A Mean quantization based synchronous digital audio watermarking method on wavelet domain

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Abstract. In this paper, a mean quantization based synchronous digital audio watermarking method on wavelet domain is proposed, which is a blind detection algorithm to detect without the original audio signal. It introduces coefficient of mean quantization algorithm, and averagely quantize low-frequency coefficients for watermarking after wavelet decomposition to audio signal. It also bring the concept of embedded synchroniztion signal to determine the position of watermark for resisting the shift, cutting and other attacks. The results of experiments validate the robustness and resistance to attack of this method.

I. INTRODUCTION

With the rapid development of network technique and multi-media technique, the Internet has become an important way to disseminate knowledge and information. However, the followed issues on piracy and copyright disputes become pressing problems to address. Digital watermark technology is now recognized as the solution is an effective solution to copyright issues, which can declare the ownership of digital products to identify the buyer or provide relevant information and content.

Digital Audio Watermarking algorithm is an essential part of audio watermarking technology. The most crucial properties of digital audio watermark are the non-auditory and robustness. In the common digital audio watermarking technology, because of the need for copyright protection of digital music products, transform domain audio watermarking algorithm has made great progress. In this algorithm, the literature[1] presented a wavelet transform of the audio watermarking algorithm, which hided watermark ininvisibility and robustness.

The paper [2] proposed a wavelet domain quantization based digital audio watermarking algorithm which extracted watermark without the original digital audio signal using modulated signals embedded by quantizing the discrete wavelet transform coefficients. The paper [3] proposed a mean quantization digital audio watermarking in wavelet domain algorithm which played better robust to mp3, quantization, re-sampling, low pass filtering and noise and other attacks.

The paper [4] is proposed based on auditory masking effect and the wavelet transform of the audio watermarking algorithm, which has a strong hidden nature of the watermark. The paper [5] proposed a synchronization of digital audio watermarking algorithm, which located the embedding position of the synchronous signal in order to improve the accuracy of extracting the watermark, and added the segment information into the watermark by redundant coding. It can enhance the error correction capability.

II. MEAN QUANTIZATION PRINCIPLE

Assuming that the set of K wavelet coefficients as $\{x_0,...,x_{K-1}\}$, its mean value is $\bar{x} = \frac{1}{K}\sum_{i=0}^{K-1} x_i$ .Assuming in the mean $\bar{x}$, the error caused by a embedded watermark bit during the

use of quantitative methods is Δ, the mean of K signals after a embedded watermark bit w_i is $\overline{x}^* = \overline{x} + \Delta$, It can be seen that the set of K wavelet coefficients of all individual coefficients will be modified accordingly, that is

$$x_i^* = x_i + \Delta, \quad i = 0,...,K-1 \tag{2-1}$$

In which x_i^* is one of the modified coefficients. Suppose embedded watermark signal suffers a certain attack in the transmission process, the error generated by the attack on each signal is δ_i, one of the K wavelet coefficients after the attack is $\hat{x}_i = x_i^* + \delta_i, \quad i = 0,...,K-1$, and the mean value of K wavelet coefficients is $\overline{\hat{x}}_i = \frac{1}{K}\sum_{i=1}^{K-1}\hat{x}_i = \overline{x}^* + \overline{\delta}_i$. Assume $\delta_i \sim N(0,\sigma^2)$, which means δ_i obeys Gaussian distribution with mean 0 and variance σ^2, the average of K random variables obey Gaussian distribution still obeys Gaussian distribution, but the variance is reduced by K times, that is $\overline{\delta} \sim N(0,\frac{\sigma^2}{K})$. Therefore, compared to quantitative methods with a single factor, Mean quantization can reduce the variance caused by the attack and coefficient changes in the and improve robustness. Fig.1 shows the statistical distribution of wavelet coefficients changes.
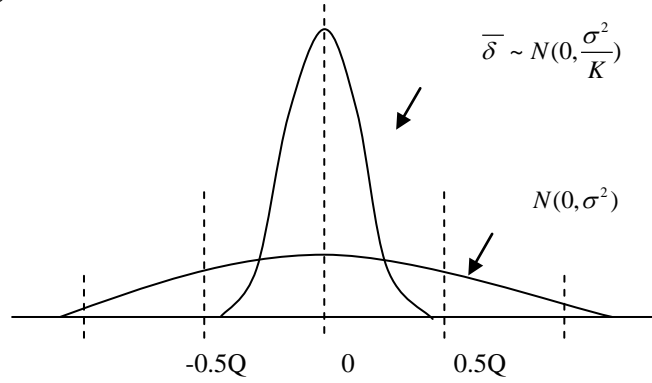

Figure 1.   Gaussian distribution with different variances

III. THE SELECTION AND DETERMINATION OF THE SYNCHRONIZATION SIGNAL

In this paper, a m-sequence is used as the synchronous signal, first transform the m-sequence $\{c_n\}\ (c_n = 1,0)$ into $\{c_n'\}\ (c_n' = 1,-1)$, that is $c_n' = 2c_n - 1$.

Assume that $\{a_n\}$ and $\{b_n\}\ (a_n, b_n = 1,-1)$ are two m-sequences with the same period T, so their autocorrelation coefficient is defined as

$$\rho_{a,b}(\tau) = \frac{1}{T}\sum_{n=1}^{T} a_n b_n - \tau \tag{3-1}$$

The autocorrelation coefficient of m-sequence $\{a_n\}$ is defined as

$$\rho_a(\tau) = \frac{1}{T}\sum_{n=1}^{T} a_n a_n - \tau \tag{3-2}$$

The autocorrelation coefficient of m-sequence has the following properties:

$$\rho_a(\tau) = \begin{cases} 1, & \tau = 0 \\ -1/T, & \tau \neq 0 \end{cases} \tag{3-3}$$

Let $\{a_n\}$ be the original m-sequence used as a synchronization signal, $\{b_n\}$ is a sequence to be tested, if $\rho_{a,b}(0) \geq n/T$, then determine that $\{b_n\}$ is a synchronization signal, in which n is an odd threshold obtained by a large number of experiments, if there are $(T+n)/2$ similarities in the sequence to be tested, then consider that $\{b_n\}$ is a synchronous signal.

IV. WATERMARK EMBEDDING ALGORITHM

A. Pretreatment of the watermark

In order to improve the robustness of digital watermarking algorithm, making the added watermark can resist signal processing attacks such as shear and resampling methods, and in order to achieve the purpose of security and confidentiality in the meantime, Arnold transform is used in this paper, use the scrambling transform on the binary image W, in which the number of transformation is use as the key. In order to embed the two-dimensional watermark in the one-dimensional digital audio signal, reduction of the dimensions must be done. In addition, to further improve the resistance abilities against simultaneously attack such as shear, translation, etc. An synchronous sequence required to be added in the embedded sequence. The m-sequence is used as a synchronous signal here. Thus, the structure of the final embedded audio signal sequence $V1$ is shown in Fig. 2.

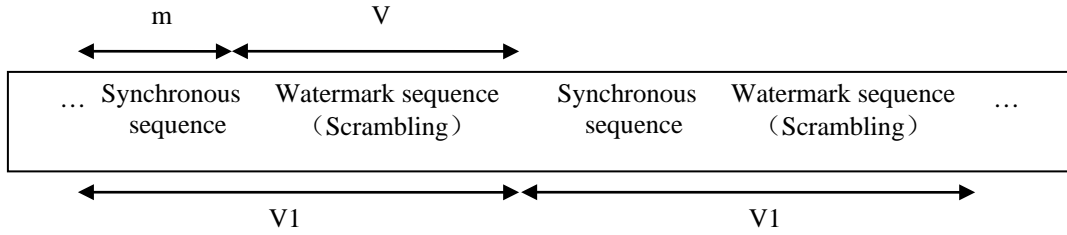

Figure 2. Composition structure of watermark sequence to be embedded

B. Audio signal segments

Suppose that A represents the original digital audio signal contains L samples of data, It can be expressed as

$$A = \{a(l), 0 \le l < L\} \tag{4-1}$$

Which $a(l) \in \{0,1,2,...,(2^P - 1)\}$ is the magnitude value of the lth data, P is the bit numbers used for each data. Separate the original audio signal into two parts, the watermark embedded relevant part A_e and the watermark embedded irrelevant part A_r, that is

$$A = A_e + A_r \tag{4-2}$$

Assuming the number of data for each audio segment is L, the total number of audio data is N, embed a pixel value in each data segment, in order to embed all $M1 \times M2$ watermark pixels, the number of original audio data that must be met:

$$N \ge M1 \times M2 \times L \tag{4-3}$$

Divided the audio A_e used to embed watermarks into $M1 \times M2$ segments

$$A_e = \{A_e(k), 0 \le k < (M1 \times M2)\} \tag{4-4}$$

C. Wavelet decomposition of each audio segment

Select the appropriate wavelet base to do discrete wavelet transform on watermark embedded related audio A_e, Daubechies-4 wavelet base is chosen in this paper, and 3-layer wavelet decomposition is used on related audio, that is

$$D = DWT(A_e) = CD_1 \oplus CD_2 \oplus CD_3 \oplus CA_3 \tag{4-5}$$

D. Watermark embedding after the low frequency coefficients mean quantization

Consider that most of the audio signal energy concentrated in the low frequency components, with good stability, resistance to noise and other signal processing abilities. To ensure the robustness of the watermark, select CA_3 as the region of watermark embedding, its coefficient is denoted by $X = \{a_i | a_i \in CA_3\}, i = 0,...,M-1\}$, M is the sum of the coefficients of the sub-band CA_3, then the watermark embedding is completed.

E. Obtain the audio signal with watermark by using the inverse wavelet transform

Replace the original CA_3 with low frequency coefficient CA_3' obtained by Watermark embedding, the detail component $CD_i (i = 1,2,3)$ maintains the same, then we have wavelet transformation territory expression after the embedding

$$A_e' = IDWT(CD_1 \oplus CD_2 \oplus CD_3 \oplus CA_3') \tag{4-6}$$

Finally carry on wavelet reconstruction, replace the original audio signal A_e with the audio signal A'_e with watermark, irrelevant part A_r remains the same, then we have the audio signal A' with watermark.

$$A' = A'_e + A_r \tag{4-7}$$

V. WATERMARK EXTRACTION ALGORITHM

A. audio signal segmentation

Separate the audio signal to be tested A_s into two parts, the watermark embedded relevant part A_{se} and the watermark embedded irrelevant part A_{sr}, that is

$$A_s = A_{se} + A_{sr} \tag{5-1}$$

Which $A_{se} = \{A_{se}(k), 0 \leq k < M1 \times M2\}, A_{se}(k)$ expresses the k th data segment.

B. Segmentation wavelet transformation

Choose the same wavelet base with the embedded watermark-- Daubechies-4 wavelet base to do 3-layer wavelet transform on watermark embedded relevant part A_{se}, choose approximate component coefficient after the transformation to do the next step processing, the result of wavelet transformation is:

$$D_s = DWT(A_{se}) = CD_{s1} \oplus CD_{s2} \oplus CD_{s3} \oplus CA_{s3} \tag{5-2}$$

C. Watermark extraction

Select the low frequency component CA_{s3}, Use the same measure during embedding to divide coefficient set composed by CA_{s3} into array, extract average value of a row. Suppose the average value is $\overline{Y'(l)}$, Extract watermark sequence $\{w'(l)\}$ from $\overline{Y'(l)}$, that is

$$w'(j) = \left\lfloor \overline{Y'(l)}/Q_1 + \frac{1}{2} \right\rfloor \% 2 \quad l = 0, ..., L-1 \tag{5-3}$$

D. Dimension rising and counter scrambling

A chaotic image $w(i,j)$ is obtained after Dimension rising on the extracted watermark bit $V'(l)$, then we can obtain the watermark image by doing inverse Arnold transform according to the key on the chaotic picture.

$$W' = \{w'(i,j), 1 \leq i \leq M1, 1 \leq j \leq M2\} \tag{5-4}$$

VI. WATERMARK EXTRACTION ALGORITHM

In the experiment, choose a section of music which broadcast time is a 21 second, quantification precision is 16, the sampling rate is 44100Hz and sampling points are 911757as the original signal, its oscillograph is shown in Fig.3. The m-sequence which synchronized signal cycle is 63 and the quantification length of stride is 0.2 cannot notice the differences after inserting watermark in the audio data, the audio frequency oscillograph of embedded watermark is shown in Fig.4.
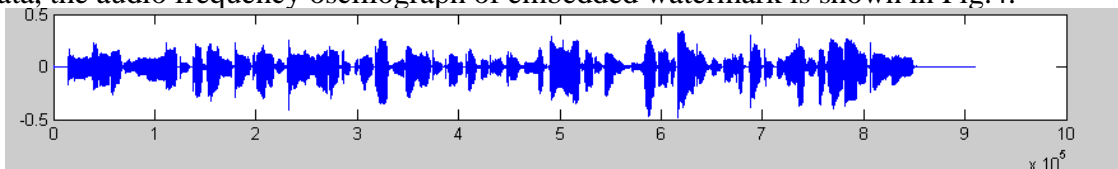

Figure 3.   Oscillograph of original audio signal
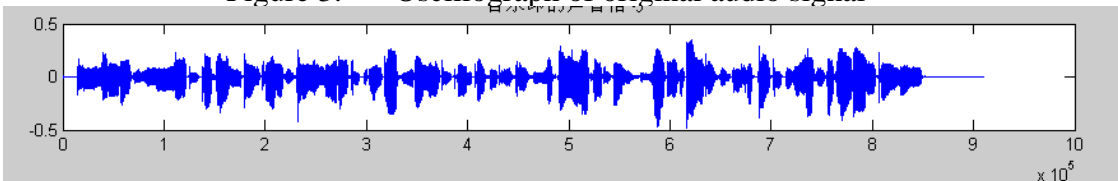

Figure 4.   Audio signal with embedded watermark

VII. CONCLUSION

In this paper, we proposed a mean quantization based self-synchronous digital audio watermarking method in wavelet domain. In this method, each watermark bit is embedded a group of average wavelet coefficients by quantizing, and the watermark is extracted without the original audio signal. Experimental results show that the mean quantization watermarking system has been enhanced relative to the overall performance of the general quantization based watermarking system; that's to say, without introducing perceptual noise, the watermark robustness of the system have been well improved.

This algorithm has the following characteristics: 1) The algorithm is a blind detection algorithm to detect without the original audio signal; 2)Bringing the concept of coefficient mean quantization, this algorithm can embed watermark after the wavelet decomposition of audio signals and coefficients mean quantization on low frequency; 3) It also bring the concept of embedded synchroniztion signal to determine the position of watermark for resisting the shift, cutting and other attacks.

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