bits of offspring are equiprobably inherited from parent 1 and parent 2. Since chromosomes in the population may have different lengths, a special scheme is used to perform the uniform crossover on chromosomes with different lengths. Figure 7 illustrates the procedure of the crossover via an example. When two chromosomes have different lengths, shorter chromosome (parent 1) randomly maps its substrings into longer chromosome (parent 2). Consequently, the uniform crossover is performed between corresponding substrings.

Two-point crossover for chromosomes only exchanges several substrings in a pair of parents but not modifying the bits of each substring. This provides a way of changing the length of chromosome. Figure 8 shows an example of two-point crossover for two chromosomes. Two-point crossover for chromosomes is performed as follows: Firstly, each chromosome is divided into three parts according to its two random points, and then two chromosomes exchange their first and third parts. Secondly, two chromosomes generated after exchanging further perform a combination operation. If a chromosome has two substrings of same position, then the two substrings are combined into a substring whose strength equals to maximum of embedding strengths of the two substrings.

L. Mutation

Three mutation means are employed in this paper, including the mutation of coding bit of chromosome, insertion of a new substring, and insertion of a new parameter block. Specially, introducing insertion of new substring or parameter block will effectively ensure the diversity of population and adapt to the need of variable-length mechanism. The mutation of coding bit of chromosome adopts the bit-flip mutation. The principles of inserting new substring or parameter block are described in Fig. 9. Figure 9a shows that a randomly generated substring is inserted into a chromosome, while another technique, inserting new parameter block, is given in Fig. 9b.

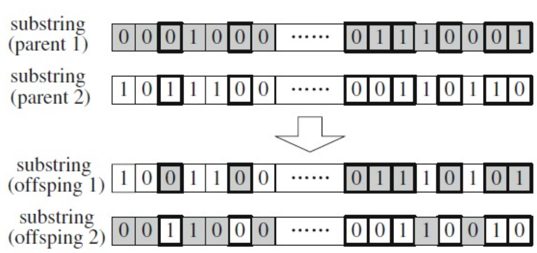

Fig. 6 An example of uniform crossover for substrings

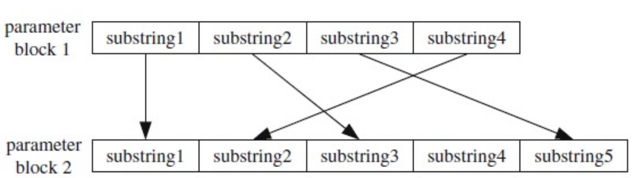

Fig. 7 An example of uniform crossover for substrings

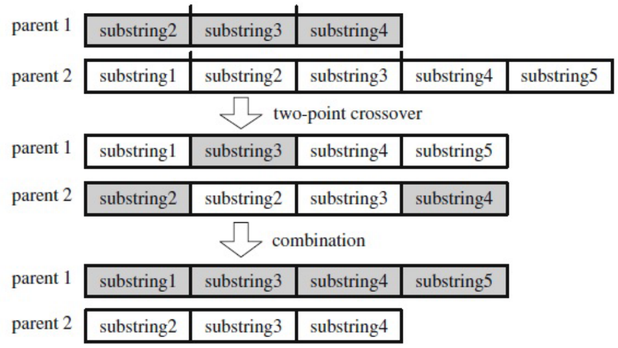

Fig. 8 An example of two-point crossover for chromosomes

M. Generating final embedding outcome

In this paper, we employ most generation number as termination criteria of the optimization procedure. After arriving the given most generation number, the best chromosome which has most fitness value in the population is regarded as our optimal result. From the best chromosome, we can obtain optimal embedding parameters and embedding positions (most suitable audio frames and their most suitable AC coefficients for watermark embedding).

According to these embedding parameters and embedding positions, we accomplish watermark embedding on the original audio signal by using watermark embedding algorithm again. Consequently, we can obtain a final embedding outcome which meets the optimal balance between both imperceptibility and robustness.

N. The decryption watermark process

The steps for decryption are just the reversal of the encryption. First, generate the pseudorandom sequence using chaos and then use the pseudorandom sequence and crossover operator to decrypt the data.

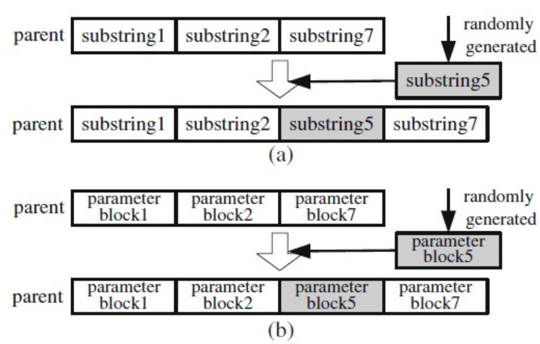

Fig. 9 An example of inserting new substring or parameter block in mutation operation. **a** Insertion of new substring. **b** Insertion of new parameter block

III. Results and Discussion

To survey presented algorithm we use two binary pictures (Fig. 10) and two audio signals, such as classic (Fig. 11a), electronic (Fig. 11b) for embedding watermark on them.

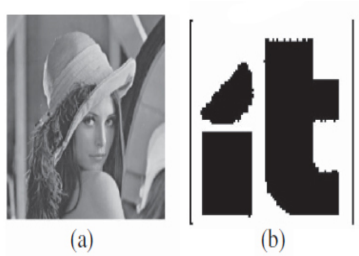

Fig. 10 Watermark images a. Lena image, b. it image

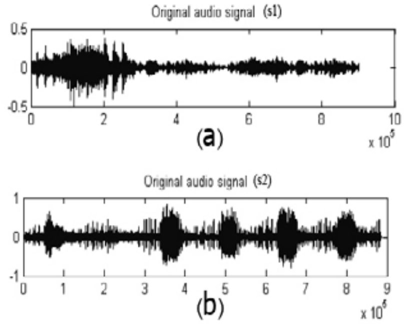

Fig. 11 The waveforms of original audio signals a. Classical music, b. Electronic music

For the above two audio signals, we perform proposed genetic optimization procedure using variable-length mechanism respectively and then obtain their optimal embedding parameters after 100th generation, However, since the number of optimal watermarking parameters contained in an audio signal after optimizing is too large and optimal watermarking parameters of different audio signals after optimizing have larger differences, we do not list these optimal watermarking parameters and their positions due to space limit of this paper. Here, we only give some information about optimal quantization steps of several tested audio signals, such as mean value and variance of optimal quantization steps of each audio signal also same values when embedded watermark don't be encrypt that say Hong Peng and Jun Wang in 2010, shown in Table I.

TABLE I
ABO MOSOSOME INFORMATIONT AUUT OPTIMAL CHORODIO MES FOR DIFFEREN TIMIZINGSIGNALS AFTER OP

Image watermark	**Different bits in this paper**	**Different bits Hong Peng and Jun Wang**
Lena	23	31
it	8	14

We encrypt the watermark binary image with presented algorithm, then after extraction this from audio signal extracted watermark image be decrypted with decryption process. The unlike bits in the extracted watermark with presented algorithm and the unlike bits in extracted watermark with Hong Peng and Jun Wang is shown in Table II. Fig. 12a,d shows image watermarks before encryption, Fig. 12b,e shows encrypted watermarks, Fig. 12 shows decrypted watermarks.

TABLE I I
ADIFFERENT BITS BETW RMEN IMAGE WATERMARKRKAND EXTRACTED WATE

Audio Signal	**Optimal chromosome size (bit) in this paper**	**Optimal chromosome size (bit) Hong Peng and Jun Wang**	**Optimal quantization step mean value in this paper**	**Optimal quantization step mean value Hong Peng and Jun Wang**
Classic	402060	420000	0.305	0.327
Electronic	436200	269472	0.312	0.351

IV. CONCLUSIONS

In this paper, we present an optimal audio watermarking scheme using variable-length genetic algorithm. In the scheme, first the image watermark encrypted using genetic algorithm, then genetic optimization procedure using variable-length mechanism is used for watermark embedding simultaneously. In the scheme, the balance between imperceptibility and robustness is viewed as an optimization problem. For extraction process we extract encrypted watermark from audio signal using variable-length genetic algorithm, then decrypt extracted watermark using genetic algorithm. The embedding parameters of each audio frame are determined automatically, and employed variable-length mechanism can search most suitable audio frames and AC coefficients for watermark embedding simultaneously. After accomplishing watermark embedding, we need to save these optimal watermarking parameters in order to use them in the future during extracting watermark. So, our watermarking method is a semi-blind watermarking method. Experimental results show that proposed optimal audio watermarking scheme using encrypt watermark before embedding to audio signal and variable-length can effectively improve the performance of an audio watermarking system and is robust against common signal processing and de synchronization attacks.

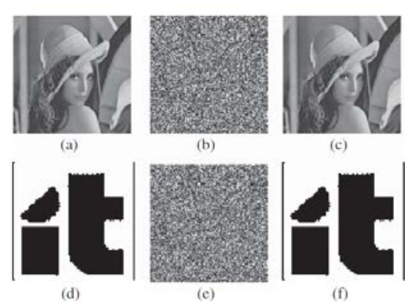

Fig. 12 a) Lena Image b) Encrypted Image c) Decrypted Image d) It Image e) Encrypted Image f) Decrypted Image

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