A Novel Audio Watermarking Algorithm for Copyright Protection of Digital Audio

Jongwon Seok, Jinwoo Hong, and Jinwoong Kim

Digital watermark technology is now drawing attention as a new method of protecting digital content from unauthorized copying. This paper presents a novel audio watermarking algorithm to protect against unauthorized copying of digital audio. The proposed watermarking scheme includes a psychoacoustic model of MPEG audio coding to ensure that the watermarking does not affect the quality of the original sound. After embedding the watermark, our scheme extracts copyright information without access to the original signal by using a whitening procedure for linear prediction filtering before correlation. Experimental results show that our watermarking scheme is robust against common signal processing attacks and it introduces no audible distortion after watermark insertion.

Manuscript received Aug. 16, 2001; revised Feb. 20, 2002.

This work is supported by the Ministry of Information and Communication (MIC) of Korea.

Jongwon Seok (phone: +82 42 860 1306, e-mail: jwseok@etri.re.kr), Jinwoo Hong (e-mail: jwhong@etri.re.kr) are with Broadcasting Content Protection Research Team, ETRI, Daejeon, Korea.

Jinwoong Kim (e-mail: jwkim@video.etri.re.kr) is with Broadcasting Media Research Department, ETRI, Daejeon, Korea.

I. INTRODUCTION

Recent years have seen a rapid growth in the availability of multimedia content in digital form. A major problem faced by content providers and owners is protection of their material. They are concerned about copyright protection and other forms of abuse of their digital content. Unlike copies of analog tapes, copies of digital data are identical to the original and suffer no quality degradation, and there is no limit to the number of exact copies that can be made. In addition, digital equipment that can make digital copies is widely available and inexpensive.

One approach to content security uses cryptographic techniques, but those encryption systems do not completely solve the problem of unauthorized copying. All encrypted content needs to be decrypted before it can be used. Once encryption is removed, there is no way to prove the ownership or copyright of the content. As a solution to this problem, digital watermark technology is now drawing attention as a new method of protecting against unauthorized copying of digital content [1]-[5]. A digital watermark is a signal added to the original digital data (namely, audio, video, or image), which can later be extracted or detected. The watermark is intended to be permanently embedded into the digital data so that authorized users can easily access it. At the same time, the watermark should not degrade the quality of the digital data. In general, digital watermark techniques must satisfy the following requirements.

• Perceptual transparency

The main requirement for watermarking is perceptual transparency. The embedding process should not introduce any perceptible artifacts, that is, the watermark should not affect the quality of the original signal. However, for robustness, the watermark energy should be maximized under the constraint of keeping perceptual artifacts as low as possible. 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.

• Security

The watermark must be strongly resistant to unauthorized detection. It is also desirable that watermarks be difficult for an unauthorized agent to forge. Watermark security addresses the secrecy of the embedded information. Where secrecy is necessary, a secret key has to be used for the embedding and extraction process.

• 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.

• Watermark recovery without the original signal

In most applications, watermark extraction processes do not have to access to the original signal because of the large size of the data. This is called public or blind watermarking. In fact, this property is essential to real environments such as broadcasting and portable players. Except for some special cases, embedded information should be recovered without the original signal.

• Data rate

The watermarking scheme should have a data rate sufficient to support the copyright information. In most applications, copyright information consists of copy control information, usage information, and the serial number of the contents. To meet the requirement for copyright information, at least several tens of bits are needed.

Several digital watermark algorithms have been proposed, but most watermark algorithms focus on image and video. Only a few audio watermark algorithms have been reported.

Bender et al. [6] proposed several watermarking techniques, which include the following: spread-spectrum coding, which uses a direct sequence spread-spectrum method; echo coding, which employs multiple decaying echoes to place a peak in the cepstrum domain at a known location; and phase coding, which uses phase information as a data space. Unfortunately, these watermarking algorithms cause perceptible signal distortion and show low robustness. Furthermore, they have a relative high complexity in the detection process. Swanson et al. [7] presented an audio watermarking algorithm that exploits temporal and frequency masking by adding a perceptually shaped spread-spectrum sequence. However, the disadvantage of this scheme is that the original audio signal is needed in the watermark detection process. Neubauer et al. [8]-[10] introduced the bit stream watermarking concept. The basic idea of this method was to partly decode the input bitstream, add a perceptually hidden watermark in the frequency domain, and finally quantize and code the signal again. As an actual application, they embedded the watermarks directly into compressed MPEG-2 AAC bit streams so that the watermark could be detected in the decompressed audio data. Bassia et al. [11] presented a watermarking scheme in the time domain. They embedded a watermark in the time domain of a digital audio signal by slightly modifying the amplitude of each audio sample. The characteristics of this modification were determined both by the original signal and the copyright owner key. Solana Technology [12] proposed a watermark algorithm based on spread-spectrum coding using a linear predictive coding technique and fast Fourier transform (FFT) to determine the spectral shape. In the watermark detection process, the watermarked audio signal is whitened before correlation using the rake receiver.

This paper presents a novel audio watermarking scheme for copyright protection of digital audio. The watermark embedding scheme accomplishes perceptual transparency after watermark embedding by exploiting the masking effect of the human auditory system, as in [7], [13]-[15]. This approach can be applied to other signals, such as image and video. For images, this can be accomplished by adopting a human visual model [16]. In the detection procedure, by applying whitening, or de-correlation, before correlation, the proposed scheme does not need the original audio signal to extract the watermark information. Linnartz et al. [17] and Depovere et al. [18] used a similar approach in image watermarking. They applied the whitenning procedure in watermark detection to remove correlation in pixels using a lowpass filter. We achieved this by removing the audio spectrum in the watermarked audio using a linear prediction analysis, which is a frequently used technique in speech signal processing.

This paper is organized as follows. Section II introduces considerations for improving watermark detection performance in the correlation scheme. Section III presents the proposed watermarking scheme, including the embedding and detection process. We show the experimental results of the proposed algorithm in section IV and conclude the paper in section V.

II. SOME CONSIDERATIONS

Figure 1 shows the basic and simple watermarking embedding and detection process using the correlation scheme. As Fig. 1 illustrates, in principle, we can regard most correlation-based embedding processes as adding an additional signal to the original signal to obtain a watermarked signal. The watermark embedding process from Fig. 1 can be represented as

$$y(n) = x(n) + w(n), \tag{1}$$

where $y(n)$ is the watermarked audio signal, $x(n)$ the original signal, and $w(n)$ the watermark signal. Performing the watermark detection by correlation, the resulting decision value d will be

$$\begin{aligned} d &= d_x + d_w \\ &= \frac{1}{N}\sum_i x(i)w(i) + \frac{1}{N}\sum_i w^2(i). \end{aligned} \tag{2}$$

From (2), it is clear that d_w is the energy of the watermark $w(n)$, and the expected value of d_x is zero. Because of the central limit theorem, if N is sufficiently large and if the contributions of the sums are sufficiently independent, d_x has a Gaussian distribution. If we set the threshold for the watermark detection $T = d_x / 2$, the probability of a false negative P_- (the watermark is present, but the detector decides it is not) equals the probability of a false positive P_+ (the watermark is not present, but the detector decides it is):

$$P_+ = P_- = \frac{1}{2} erfc\left(\frac{d_w}{\sqrt{8}\sigma_x}\right), \tag{3}$$

where $erfc(\cdot)$ is the complementary error function. From (3), we may draw the following preliminary conditions to decrease the false positive and negative probabilities.

1. The false positive and negative probabilities decrease as the strength of watermark d_w to be embedded increases. However, the strength of the embedded watermark signal is limited and depends on the human perceptual characteristics of the audio signal.
2. The false positive and negative probabilities decrease as the standard deviation σ_x increases. This can be obtained by applying whitening, or de-correlation, before correlation in the detection procedure. Application of the whitening procedure should significantly remove the correlation in the signal.

III. WATERMARKING SCHEME

1. Watermark Embedding

Our watermark embedding scheme is based on a direct-sequence spread-spectrum (DSSS) method. The auxiliary information that is to be embedded is modulated by a pseudo-noise (PN) sequence. The spread-spectrum signal is then shaped in the frequency domain and inserted into the original audio signal. The embedding strength determines the energy of the watermark signal and can be considered as the energy of the noise added to the audio signal.

Our goal was to design the weighting function so that the watermark energy was maximized subject to a required maximal acceptable distortion. The strength of the embedded watermark signal depends on the human perceptual characteristics of the audio signal. We accomplished our goal with an embedding scheme that exploits the masking effect of the human auditory system. We intended to adapt the watermark so that

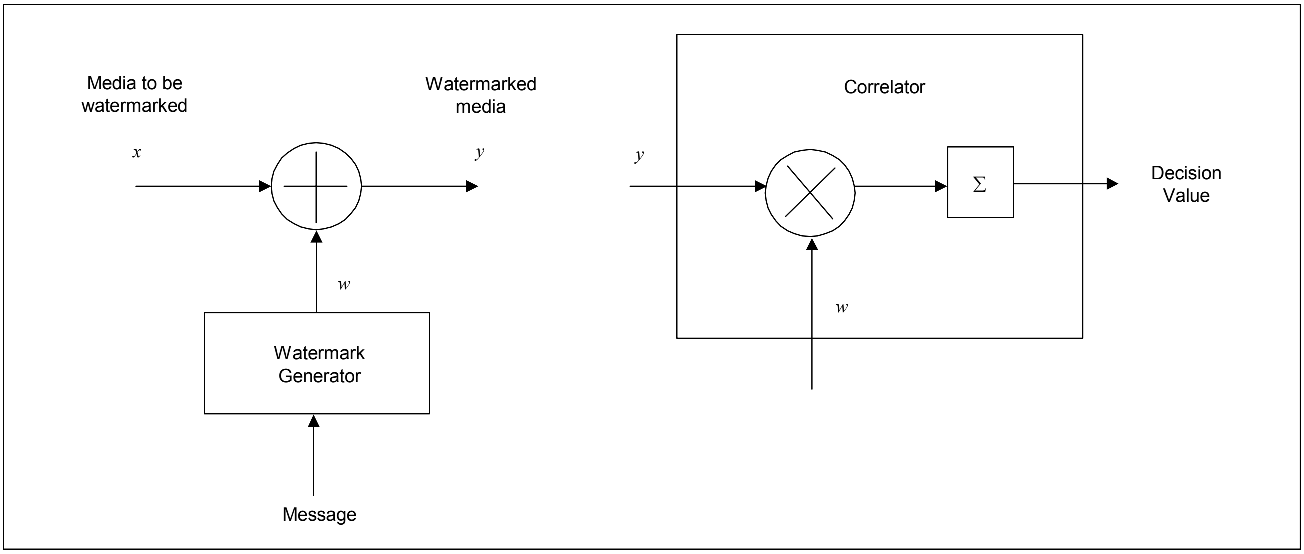

Fig. 1. Basic watermark embedding and detection.

the energy of the watermark was maximized and the auditory artifact was kept as low as possible. We used the masking model defined in the ISO-MPEG Audio Psychoacoustic Model [19]. The detailed procedure to obtain the Psychoacoustic Model is well described in [15], [19], [20].

The inner ear performs a short-time critical band analysis where the frequency-to-place transform occurs along the basilar membrane. The power spectra are not represented on a linear frequency scale but on limited frequency bands called critical bands. The auditory system can roughly be described as a band-pass filter bank. Simultaneous masking is a frequency domain phenomenon where a low-level signal can be made inaudible by a simultaneously occurring stronger signal. Such masking is largest in the critical band in which the masker is located, and it is effective to a lesser degree in neighboring bands. The masking threshold can be measured, and low-level signals below this threshold will not be audible. The final masking threshold curve obtained from the analysis audio segment is approximated with a 12th order all-pole filter. The filtering is then applied to the PN sequence using the filter coefficients obtained from the all-pole modeling to shape the watermark signal to be imperceptible in the frequency domain. This masking threshold is the limit for maximizing the watermark energy while keeping the perceptual audio quality.

In our embedding process, we repeatedly apply an embedding operation on short segments of the audio signal. Each one of these segments is called a frame. A diagram of the audio watermark embedding scheme is shown in Fig. 2. Using the concept of the Psychoacoustic model and DSSS method, our watermark embedding works as follows:

a) Calculate the masking threshold of the current analysis audio frame using the Psychoacoustic model with an analysis audio frame size of 1024 samples and a 1024 point FFT.
b) Generate the PN sequence with a length of 1024.
c) Perform the FFT to the copyright information, which is modulated by the PN sequence.
d) Using the masking threshold, shape the watermark signal to be imperceptible in the frequency domain.
e) Compute the inverse FFT of the shaped watermark signal.
f) Create the final watermarked audio signal by adding the watermark signal to the original audio signal in the time domain.

2. Watermark Detection

When designing a watermark detection system, we need to consider the desired performance and robustness of the system. The watermark should be able to be detected under common signal processing operations, such as digital-to-analog and analog-to-digital conversion, linear and nonlinear filtering, compression, and scaling. Furthermore, in most applications, watermark extraction processes do not have to access the original signal. In fact, not having access to the original signal is essential in real environments such as broadcasting.

The proposed watermark detection procedure does not require access to the original audio signal to detect the watermark signal. Most correlation detector schemes assume that the channel is white Gaussian; howerer, this is not the case for real audio signals because the audio samples are highly

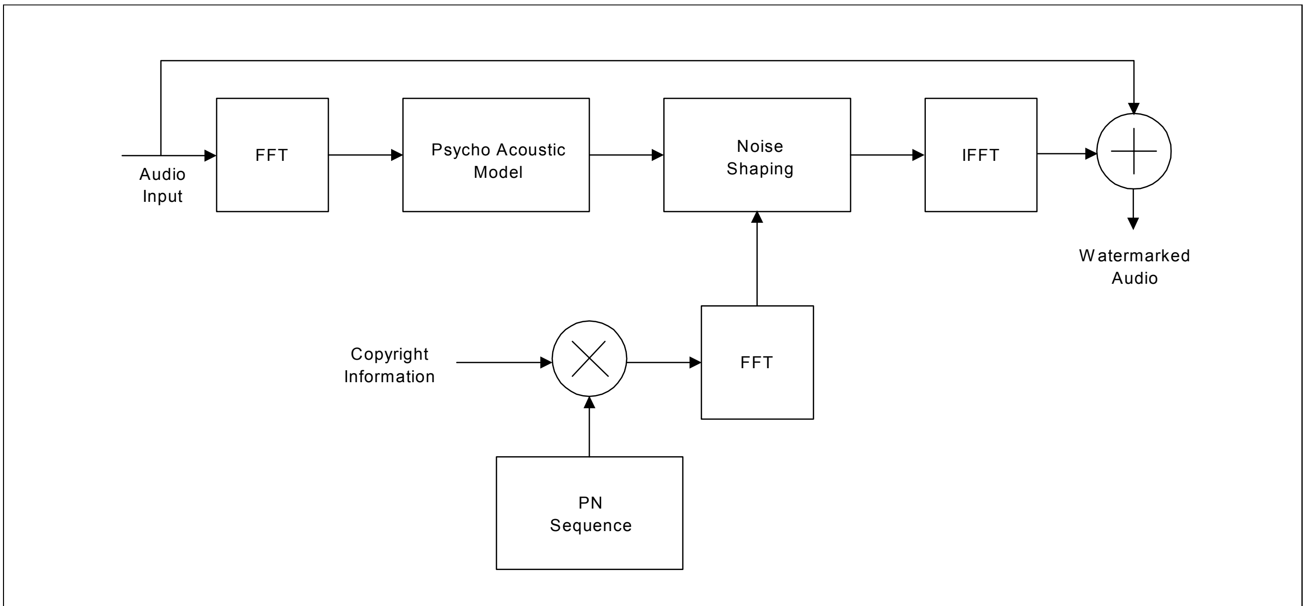

Fig. 2. Diagram of the audio watermark embedding procedure.

correlated. For a non-white Gaussian channel, it is possible to increase detection performance by pre-processing before correlation. This can be achieved by applying whitening or de-correlation before correlation.

Application of the whitening procedure should significantly remove the correlation in the signal to achieve optimum detection. To remove the correlation in the audio signal, we used an autoregressive modeling named linear predictive coding (LPC) [21].

The LPC method can approximate the original audio signal $x(n)$ as a linear combination of the past p audio samples, such that

$$x(n) \approx a_1 x(n-1) + a_2 x(n-2) + \cdots + a_p x(n-p) , \qquad (4)$$

where the coefficients $a_1, a_2, \cdots a_p$ are assumed constant over the analysis audio segment. We convert (4) to an equality by including an error term $u(n)$, giving

$$x(n) = \sum_{i=1}^{p} a_i x(n-i) + u(n) , \qquad (5)$$

where $u(n)$ is an excitation or residual signal of $x(n)$.

Using (5), we can express the watermarked audio signal as follows:

$$y(n) = \sum_{i=1}^{p} a_i x(n-i) + u(n) + w(n) . \qquad (6)$$

Based on the above analysis, the watermarked audio signal can be expressed as

$$y(n) = \sum_{k=1}^{p} a_k y(n-k) + m(n) , \qquad (7)$$

where $m(n)$ is the residual signal of the watermarked audio signal. We assume that, by the characteristics of the linear predictive analysis, $m(n)$ has the characteristics of both $u(n)$ and $w(n)$. We can consider the linear combination of the past audio sample as the estimate $\tilde{y}(n)$, defined as

$$\tilde{y}(n) = \sum_{k=1}^{p} a_k y(n-k) . \qquad (8)$$

We now form the prediction error $e(n)$, defined as

$$\begin{aligned} e(n) &= y(n) - \tilde{y}(n) \\ &= y(n) - \sum_{k=1}^{p} a_k y(n-k) \\ &= m(n). \end{aligned} \qquad (9)$$

Eq. (9) reveals that we can estimate the signal $m(n)$ with the audio spectrum removed, which has the characteristics of both the excitation signal of the original audio $u(n)$ and the watermark signal $w(n)$, from the watermarked audio signal using the linear predictive analysis.

This method transforms the non-white watermarked audio signal to a whitened signal by removing the audio spectrum. Figures 3(a) and (b) show the probability density function of a small segment of the watermarked audio signal before LPC filtering and the residual or error signal of the watermarked audio signal after LPC filtering. The probability density function of the watermarked audio signal is clearly not smooth and has a large variance. On the other hand, the probability density function of the residual signal after LPC filtering has a smoother distribution and a smaller variance than the watermarked audio signal. Furthermore, this distribution can be easily modeled using the Gaussian probability density function. Using the residual signal and pseudo-noise sequence as a template, we applied matched filtering to extract the embedded information. Figure 4 shows a diagram of the audio watermark detection scheme.

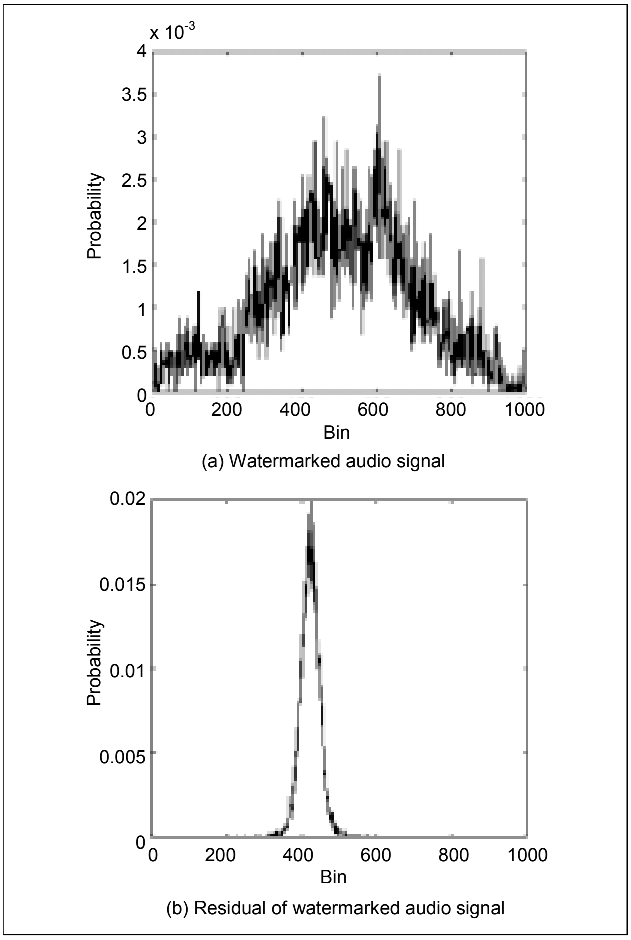

Fig. 3. Probability density function before and after linear prediction filtering.

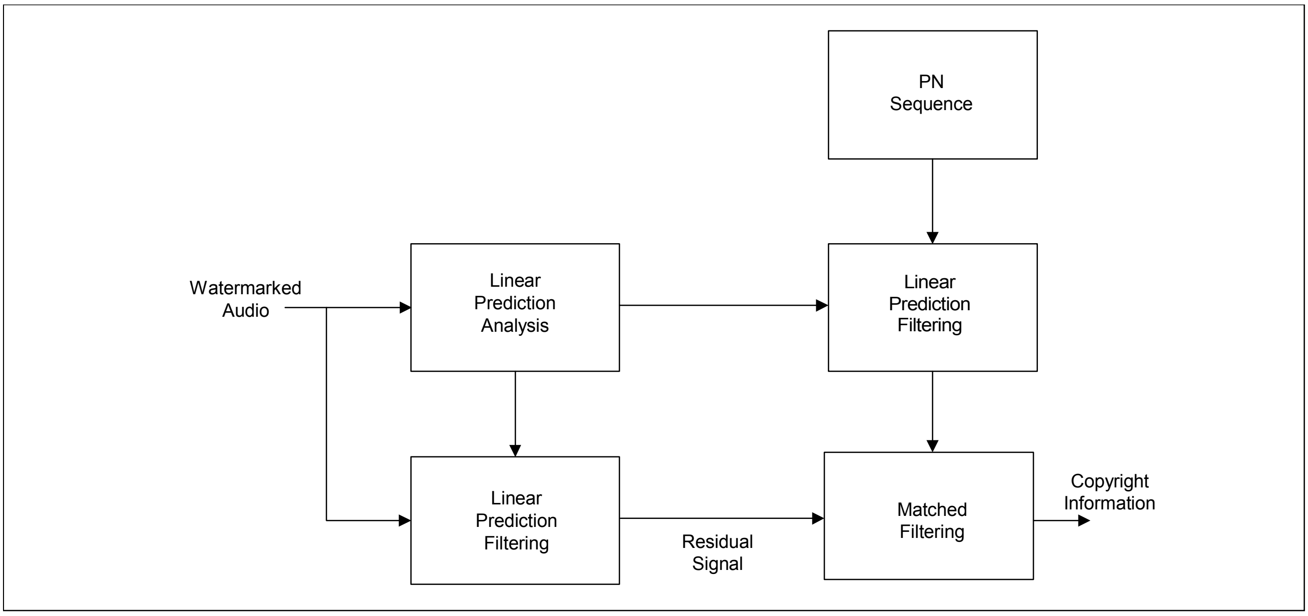

Fig. 4. Diagram of the audio watermark detection procedure.

IV. EXPERIMENTAL RESULTS

A total number of six audio pieces were prepared for the experiments from "The Ultimate Demonstration Disc/Chesky Records Guide," including jazz, pop, a cappella, and classical. Table 1 lists the music items used in the experiment. Each audio piece has a duration of 15 seconds.The audio signals used in the experiments were sampled at 44.1 kHz with a 16 bits/sample using a YAMAHA DS241 sound card. Information was embedded into the audio at a rate of 128 bits per 15 seconds. Figure 5 shows one frame of the audio signal and watermark signal which is to be embedded.

Table 1. The list of music items used in the experiments.

Music Items	Characteristics
Maiden Voyage – Leny Andrade	Base, Female vocal
Played Twice – The Fred Hersch Trio	Piano, Drum, Base
I Love Paris – Johnny Frigo	Violin
Sweet Georgia Brown – Monty Alexander	Drum, Base
Grandma's Hand – Livingston Taylor	Male a cappella
Flute Concerto in D – Vivaldi	Flute Concerto

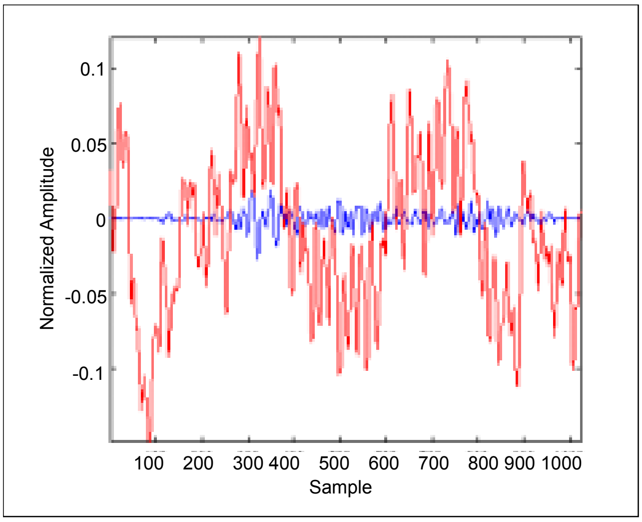

Fig. 5. A frame of the audio signal and resulting watermark signal to be embedded.

1. Robustness Test

To evaluate the performance of the proposed watermarking algorithm, we tested its robustness according to the SDMI (Secured Digital Music Initiative) Phase-1 robustness test procedure [22]. The detailed robustness test procedure is as follows.

• D/A, A/D conversion

Using a 16-bit full duplex sound card and a SONY TCD-D8 DAT, we performed the D/A and A/D conversions. First, the watermarked audio signal was sent to the DAT through the line-out terminal in the PC sound card. Next, the recorded watermarked audio signal in the DAT was sent to the PC. This means that the D/A and A/D conversions occurred twice.

• Mix down and amplitude compression

A two-channel stereo audio signal was mixed down to a one-channel mono signal, and the amplitude of the audio signal was compressed with a non-linear gain factor.

• **Band-pass filtering**

Using a 2nd order Butterworth filter which has 100 Hz and 6 kHz cutoff frequencies, band-pass filtering was applied.

• **Echo addition**

An echo signal with a delay of 100 ms and a decay of 50% was added to the original audio signal.

• **MPEG compression**

To evaluate the robustness against data compression attack, we selected an MPEG-1 Audio Layer 3 with a bit rate of 64 kbps/stereo and MPEG-2 AAC Audio Coding with a bit rate of 128 kbps/stereo.

• **Noise addition**

White noise with a constant level of 36 dB was added to the watermarked audio under the averaged power level of the audio signal.

• **Equalization**

We used a 10-band equalizer with the characteristics listed below:
Frequency [Hz]: 31, 62, 125, 250, 500, 1000, 2000, 4000, 8000, 16000
Gain [dB]: –6, +6, –6, +6, –6, +6, –6, +6, –6, +6

Table 2. Robustness test: detection results for various attacks.

Type of Attack	Bit Error	BER (%)
D/A, A/D	10/768	1.30
Mix Down	0/768	0.00
Amplitude Compression	0/768	0.00
Band-Pass Filtering	18/768	2.34
MPEG-1 Layer3	23/768	2.99
MPEG-2 AAC	7/768	0.91
Echo Addition	0/768	0.00
Noise Addition	21/768	2.73
Equalization	0/768	0.00

Table 2 shows the robustness test's detection results for the various attacks. We obtained a perfect detection result for the mix down, echo addition, and amplitude compression. Although some errors occurred in the MPEG-1 Audio Layer 3 coding and band-pass filtering, their detection results were acceptable.

2. Subjective Listening Test

To further evaluate the watermarked audio quality, we also performed an informal subjective listening test according to a double blind triple stimulus with hidden reference listening test [23]. The subjective listening test results are summarized in Table 3 under Diffgrade and # of Transparent Items. The Diffgrade is equal to the subjective rating given to the watermarked test item minus the rating given to the hidden reference: a Diffgrade near 0.00 indicates a high level of quality. The Diffgrade can even be positive, which indicates an incorrect identification of the watermarked item, i.e., the quality is transparent. The Diffgrade scale is partitioned into five ranges: imperceptible (> 0.00), not annoying (0.00 to –1.00), slightly annoying (–1.00 to –2.00), annoying (–2.00 to –3.00), and very annoying (–3.00 to –4.00)." The number of transparent items represents the number of incorrectly identified items. Fifteen listeners participated in the listening test. The quality of the watermarked audio signal was acceptable for all the test signals.

Table 3. List of music items used in the experiments.

Music Items	Diffgrade	# of Transparent Items
Maiden Voyage	0.02	8
Played Twice	–0.01	6
I Love Paris	0.05	8
Sweet Georgia Brown	0.13	7
Grandma's Hand	–0.02	5
Flute Concerto	0.01	7

V. CONCLUSION

In this paper, we described a new algorithm for digital audio watermarking. The proposed watermark embedding scheme accomplishes perceptual transparency after watermark embedding by exploiting the masking effect of the human auditory system. This embedding scheme adapts the watermark so that the energy of the watermark is maximized under the constraint of keeping the auditory artifact as low as possible. Our detection procedure extracts copyright information without access to the original signal by applying whitening, or decorrelation, before correlation. The whitening procedure effectively removes the correlation in the signal. Experimental results showed that our watermarking scheme is robust to common signal processing attacks, such as mix down, amplitude compression, and data compression.

Despite the success of the proposed method, it also has a drawback: a synchronization problem. The use of a PN sequence to generate the watermark signal is vulnerable to time scale modification attacks, such as linear speed change, pitch-invariant time scale modification, and wow and flutter. This phenomenon is caused by the loss of synchronization between the PN sequence and the watermarked audio signal with time scale modification. Further research will focus on overcoming this problem.

REFERENCES

[1] M. Swanson, M. Kobayashi, and A. Tewfik, "Multimedia Data Embedding and Watermarking Technologies," *Proc. of IEEE*, vol. 86, no. 6, June 1998, pp. 1064-1087.

[2] C. Cox, J. Killian, T. Leighton, and T. Shamoon, "Secure Spread Spectrum Communication for Multimedia," *Technical report*, N.E.C. Research Institute, 1995.

[3] R.J. Anderson and F. Peticolas, "On the Limit of Steganography," *IEEE J. Select. Areas Comm*., vol. 16, May 1998, pp. 474-481.

[4] A. Shizaki, J. Tanimoto, M. Iwata, "A Digital Image Watermarking Scheme Withstanding Malicious Attacks," *IEICE Trans. Fundamentals*, vol. E-83-A, no. 10, Oct. 2000, pp. 2015-2022.

[5] F. Hartung and M. Kutter, "Multimedia Watermarking Techniques," *Proc .of IEEE*, vol. 86, no. 6, June 1998, pp. 1079-1107.

[6] W. Bender, D. Gruhl, and N. Morimoto, "Techniques for Data Hiding," *Proc. SPIE*, vol. 2420, San Jose, CA, Feb. 1995, p. 40.

[7] M. Swanson, B. Zhu, A. Tewfik, and L. Boney, "Robust Audio Watermarking Using Perceptual Masking," *Signal Processing*, vol. 66, no. 3, May 1998, pp. 337-355.

[8] C. Neubauer and J. Herre, "Digital Watermarking and its Influence on Audio Quality," *105th AES Convention*, Audio Engineering Society preprint 4823, San Francisco, Sept. 1998.

[9] C. Neubauer and J. Herre, "Audio Watermarking of MPEG-2 AAC Bitstreams," *108th AES Convention*, Audio Engineering Society preprint 5101, Paris, Feb. 2000.

[10] C. Neubauer and J. Herre, "Advanced Audio Watermarking and its Applications," *109th AES Convention*, Audio Engineering Society preprint 5176, Los Angeles, Sept. 2000.

[11] P. Bassia, I. Pitas, and N. Nikolaidis, "Robust Audio Watermarking in the Time Domain," *IEEE Trans. on Multimedia*, vol. 3, no. 2, June 2001, pp. 232-241.

[12] C. Lee, K. Moallemi, and R. Warren, "Method and Apparatus for Transporting Auxiliary Data in Audio Signals," U.S. Patent 5,822,360, Oct. 1998.

[13] Recardo A. Garcia, "Digital Watermarking of Audio Signals Using a Psychoacoustic Auditory Model and Spread Spectrum Theory," *Proc. of 107th AES Convention*, preprint 5073, Sept. 1999.

[14] M. Veen, W. Oomen, F. Bruekers, J. Haitsma , T. Kalker, and A. Lemma, "Robust, Multi-Functional, and High Quality Audio Watermarking Technology," *Proc. of 110th AES Convention*, preprint 5345, May 2001.

[15] L. Boney, A. Tewfik, and K. Hamdy, "Digital Watermarks for Audio Signals," *Proc. of Multimedia'96*, 1996, pp. 473-480.

[16] C. Podilchuk and W. Zeng, "Image-Adaptive Watermarking Using Visual Models," *IEEE J. on Selected Areas in Comm., special issue on Copyright and Privacy Protection*, vol. 16, no. 4, May 1998, pp. 525-539.

[17] J.P.M.G. Linnartz, A.A.C. Kalker, G. Depovere, "On the Design of a Watermarking System: Considerations and Rationales," *Information Hiding Workshop, Preliminary Proc. and Springer Verlag, LNCS* 1768, Sept. 1999, pp. 254-269.

[18] G. Depovere, T. Kalker, and J.-P. Linnartz, "Improved Watermark Detection Reliability Using Filtering before Correlation," *Proc. ICIP*, vol. 1, 1998, pp. 430-434.

[19] ISO/IEC IS 11172, Information Technology – Coding of Moving Pictures and Associated Audio for Digital Storage up to about 1.5 Mbits/s.

[20] T. Painter and A. Spanias, "Perceptual Coding of Digital Audio," *Proc. of IEEE*, vol. 88, no. 4, Apr. 2000, pp. 451-515.

[21] B.S. Atal and S.L. Hanauer, "Speech Analysis and Synthesis by Linear Prediction of the Speech Wave," *J. Acoust. Soc. Am*., vol. 50, 1971, pp. 637-655.

[22] SDMI Portable Device Specification 1.0, PDWG99070802, July 1999.

[23] ITUT-R Rec. BS. 1116, "Method for the Subjective Assessment of Small Impairments in Audio Systems Including Multichannel Sound Systems," International Telecommunication Union, Geneva, Switzerland, 1994.

Jongwon Seok received his BS, MS and PhD degrees in electronics from Kyungpook National University, Taegu, Korea in 1993, 1995 and 1999. Since 1999, he has been a Senior Member of Research Staff in Electronics and Telecommunications Research Institute (ETRI), Korea. He has been engaged in the development of multichannel audio codec system, digital broadcast content protection and management. His research interests include digital signal processing in the field of multimedia communications, multimedia systems, and digital content protection and management.

Jinwoo Hong was born on April 15, 1959 in Yujoo, Korea. He received his BS and MS degrees in electronic engineering from Kwangwoon University, Seoul, Korea, in 1982 and 1984, respectively. He also received his PhD degree in computer engineering from the same university in 1993. Since 1984, he has been with ETRI in Daejeon, Korea, as a Principal Member of Engineering Staff, where he is currently a Team Manager of the Broadcasting Content Protection Research Team. From 1998 to 1999, he researched at Fraunhofer Institute in Erlangen, Germany, as a Visiting Researcher. He has been engaged in the development of the TDX digital switching system, multichannel audio codec system, digital broadcast content protection and management. His research interests include audio & speech coding, acoustic signal processing, audio watermarking, and IPMP.

Jinwoong Kim received his BS and MS degrees from Seoul National University, Seoul, Korea in 1981 and 1983, respectively, and his PhD degree in the Department of Electrical Engineering from Texas A&M University, United States in 1993. In 1983, he joined ETRI, Korea, where he is currently the Director in the Broadcasting Media Research Department. He has been engaged in the development of TDX digital switching system, MPEG-2 video encoder, HDTV encoder system, and MPEG-7 technology. His research interests include digital signal processing in the field of video communications, multimedia systems, and interactive broadcast systems.