An Audio Blind Watermarking Scheme Based on DWT-SVD

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***Abstract*— In order to protect the digital audio and video products copyright in the network, an improved audio blind watermarking algorithm scheme based on discrete wavelet transform (DWT) and singular value decomposition (SVD) is proposed. In the algorithm, an original audio is split as blocks and each block is decomposed on discrete wavelet transform for two degree, then first quarter audio approximate sub-band coefficients are decomposed on SVD transform, obtain a diagonal matrix. The watermarking information is embedded into the diagonal matrix. Experiments display that the transparency of the proposed algorithm is better, and robustness is strong against the popular audio signal attack such as resampling, low-pass filter, requantization, Gaussian white noise, MP3 compression and popular audio signal attack method has stronger robustness, average normalized correlation coefficient NC > 0.950, average BER<0.048.**

***Index Terms*—audio blind watermarking; discrete wavelet transform; singular value decomposition**

I. INTRODUCTION

With the development of Internet technology and digital multimedia technology, large amounts of digital media in various forms are transmitted. It is increasingly serious that the digital media has suffered from violation illegally in the network. In order to effectively protect the copyright of multimedia information and owners, issuers and users the legitimate rights and interests, many researchers pay more attention to copyright management and protection nowadays. Embedding a certain form of watermark into multimedia data is considered as a potential solution.

Compared with digital image watermark, audio watermark has less sampling points in each interval and one of the auditory system is more sensitive than the human visual system. Thus digital audio watermark research is of challenge[1]. At present, digital image watermark[2-3] is researched mostly and the algorithm is more mature. However, audio digital watermark is researched rarely.

Digital watermark is classified into two categories. One is in the temporal domain; the other is in the transform domain. From the view of the performance of watermarks against attacks, the performance of the transform domain methods are commonly considered better than that of the time domain methods. The transform domain algorithm include Discrete Fourier Transform (DFT), Discrete Cosine Transform (DCT) and Discrete Wavelet Transform (DWT), etc.

Wang R [9], puts forward an audio watermarking algorithm based on DWT. In the algorithm, the watermarking information is embedded in the low frequency coefficient after audio signal discrete wavelet transform, embedded watermark was extracted by linear prediction method, this algorithm is a blind watermarking algorithm. Hamza O [10], proposed based on SVD audio watermarking algorithm. Firstly, the original audio file does short time Fourier transform. Then by the singular value decomposition of this sparse matrix, get embedded matrix, embed watermark information in this algorithm. Robustness of this algorithm is better than Cox method. Zezula R[11] presented a kind of SVD complex modulation overlap transform audio watermarking algorithm, but the algorithm involves less attacks type for audio signal. Chen YinQiu, Wu XiangSheng [12] proposed audio watermark algorithm based on SVD and DWT. However, original audio signal was not split as blocks. Wavelet transform was presented, then carries out SVD processing for approximate coefficient to obtain S matrix, image watermark was embedded in S matrix. But it needs to be improved for robustness. Ali A H [13] proposed

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audio watermarking algorithm based on DWT and SVD. Four stages wavelet transform was presented to obtain cA4, cD4, cD3, cD2 and cD1. Carry out SVD processing to the matrix that is constituted of cD4, cD3, cD2 and cD1. The watermark information is embedded in the matrix. But it was a non-blind watermarking algorithm.

An audio blind watermarking based on Discrete Wavelet Transform and singular value decomposition is proposed in this paper. First, the audio signal is divided into sections. Each section audio is decomposed on discrete wavelet transform for two degree. Then each approximate component is singular value decomposed. Last the watermark image is embedded into the relative singular values chosen. In the experiment, we adopt re-sampling, low-pass filtering, re-quantization, Gaussian white noise, MP3 compression and popular attack means of audio signal processing. The results show that the algorithm is transparency and robustness is better.

II. BACKGROUND

A. Discrete Wavelet Transform

Wavelet Transform[14] provides a time-frequency representation of the signal. It was developed to overcome the short coming of the Short Time Fourier Transform, which can also be used to analyze non-stationary signals. While STFT gives a constant resolution at all frequencies, the Wavelet Transform uses multi-resolution technique by which different frequencies are analyzed with different resolutions. The Discrete Wavelet Transform (DWT), which is based on sub-band coding, is found to yield a fast computation of Wavelet Transform. It is higher frequency resolution and lower temporal resolution in the low frequency part and higher temporal resolution lower frequency resolution in the high frequency part.That is to say, Low- frequency signals change slowly and high-frequency signals change rapidly, which is likely to implement and reduces the computation time and resources required.

Continuous wavelet transform is defined as shown formula (1)

$$CWT_x(\tau,a)=\frac{1}{\sqrt{|a|}}\int x(t)h^*\left(\frac{t-\tau}{a}\right)dt \qquad (1)$$

where $h(t)$ is Mother Wavelet. $x(t)$ is the entire signal. τ is shift factor that can make wavelet to shift along timeline to analyse signal. $\frac{1}{\sqrt{|a|}}h\left(\frac{t-\tau}{a}\right)$ expresses wavelet stretching and translating. The following criteria must be met to Mother Wavelet.

$$h(t)\in L^2(R) \qquad (2)$$

$L^2(R)$ is energy limited signal space. When a increases, the entire signal $x(t)$ is observed with $h(t)$ stretching waveform. In contrast, when a reduces, the local signal of $x(t)$ is observed with $h(t)$ narrow waveform. Discrete wavelet transform is usually based on binary wavelet and is discreted on the power series.

After Discrete Wavelet Transform, audio signal's energy are invariable. Audio signals are decomposed into low frequency and high frequency by discrete wavelet transform for one degree. Low frequency component concentrate most of the energy of audio signal, which is the main part of the original audio signal. cA presents approximate component. High frequency component concentrate little of the energy of audio signal. cD presents detail component. Wavelet Basis and wavelet degrees can be selected according to the characteristics of the algorithm. Hence, digital watermarking has very flexible in design. Audio digital signal can be decomposed on multi-level discrete wavelet transform. It is audio level 2 discrete wavelet transform, as shown in Figure 1.

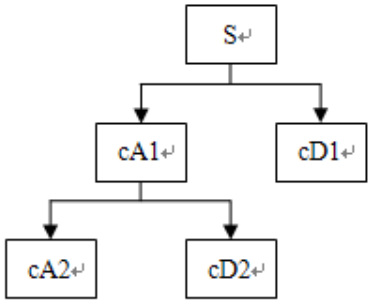

Figure 1. Audio level 2 Discrete Wavelet Transform

B. Singular Value Decomposition

In linear algebra, the Singular Value Decomposition[15] (SVD) is a factorization of a real or complex matrix, with many useful applications in signal processing and statistics. SVD is orthogonal transformation, which can diagonalize matrix. Singular Value Decomposition was used to process the image information hiding. In recent years the researchers have tried to use SVD to the field of audio digital watermark. Digital watermark algorithm based on Singular Value Decomposition is the robust for transposed, rotation, scaling and general geometric distortion.

If A is a $m\times n$ real matrix with $m>n$, then A can be written using a so-called singular value decomposition of the form

$$U^TAV=S \qquad (3)$$

$$USV^T=A \qquad (4)$$

where U is a $m\times m$ matrix, A is a $m\times n$ matrix, and V is a $n\times n$ matrix. U and V are orthogonal matrices,which are the field of real numbers.

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$U=[u_1,u_2,...,u_m]$. $V=[v_1,v_2,...,v_n]$. $S=diag$ ($\sigma_1,\sigma_2,\ldots,\sigma_p$), where σ is singular values of A, $\sigma_1\geq\sigma_2\geq\ldots,\geq\sigma_p$.

If A is a $m\times n$ complex matrix, then the singular value decomposition is defined as:

$$UAV^H=S \tag{5}$$

where U and V are unitary matrices, V^H is the conjugate transpose of V, and S is a diagonal matrix whose elements are the singular values of the original matrix.

For any real matrix A, there always exists such a decomposition with positive singular values U, V and S. U and V are orthogonal matrices, S is a diagonal matrix.

The singular value decomposition characteristics are as follows:

- When singular values have small changes, great changes have not taken place for the original matrix after inverse transformation.
- When a matrix values have small change, great changes have not taken place for singular values after singular value decomposition. In other words, great changes have not taken place for singular values after all kinds of common audio signal processing.

III. Blind Watermark Algorithm

A. Digtal Watermark Image Preprocessing

Binary watermark image $f(i,j)$ is embedded into audio signal in the algorithm.Suppose the image size is $n_1\times n_2$, watermark image can be represented as:

$$F=\{f(i,j),0\leq i<n_1,0\leq j<n_2\} \tag{6}$$

where $f(i,j)\in(0,1)$. Watermark images are two-dimensional and audio signal is one-dimensional, so researchers need to preprocess the image. Reduce the dimensions of the image as shown in equations (7).

$$F=\{f(i)=f(h_1,h_2)\\ 1\leq h_1\leq n_1,1\leq h\leq n_2,i=(h_1-1)\times n_2+h_2\} \tag{7}$$

B. Embedding algorithm

The watermark embedding process is illustrated in Figure 2.

Suppose A is the mono original audio signal. $A=\{A_1,A_2,...,A_t\}$. Binary image size $n_1\times n_2$. The mono original audio signal length is not less than $n_1^2\times n_2^2$.

Details of embedding are elaborated as following:

Step 1. Audio signal segmenting: The original audio signal A is split into many segments, which are denoted as A_i.

Step 2. 2-level DWT: 2-level DWT is performed on each segment. Choose "db1" as wavelet base of Discrete Wavelet Transform. Get the detail component and the approximation component of the audio signal after 2-leve DWT , which are denoted as cA_i2、cD_i2、cD_i1

Step 3. Selecting low frequency coefficients: In order to obtain good imperceptibility, selecting low frequency coefficients of each segment as the dataset for watermark embedding.

Step 4. SVD: Diagonal matrix S, also known as singular matrix, is obtained by the singular value decomposition of the approximation component cA_i2.

Step 5. Embedding watermark bit: Take out a value of diagonal matrix, a value of preprocessed binary image is embedded into diagonal matrix. Repeat step 2~step 4 until all values of watermark image are embedded into corresponding diagonal matrix. Lastly, Obtain a diagonal matrix that is embedded watermark image, is denoted as Sw.

If $\lfloor S(1,1)/S(2,2)\theta\rfloor$ is even number and watermark image bit is 1, then $Sw(1,1)=S(2,2)\theta(\lfloor S(1,1)/S(2,2)\theta+1\rfloor)$; watermark image bit is 0, $Sw(1,1)=S(2,2)\theta(\lfloor S(1,1)/S(2,2)\theta\rfloor)$.

If $\lfloor S(1,1)/S(2,2)\theta\rfloor$ is odd number and watermark image bit is 1, then $Sw(1,1)=S(2,2)\theta(\lfloor S(1,1)/S(2,2)\theta\rfloor)$; watermark image bit is 0, $Sw(1,1)=S(2,2)\theta(\lfloor S(1,1)/S(2,2)\theta+1\rfloor)$, θ=0.5, where $\lfloor\ \rfloor$ is the floor function.

Step 6. Inverse SVD: Diagonal matrix S' is obtained by the inverse singular value decomposition of embedded watermark diagonal matrix Sw.

Step 7. Modifying the corresponding coefficients: Change S' into one-dimensional vector. Replace step 2 approximate component with containing watermark values. It is denoted as cA_i2'.

Step 8. Inverse DWT: Obtain audio segments embedded watermark image by inverse DWT. cAi2' is the approximate component, cDi1 and cDi2 are the detail component.

Step 9. Fitting together: Combine each watermarked audio segment together to form the final watermarked audio signal.

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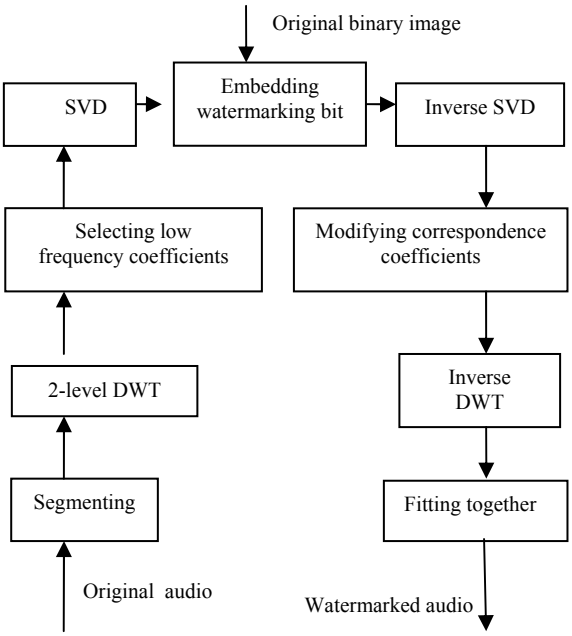

Figure 2. The watermark embedding algorithm flowchart

C. Extraction Algorithm

The extraction process does not require the original host audio signal. It is almost the reverse of the embedding process. The overall flowchart indicates in Figure 3.

Step 1. Watermarked audio signal segmenting: The watermarked audio signal is split into many segments, which are denoted as A_w.

Step 2. 2-level DWT: 2-level DWT is performed on each segment. Choose "db1" as wavelet base of Discrete Wavelet Transform. Get the detail component and the approximation component of the audio signal after 2-leve DWT, which are denoted as cAi2', cDi2', and cDi1'.

Step 3. Selecting low frequency coefficients: select low frequency coefficients of each segment as the dataset for watermark extracting.

Step 4. SVD: In order to retrieve watermark image, we need to get singular matrix by the singular value decomposition for the approximation component.

Step 5. Extracting watermark bit: Take out a value of diagonal matrix, a value of preprocessed binary image is embedded into diagonal matrix. Repeat step 2~step 4 until all values of watermark image are extracted from corresponding diagonal matrix. Lastly, obtain one-dimensional vector of embedded image.

Step 6. Into a two-dimensional image matrix: Turn one-dimensional vector extracted into two-dimensional matrix. Output extracted watermark image.

IV. Experimental Results and Discussions

In this section, some experimental results demonstrate the performance of the proposed DWT–SVD algorithm. In the experiment, binary image with size 32 × 32, displayed in Figure 4, is taken as the original watermark signal for the audio signals. Each audio signal is a 16-bit mono file in the WAVE format with the 44.1 kHz sampling rate. Three host audio signals including music, classic, and speech signals are used.

Figure 4. The original binary image

A. Performance Analysis

In this Section, we evaluate the performance of our proposed watermarking scheme. The watermark performance, such as embedding capacity, signal-to-noise ratio (SNR), The Mean Opinion Score(MOS), The normalized cross correlation (NC) and the bit error rate (BER) is investigated.

1) Embedding capacity

Suppose that the sampling rate of the audio signal is H_A (Hz), and the number of samples of each segment is N The embedding capacity P of the proposed scheme can be expressed as:

$$p = \frac{H_A}{N} \qquad (8)$$

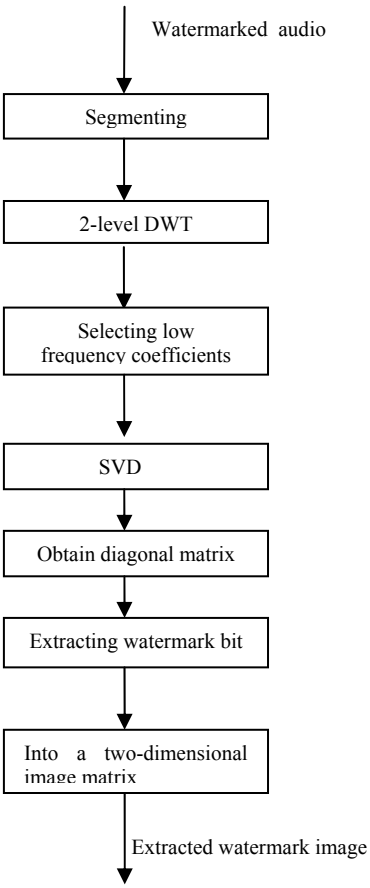

Figure 3. The watermark extracted algorithm flowchart

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where the unit of embedding capacity p is bit/s. The embedding capacity is improved as N decreases. However, a less N causes higher distortion. P=44100/1600=27.56 bit/s.

2) Perceptual transparency

The main requirement for watermarking is perceptual transparency. The embedding process should not introduce any perceptible artifacts. Namely, the watermark should not affect the quality of the original signal.

Generally, there are two methods to detect transparency of watermarked audio. One is subjective, The Mean Opinion Score (MOS). It depends mostly on the listener's familiarity with artifacts. Due to the subjectivity, such opinions should be considered only as rough estimates.

MOS is reliable but the listening test is time-consuming. subjective evaluation standard are shown in TABLE Ⅰ. The MOS experiment results are obtained as

TABLE Ⅱ
SNR VALUE AND MOS VALUE

Audio Type	SNR	MOS
Speech	20.7	4.4
Classic	20.0	4.6
popular	22.4	4.7

shown in TABLE Ⅱ.

On the other hand, it is objective quality assessment methods, such as signal-to-noise ratio (SNR). SNR is a statistical difference metric which is used to measure the perceptual similarity between the undistorted original and the distorted watermarked audio signal. The following signal-to-noise radio (SNR) equation is used:

$$SNR = 10\log_{10}\frac{\sum_{n=1}^{N} Y^2(n)}{\sum_{n=1}^{N}[Y(n)-Y'(n)]^2} \quad (9)$$

where $Y(n)$ and $Y'(n)$ are original audio signal and watermarked audio signal respectively. In this experiment, SNR is shown TABLE Ⅱ.

3) Robustness

For all watermarking applications, robustness is one of the major algorithm design issues because it determines the algorithm behavior towards data distortions introduced through standard and malicious data processing. The watermark should be robust in common signal processing, including digital-to-analog and analog-to-digital conversion, linear and nonlinear filtering, compression, and scaling etc.

Robustness and transparency are interrelated and interact. Thus, there must be a trade-off between perceptual transparency and robustness. This problem can be solved by applying human perceptual modeling in the watermark embedding process. The normalized cross correlation (NC) is adopted to appraise the similarity between the extracted watermark and the original one.

TABLE Ⅰ
MOS EVALUATION STANDARD

Grade	Audio quality	Descriptions
5	Excellent	Imperceptible
4	Good	Perceptible, but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

It's definition is

$$NC(W,W') = \frac{\sum_{i=1}^{n_1}\sum_{j=1}^{n_2} W(i,j)W'(i,j)}{\sqrt{\sum_{i=1}^{n_1}\sum_{j=1}^{n_2} W(i,j)^2} \times \sqrt{\sum_{i=1}^{n_1}\sum_{j=1}^{n_2} W'(i,j)^2}} \quad (10)$$

where W and W' are the original watermark image and extracted watermark image. If the NC exceeds a certain threshold, we can conclude that this audio signal is protected, otherwise is not protected.

In addition, reliability was measured by the bit error rate (BER) of extracted watermark. It is defined as

$$BER = \frac{E}{n_1 \times n_2} \times 100\% \quad (11)$$

where E is the number of erroneously detected bits. $n_1 \times n_2$ is original watermark image size.

B. Experimental Results

Popular signal processing operations may affect the host signal perceived quality and corrupt the watermark image embedded. In order to illustrate the robustness of our watermarking scheme, we implemented a great number of popular attacks on watermarked audio signals. Some attacks are performed using MATLAB 7.5 and GoldWave 5.55. They are popular tool-sets for professional audio processing and editing. The common signal processing attacks are described as follows:

- Down-sampling: As the original audio signal is sampled with a sampling rate of 44.1kHz, thus the watermarked audio signal is down-sampled to 22.05kHz, and then up-sampled back to 44.1kHz.
- Up-sampling: As the original audio signal is sampled with a sampling rate of 44.1kHz, thus the watermarked audio signal is down-sampled to 88.2kHz, and then up-sampled back to 44.1kHz.
- Gaussian noise: White Gaussian noise is added to the watermarked signal until the resulting signal has a SNR of 20 dB.
- Low-pass filtering: The low-pass filter with the 11.025kHz cut off frequency is applied to watermarked audio signal.
- Denoising: The watermarked audio signal is denoised by using the ''Hiss removal'' function of GoldWave.

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- Re-quantization: 16-bit watermarked audio signal is quantized down to 8-bit/smaple and then back to16-bit bit/smaple.
- MP3 Compression: The coding/decoding is performed using a software implementation of the GoldWave with different bit rates (32 kbps, 64 kbps and128kbps) and then back to the WAVE format.
- Echo addition: An echo signal with a delay of 50ms and a decay of 10% is added to the watermarked audio signal.
- Reverse amplitude: Reverse the signs of the sample amplitudes.
- Amplitude variation: The watermarked signal is attenuated up to 120%.

C. Discussions

Watermark detection results against various popular signal processing attacks is shown in the TABLE Ⅲ, Ⅳ, Ⅴ. 0riginal watermark signal and the extracted one without any attack are same, NC=1, BER=0, which verifies that the watermark can be extracted clearly without any attack.

A great variety of watermarking approaches have been presented in the literature, for example echo, spread spectrum, Least Significant Bit, frequency masking methods and SVD methods etc. As we known, SNR is merely a measure of the noise power relative to the signal power. It is not closely correlated to human perception as SNR does not take into account perceptual models. Therefore, in order to make an appropriate comparison with the traditional watermarking methods, SNR result s and MOS grades should be both selected.

From the experiment results, it can be seen that our proposed method is obviously better than reference [12]. Finally, it should be pointed out that the proposed algorithm satisfies the desired features of audio watermarking. Experiments show that the transparency of the proposed algorithm is better, and robustness is strong against the popular audio signal attack such as re-sampling, low-pass filter, re-quantization, Gaussian white noise, MP3 compression and popular audio signal attack method has stronger robustness.

V. CONCLUSIONS

In this paper, we propose a blind digital audio watermarking scheme based on DWT and SVD. The experimental results have illustrated the inaudible and robust nature of our watermarking scheme. The easy operational proposed scheme is practicable for audio data copyright protection. Despite the success of the proposed audio watermarking scheme, it also has a drawback, that is, the proposed scheme is not robust against random cropping and time scale modification. Therefore, future research will focus on overcoming these problems. Moreover, Redundant Discrete Wavelet Transform model may be adopted to improve the imperceptibility of our scheme.

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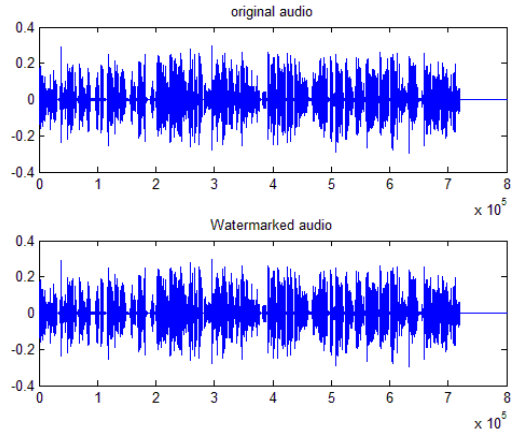

Figure 5. Speech audio signal

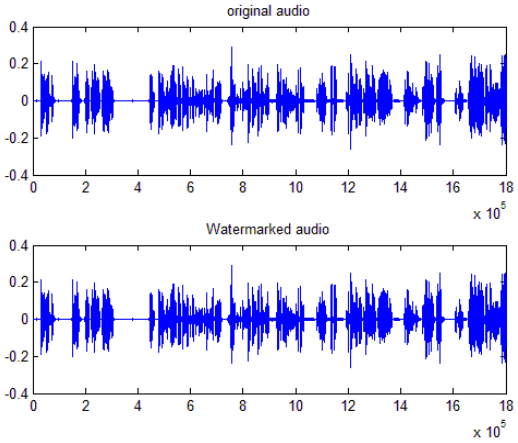

Figure 6. Classical audio signal

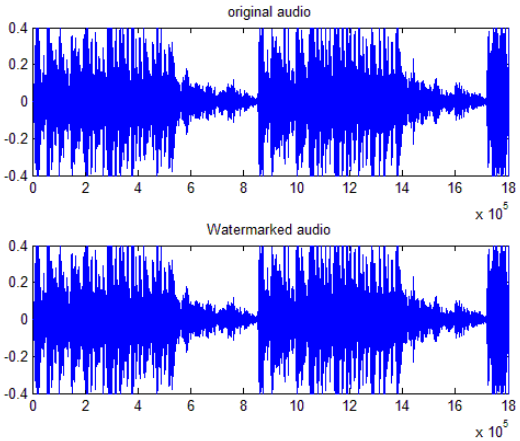

Figure 7. Popular audio signal

TABLE III
WATERMARK DETECTION RESULTS AGAINST VARIOUS COMMON SIGNAL PROCESSING ATTACKS(SPEECH)

Attack Type	Appearance		
	NC	BER	Watermark image
No attack	1	0	
Down-sampling (44.1KHz-22.05KHz -44.1KkHz)	1	0	
Up-sampling (44.1KHz-88.2KHz -44.1KHz)	1	0	
Additive White Gaussian noise(20db)	0.8603	0.1523	
Low-pass filtering (11.025KHz, Goldenwave)	0.9999	0.0059	
Denoising (Hiss removal ,Gold Wave)	0.8661	0.1377	
Re-quantization (16bit-8 bit to 16 bits/sample.)	0.7311	0.2676	
MP3 Compression (32kpbs)	1	0	
MP3 Compression (64kpbs)	1	0	
MP3 Compression (128kpbs)	1	0	
Echo addition（50 ms and a decay of 10%）	0.9430	0.0586	
Reverse amplitude	1	0	
Amplitude variation	1	0	

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TABLE Ⅳ
WATERMARK DETECTION RESULTS AGAINST VARIOUS COMMON SIGNAL PROCESSING ATTACKS(CLASSIC)

Attack	Appearance		
	NC	BER	Watermark image
No attack	1	0	
Down-sampling (44.1KHz-22.05KHz -44.1KkHz)	1	0	
Up-sampling (44.1KHz-88.2KHz -44.1KHz)	1	0	
Additive White Gaussian noise(20db)	1	0	
Low-pass filtering (11.025KHz, Goldenwave)	1	0	
Denoising (Hiss removal ,Gold Wave)	0.9972	0.0029	
Re-quantization (16bit-8 bit to 16 bits/sample.)	1	0	
MP3 Compression (32kpbs)	1	0	
MP3 Compression (64kpbs)	1	0	
MP3 Compression (128kpbs)	1	0	
Echo addition（50 ms and a decay of 10%）	1	0	
Reverse amplitude	1	0	
Amplitude variation	1	0	

TABLE Ⅴ
WATERMARK DETECTION RESULTS AGAINST VARIOUS COMMON SIGNAL PROCESSING ATTACKS(POP)

Attack	Appearance		
	NC	BER	Watermark image
No attack	1	0	
Down-sampling (44.1KHz-22.05KHz -44.1KkHz)	1	0	
Up-sampling (44.1KHz-88.2KHz -44.1KHz)	1	0	
Additive White Gaussian noise(20db)	1	0	
Low-pass filtering (11.025KHz, Goldenwave)	0.9915	0.0088	
Denoising (Hiss removal ,Gold Wave)	0.8579	0.1445	
Re-quantization (16bit-8 bit to 16 bits/sample.)	1	0	
MP3 Compression (32kpbs)	1	0	
MP3 Compression (64kpbs)	1	0	
MP3 Compression (128kpbs)	1	0	
Echo addition（50 ms and a decay of 10%）	0.9916	0.0088	
Reverse amplitude	1	0	
Amplitude variation	1	0	

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