Anti-Synchronization Audio Watermarking Algorithm Based on Absolute Mean Quantization and Parameter Estimation

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Abstract **— The paper proposes an anti-synchronization audio watermarking algorithm based on absolute mean quantification and parameter estimation, and combines implicit synchronization technology and explicit synchronization technology, for implementing precise positioning of feature points. By theoretical analysis and experimental results of quantification scheme, the paper designs a multi-valued quantification based on absolute mean. For de-synchronized attack such as time scaling, under-sampling and spring attack, the paper proposes a TSM parameter estimation algorithm for audio recovery to implement re-synchronization of watermark embedding and watermarking extraction. The simulation experiment indicates that the algorithm in the paper not only has great robustness for conventional attack, but also can effectively resist de-synchronized attack.**

Keywords - Digital Security, Absolute Mean Qualification, Audio Watermarking, Anti-synchronization.

I. INTRODUCTION

With mass production and issue of digital audio, copyright protection of digital audio has become an important research direction of digital watermarking. According to the definition of robustness of audio watermarking by IFPI, audio watermarking needs to satisfy the following requirements, (1) imperceptibility, watermark can't influence the audio quality, and the audio embedding watermark can provide 20dB or higher signal to noise ratio. (2) a certain watermarking capacity, the data channel embedding watermark has 20bps bandwidth at least. (3) Robustness, watermark can resist various attacks such as additive or multiplicative noise, MP3 compression, two continuous D/A and A/D conversion, time stretch (10%), resampling, re-quantification and filtering.

After years of research, the robust audio watermarking algorithms are proposed, which can solve some common audio signal process and attacks such as MP3 compression, re-sampling, re-quantification, noise and random cutting. However, audio watermarking can't solve the robustness of de-synchronized attacks including TSM, under-sampling and Jitter attack. At present, the methods solving de-synchronized attack are as follows. (1) Exhaustive search, the simplest and direct test scheme, but it has the problem of large computation and high false alarm rate. (2) Explicit synchronization. A synchronous code (such as bark code/M sequence) is embedded into the carrier to mark the embedding position of the watermark, but the synchronous code is an additional hidden information, which not only can be broken by conventional audio signal process and hostile attack, but also has the problems of missing inspection and false detection. (3) Implicit synchronization. The feature points of audio carrier are used to identify the embedding position of the watermark. Implicit synchronization is based on audio, so it can't change the audio. But it has the following problems. 1) Different audios require different parameters, and the selection of parameters is the key of synchronization technique. 2) The detection process of audio feature points must consider the complicity of the algorithm. 3) When the audio receives severe attack, the accuracy of feature point detection needs to be improved. (4) Invariant watermark. The watermark is embedded into the stable quantity of audio signals (it still keeps invariant quantities although it receives de-synchronization attack), which makes audio watermark have the capacity of anti-synchronization attack.

The quality of audio watermarking algorithm of anti-synchronization attack depends on if it can locate the embedding position of the watermark accurately. The paper proposes an anti-synchronization attack audio watermarking algorithm based on the combination of implicit and explicit synchronization. Firstly, the implicit synchronization technique based on synchronizing signal is used to rapidly locate the embedding position of the possible watermarks effectively. Then, the explicit synchronization technique embedding synchronizing signal is used to screen the sequence of feature points and rectify position displacement, for implementing accurate positioning. Next, the existing quantization algorithms receive theoretical analysis and simulation experiment, and the paper gets a watermarking quantization embedding algorithm based on absolute mean. In addition, for de-synchronization attack such as TSM, the paper proposes a TSM parameter estimation method, which recovers the audio according to the estimated parameters and implements the synchronization of signal embedding and detection. Lastly, the simulation experiment tests the watermark robustness of the audios under various attacks, and achieves the experimental data.

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II. DUAL-SYNCHRONOUS MECHANISM

A. Implicit synchronization

In audio watermark, the audio generally includes massive rhythms and beats which are important information for auditory perception of the people. They correspond to the sudden promotion and demotion of tones, transition of mixing music and the sound of percussion instrument, and have close relationship with rapid change of local energy of audio frequency. In order to ensure that the music quality has no mistakes, the important information is commonly retained in making process of music. Implicit synchronization uses the characteristic of local area of music stably processing audio signals, and uses the areas as the embedding and detection synchronization point of watermark, for achieving self-synchronization effect of watermarks. We can see that the key of implicit synchronization is how to accurately find out the local area for watermark embedding and detection.

B. Explicit synchronization

The paper selects M sequence as synchronizing signal. $\{a_n\}$ and $\{b_n\}$ are two M sequences with the same period p , $\{a_n\},\{b_n\}\in\{-1,1\}$, and the definition of the functions is

$$\rho_{a,b}(\tau)=\frac{1}{p}\sum_{n=1}^{p}a_n b_{n-\tau}.$$

The self-correlations functions of M sequence $\{a_n\}$ have the following features,

$$\rho_{a,a}(\tau)=\begin{cases}1, & \tau=0\\ -1/p, & \tau\neq 0\end{cases}.$$

$\{a_n\}$ is the initial M sequence of synchronizing signal, and $\{b_n\}$ is a sequence to be detected, if

$$\rho_{a,b}(0)\geq T/p.$$

We consider that $\{b_n\}$ is a synchronizing signal. And synchronizing signals are embedded into DWT domain of high energy region after feature points.

C. Dual-synchronous positioning

The embedding of synchronizing signals and watermark information, and the process of audio signals makes the audio receive different damages, which must influence the accuracy of extracting synchronizing signals based on audio features, which makes the position of feature point detection displacement. Literature [4] indicates that there are only 63.6% of feature points which can be accurately extracted. So implicit synchronization technique based on audio feature only can approximately locate the embedding position of possible watermarks. In fact, under the condition that audio is not damaged any more, the position displacement amplitude of feature points is very small. And explicit synchronization technique can be used to screen near possible feature points, for achieving the accurate positioning of embedding area of watermark. The advantage of combining implicit and explicit synchronization technique is that implicit synchronization technique reduces the blindness and calculation of searching for synchronizing signals, and explicit synchronization reduces the false detection rate of implicit synchronization.

III. WATERMARKING QUANTIZATION EMBEDDING ALGORITHM

A. Theoretical analysis of quantization scheme

(1) Single-valued quantization scheme 1 (odd-even quantization)

Watermark embedding. According to the quantifying step size Δ_1, the coordinate axis is segmented into odd-even interval set, A and B .

$$\begin{cases}A_k=\{2k\Delta_1<f_i\leq(2k+1)\Delta_1\\ B_k=\{(2k+1)\Delta_1<f_i\leq(2k+2)\Delta_1\end{cases},\quad k\in z.$$

When $w=1$, f_i is quantified to be the midpoint $\hat{f}_i$ of the closest interval A. When $w=-1$, is quantified to be the midpoint $\hat{f}_i$ of the closest interval B .

Watermark extraction：

$$w_i=\begin{cases}1 & if\ \hat{f}_i\in A\\ -1 & if\ \hat{f}_i\in B\end{cases}.$$

The maximal error of frequency coefficients caused by quantification is $\left|\hat{f}-f\right|\leq\Delta_1$, and the signal-to-noise ratio is

$$Snr=10\lg_{10}\frac{\|x\|^2}{\|\hat{x}-x\|^2}=10\lg_{10}\frac{\|f\|^2}{\|\hat{f}-f\|^2}\propto 10\lg_{10}\frac{\|f\|^2}{m\cdot\Delta_1^2}.$$

For watermark extraction, if audio signal process and attack makes that the variation of frequency coefficient value greater than $\Delta_1/2$, watermark can't detect correctly. If the influence of audio signal process and attack on frequency similar coefficient obeys normal distribution $N(0,\sigma^2)$, the error rate of watermark is represented as

$$Ber=p(|x|\geq\frac{\Delta_1}{2})=\int_{|x|\geq\frac{\Delta_1}{2}}\frac{1}{\sqrt{2\pi}\sigma}\cdot e^{-\frac{x^2}{2\sigma^2}}dx$$

$$=\int_{|y|\geq\frac{\Delta_1}{2\sigma}}\frac{1}{\sqrt{2\pi}}\cdot e^{-\frac{y^2}{2}}dy,\ \{y=\frac{x}{\sigma}\}$$

$$=2(1-\Phi(\frac{\Delta_1}{2\sigma})),\ \{\Phi(x)=\int_{-\infty}^{x}\frac{1}{\sqrt{2\pi}}\cdot e^{-\frac{x^2}{2}}dx\}.$$

(2)Single-valued quantization scheme 2

Watermark embedding

$$\hat{f}_i = \begin{cases} f_i - f_i\%\Delta_2 + 3\Delta_2/4 & if \quad w_i = 1 \\ f_i - f_i\%\Delta_2 + \Delta_2/4 & if \quad w_i = -1 \end{cases} \%$$

is MOD function, and Δ_2 is quantifying step size.

Watermark extraction：

$$w_i = \begin{cases} 1 & if \;\; \hat{f}_i\%\Delta_2 > \Delta_2/2 \\ -1 & if \;\; \hat{f}_i\%\Delta_2 < \Delta_2/2 \end{cases}.$$

The maximal error of frequency coefficients caused by quantification is $\left|\hat{f} - f\right| \le 3\Delta_2/4$, and the signal-to-noise ratio is

$$Snr \propto 10\lg_{10}\frac{\|f\|^2}{m\cdot(3\Delta_2/4)^2} = 10\lg_{10}\frac{\|f\|^2}{m\cdot\Delta_1^{\;2}}\{\Delta_1 = 3\Delta_2/4\}$$

For watermark extraction, if audio signal process and attack makes that the variation of frequency coefficient value greater than $\Delta_2/4$，watermark can't detect correctly. The error rate of watermark is represented as

$$Ber = p(|x| \ge \frac{\Delta_2}{4}) = 2(1-\Phi(\frac{\Delta_2}{4\sigma})) = 2(1-\Phi(\frac{\Delta_1}{3\sigma})), \{\Delta_1 = 3\Delta_2/4\} \tag{3}$$

Single-valued quantization scheme 3

Watermark embedding: $R_i = \lfloor f_i/\Delta_3 + 0.5 \rfloor$， Δ_3 is quantifying step size.

$$\hat{f}_i = \begin{cases} R_i \times \Delta_3 & if \; R_i\%2 = w_i \\ (R_i+1)\times\Delta_3 \; if \; R_i\%2 \ne w_i \;\&R_i = \lfloor f_i/\Delta_3 \rfloor \\ (R_i-1)\times\Delta_3 \; if \; R_i\%2 \ne w_i \;\&R_i \ne \lfloor f_i/\Delta_3 \rfloor \end{cases} \%$$

is MOD function, and $\lfloor\cdot\rfloor$ is downward integral function.

Watermark extraction:

$$w_i = \lfloor \hat{f}_i/\Delta_3 + 0.5 \rfloor\%2 \; .$$

The maximal error of frequency coefficients caused by quantification is $\left|\hat{f} - f\right| \le \Delta_3$, and the signal-to-noise ratio is

$$Snr \propto 10\lg_{10}\frac{\|f\|^2}{m\cdot\Delta_3^{\;2}} = 10\lg_{10}\frac{\|f\|^2}{m\cdot\Delta_1^{\;2}}\{\Delta_1 = \Delta_2\}$$

For watermark extraction, if audio signal process and attack makes that the variation of frequency coefficient value greater than $\Delta_3/2$，watermark can't detect correctly. The error rate of watermark is represented as

$$Ber = p(|x| \ge \frac{\Delta_3}{2}) = 2(1-\Phi(\frac{\Delta_3}{2\sigma})) = 2(1-\Phi(\frac{\Delta_1}{2\sigma})), \{\Delta_1 = \Delta_3\} \tag{4}$$

Mean quantification scheme 4

Watermark embedding. Firstly, f is segmented into m sections. $M_i = \lfloor F_i/\Delta_4 + 0.5 \rfloor$, in which $F_i = \frac{1}{n}\sum_{k=0}^{n-1} f(k + i\cdot n)$, $i = 1, 2\cdots m$, and Δ_4 is the quantifying step size.

$$\hat{F}_i = \begin{cases} M_i \times \Delta_4 & if \; M_i\%2 = w_i \\ (M_i+1)\times\Delta_4 \; if \; M_i\%2 \ne w_i \;\&M_i = \lfloor F_i/\Delta_4 \rfloor \\ (M_i-1)\times\Delta_4 \; if \; M_i\%2 \ne w_i \;\&M_i \ne \lfloor F_i/\Delta_4 \rfloor \end{cases}.$$

Watermark extraction：

$$w_i = \lfloor \hat{F}_i/\Delta_4 + 0.5 \rfloor\%2 \; .$$

B. Experimental test

The experiment tests the robustness with the length of time of 18.8s, the sampling frequency of 44.1kHz, the quantization accuracy of 16bit of mono wave under the platform of matlab7.0 and goldwave5.52. M sequence with the cycle as 31 is synchronizing signal, and the watermark information is $m = 200$ and $n = 20$. Under dual-synchronizing detection technique, the watermarking robustness of the extracted watermarks being quantified into embedding scheme is as follows.

TABLE 1. COMPARISON OF SNR FOR DIFFERENT WATERMARK QUANTIZATION EMBEDDING ALGORITHMS (DB)

	Literature 8	Literature9	Literature10
Δ	0.15	0.23	0.15
SNR	32.10	32.14	32.30

	Literature11	Literature12	The paper
Δ	0.0335	0.15	0.6
SNR	32.49	32.14	32.12

Table 1 shows the relationship of the quantifying step size Δ of quantization schemes under the condition of ensuring that SNR is the same. The quantifying step size of odd-even scheme in literature [8] and that in literature [11] should meet $0.15 \approx 0.0335\cdot\sqrt{20}$.

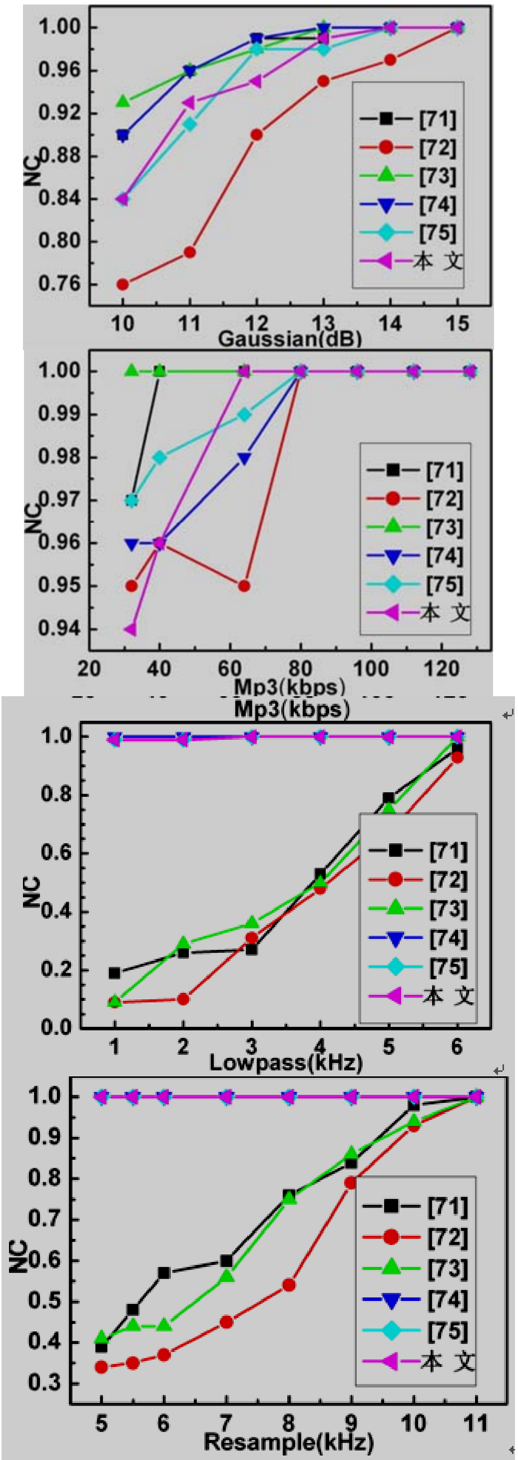

Figure 1. Comparison of robustness test results of quantization algorithms

Figure 1 is the comparison of robustness test results of quantization embedding algorithms under dual synchronous detection technique after audio signals receive the attacks including Guassian noise, LPS (6 orders), resampling and MP3 compression. From the figure, we can see that under the premise of embedding the watermark information and ensuring audio quality, for noise adding and Mp3 compression attack, the robustness of the other quantization algorithms is effective except for literature [9]. For LPS and resampling attack, the robustness of multi-valued quantization scheme is evidently greater than that of single-valued quantization scheme. In addition, when the audio signals receive the attacks of re-quantization and random cutting, the correlation coefficient of watermark detection is 1.0. And the watermarks can't be normally extracted after receiving de-synchronization attack such as TSM, under-sampling and spring attack (Ber>20%).

C. Quantization scheme of absolute mean

Based on theoretical analysis and experimental results of quantization embedding scheme, in order to improve the robustness of the algorithm for attacks, the number n of wavelet coefficient for embedding a watermark information should be increased. With the increase of n, the mean of wavelet domain tends to 0, $F_i \xrightarrow{n} 0$, which is not good for quantization embedding. The paper proposes a multi-valued quanitization scheme based on absolute mean.

$$Fabs_i = \frac{1}{n}\sum_{k=0}^{n-1} \left| f(k+i\cdot n) \right|, i = 1, 2\cdots m,$$

and $Fabs_i$ is quantified to be $Fabs_i^*$ according to odd-even quantification principle, and the wavelet coefficient $\hat{f}_{i,k} = f_{i,k}\cdot\frac{Fabs_i^*}{Fabs_i}$ is modified according to proportions, for implementing subsection self-adaptive embedding watermark.

The error of attacks for wavelet coefficients is represented as $f_i^* = f_i + \delta_i$. And the influence on detection indicator:

$Fabs$ is $Fabs^* \approx Fabs + \frac{1}{n}\sum_{i=1}^{n}\left|\delta_i\right|$.

If $\delta_i \sim N(0,\sigma^2)$, $Var(\frac{1}{n}\sum_{i=1}^{n}\left|\delta_i\right|) = (1-\frac{2}{\pi})\frac{\sigma^2}{n}$, the proving process is as follows.

Theorem: $x_1, x_2, \cdots, x_n$ is the subsample of the overall $x \sim N(0,\sigma^2)$, $E(x) = 0$, $D(x) = \sigma^2$, and the definition is $d = \frac{1}{n}\sum_{i=1}^{n}\left|x_i\right|$. Proving: $D(d) = (1-\frac{2}{\pi})\frac{\sigma^2}{n}$.

Prove: $E(|x|) = \int_{-\infty}^{\infty}|x|\cdot\frac{1}{\sqrt{2\pi}\sigma}e^{-\frac{x^2}{2\sigma^2}}dx = 2\int_0^{\infty}\frac{x}{\sqrt{2\pi}\sigma}e^{-\frac{x^2}{2\sigma^2}}dx$

$= \frac{2}{\sqrt{2\pi}\sigma}\int_0^{\infty} x\cdot e^{-\frac{x^2}{2\sigma^2}}dx = \frac{2\sigma}{\sqrt{2\pi}}\int_0^{\infty} y\cdot e^{-\frac{y^2}{2}}dy, (x = \sigma y)$

$$= \frac{2\sigma}{\sqrt{2\pi}}(-e^{-\frac{y^2}{2}}\Big|_0^\infty + \int_0^\infty e^{-\frac{y^2}{2}} dy) = \sqrt{\frac{2}{\pi}}\sigma .$$

$$E(|x|^2) = E(x^2) = D(x) + E^2(x) = \sigma^2 ,$$

so $D(|x|) = E(|x|^2) - E^2(|x|) = (1-\frac{2}{\pi})\sigma^2$.

$$D(d) = D(\frac{1}{n}\sum_{i=1}^{n}|x_i|) = \frac{1}{n^2}\sum_{i=1}^{n}D(|x_i|) = (1-\frac{2}{\pi})\frac{\sigma^2}{n}$$

The conclusion is proved.

From the above formula, we can see that compared with mean quantization scheme, the quantization scheme in the paper not only can reduce the variance of attacks for watermark detection indexes, but also can improve the robustness of the algorithm.

IV. TSM PARAMETER ESTIMATION METHOD

It is the necessary condition for correct watermark detection to ensure the synchronism of watermark detection and watermark embedding. Audio signals have the scaling of time domain in the process of DA/AD and under-sampling, which is fatal for the synchronization of watermark detection and watermark embedding. The paper proposes a TSM parameter estimation method. According to the features of audio, TSM parameters to be attacked are estimated. Then, the audio signals are recovered to implement the synchronism of watermark detection and watermark embedding. The audio signals between two feature points are selected, and the absolute amplitude energy of the audio signal is selected as parameter estimation feature value. And TSM parameter estimation algorithm is as follows.

If $f(x)$ is the audio signal of two feature points, and the number of sampling points is len. The voice frequency of len/a after time domain scaling is $f^*(x) = f(x/a)$. If $a > 1$, it means audio compression. If $0 < a < 1$, it means audio extension. The characteristic quantity of TSM parameter estimation of audio signal $f(x)$ is represented as

$$Tp = \int_0^{len} |f(x)| dx .$$

The characteristic quantity Tp^* of TSM parameter estimation of $f(x)$ is

$$Tp^* = \int_0^{len/a} |f^*(x)| dx = \frac{1}{a}\int_0^{len} |f^*(t/a)| dt \quad (x = t/a)$$

$$= \frac{1}{a}\int_0^{len} |f(x)| dx \quad (f^*(x) = f(x/a)) = \frac{1}{a}Tp .$$

We can get $a = Tp/Tp^*$. The characteristic quantity Tp^* of TSM parameter estimation of $f(x)$ can be used as the secret key for watermark detection.

Figure 2 is the bi-logarithmic graph with TSM parameter a as the abscissa and with the characteristic value Tp^* of TSM parameter estimation as the vertical coordinate. From the figure, we can know that $\log_{10} Tp^*$ and $\log_{10} a$ meets the linear relationship that the slope is negative, and we can get $a \propto 1/Tp^*$, which verifies the correctness of the above formula. Figure 3 is TSM parameter estimated by the algorithm. The actual value of TSM parameter a is the abscissa, and the estimation value of TSM parameter a is the vertical coordinate. The experimental results are very perfect, which verifies the feasibility of the algorithm.

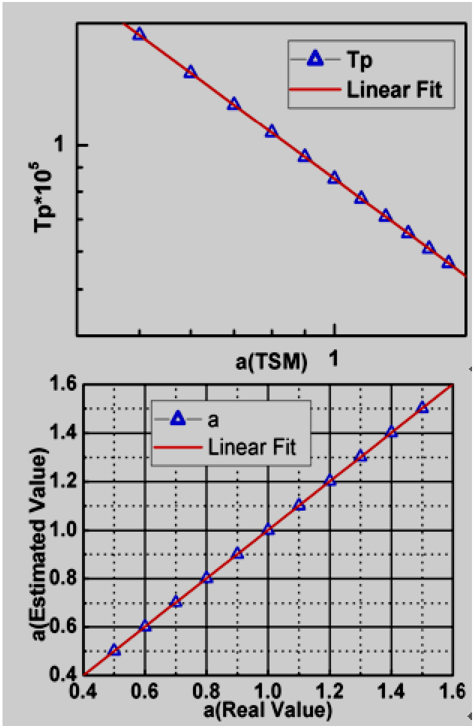

Figure 2. Characteristic quantity of TSM parameter estimation Figure 3 TSM parameter estimation

V. ALGORITHM DESIGN AND EXPERIMENT SIMULATION

A. *Algorithm design*

Watermark embedding. Firstly, implicit synchronization technique based on audio feature is used for feature point detection. In the high energy region after each feature point, the synchronizing signals and watermark information is embedded into wavelet coefficients according to absolute mean quantization scheme.

Watermark extraction.

(1)After the audio signals to be detected receive implicit and explicit synchronization detection, we can get the correctional feature sequence $\hat{p}$.

(2) If the characteristic quantity of parameter estimation of feature point sequence $\hat{p}$ is $Tp^* \neq Tp$, TSM parameter estimation algorithm in the above chapter is used for audio parameter estimation. According to the parameter estimation value, the audio is recovered, and the watermark is detected again.

(3) In wavelet domain of the high energy region after each feature point of the sequence $\hat{p}$, the detection indicator $Fabs$ is computed. And the watermark information:

$$w_i = 2*(\lfloor Fabs_i / \Delta + 0.5 \rfloor \%2) - 1$$

is extracted.

B. *Experimental simulation*

Under matlab7.0 and audio processing software Goldwave v5.52, the paper uses the mono wave audio with the time length of 18.8s, the sampling frequency of 44.1kHz and the quantification accuracy of 16bit for simulation experiment. In the experiment, M sequence with the cycle of 31 is used as synchronizing signal, and the used parameters are $L = 2000$, $T_1 = 5$, $T_2 = 400$, $T = 20$, M sequence $l = 31$, watermark sequence $m = 100$ and $n = 24$. The paper uses self-adaptive embedding synchronizing code and watermark information, which ensures that SNR of each audio section greater than 20dB.

When the audio receives conventional audio signal process such as re-quantification, resampling, Gaussian noise, Mp3 compression, LPF (6 orders), random cutting, inversion and echo, the watermark can be accurately detected out. The experimental results are shown in Table (1).

TABLE 2. ROBUSTNESS TEST OF WATERMARK OF AUDIO SIGNAL UNDER CONVENTIONAL ATTACKS

Attack Types	Ber (%)	Attack Types	Ber (%)
Gaussian(10dB)	0	Invert	0
Re-quantization	0	Resample	0
Cropping (10%)	0	Cropping (30%)	0
Mp3(128kbps)	0	Mp3(80kbps)	0
Mp3(40kbps)	0	Mp3(32kbps)	0

Attack Types	Ber (%)
Low-pass(4k)	0
Low-pass(1k)	0
Cropping (50%)	0
Mp3(80kbps)	0
Echo(0.5s)	0

Although audio signals receive the attacks of TSM ($\pm 1\%$), under-sampling (Un-sample $\pm 1\%$) and Jittering attack (Jittering 1/100) and doesn't recover audio, the watermark detection fails (Ber>20%). After using TSM parameter estimation algorithm to recover audio signals, the error rate of watermark is less than 10%. The experimental results are shown in Table (2).

TABLE 3. ROBUSTNESS TEST OF WATERMARK OF AUDIO SIGNAL UNDER DE-SYNCHRONIZATION ATTACKS

Attack Types	Ber (%)	Attack Types	Ber (%)
TSM (-30%)	10	TSM (30%)	8
TSM (-25%)	8	TSM (25%)	8
TSM (-20%)	7	TSM (20%)	8
TSM (-15%)	8	TSM (15%)	9
TSM (-10%)	8	TSM (10%)	8
TSM (-5%)	8	TSM (5%)	8

Attack Types	Ber (%)
Jittering(1/50)	0
Jittering(1/10)	0
Un-sample (10%)	0
Un-sample (-10%)	0
Un-sample (30%)	0
Un-sample (-30%)	0

The literature [8~11] is only effective for common attacks and random cutting. But the algorithm in the paper not only is robust for conventional audio signal process such as re-quantification, resampling, Gaussian noise, Mp3 compression, LPF (6 orders), random cutting, inversion and echo, but also can effectively resist de-synchronization attacks including 30% of TSM and 30% of under-sampling, and 1/50 jittering attack, which is greater than 60bits/20 in literature [7].

VI. CONCLUSION

The paper proposes a anti-synchronization audio watermark algorithm based on absolute mean quantization and parameter estimation. Firstly, the paper uses double synchronization technique combining implicit synchronization and explicit synchronization to implement accurate positioning of feature points. The advantage of double synchronization technique is that implicit synchronization reduces the blindness and computation of explicit synchronization searching for synchronizing signal, and explicit synchronization reduces the false detecting rate of implicit synchronization. Then, based on theoretical analysis and experimental results of quantification scheme, the paper proposes a multi-valued quantification scheme based on absolute value. Lastly, for the de-synchronization attacks of time scaling, under-sampling and jittering attack, the paper proposes a TSM parameter estimation algorithm for audio recovery to implement synchronization of watermark embedding and watermark detection. Simulation experiment indicates that the algorithm in the paper not only can resist conventional attacks, but also is robust for de-synchronization attacks including time scaling, under-sampling and jittering attack.

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