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Minh-Nhut Ngo
Ho Chi Minh City University of Science

Ryota Miyauchi
Japan Advanced Institute of Science and Technology

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Nhut Minh Ngo, Masashi Unoki, and Ryota Miyauchi

Japan Advanced Institute of Science and Technology
1-1 Asahidai, Nomi, Ishikawa, 923-1292, Japan
nmnhut@jaist.ac.jp; unoki@jaist.ac.jp; ryota@jaist.ac.jp

Yôiti Suzuki

Research Institute of Electrical Communication and
Graduate School of Information Sciences, Tohoku University
2-1-1, Katahira, Aoba-ku, Sendai, 980-8577, Japan
yoh@riec.tohoku.ac.jp

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Abstract. *This paper proposes a data hiding scheme for amplitude-modulation (AM) radio broadcasting systems. The method of digital audio watermarking based on cochlear delay (CD) that we previously proposed is employed in this scheme to construct a data-hiding scheme in the AM domain. We investigate the feasibility of applying the method of CD-based inaudible watermarking to send inaudible additional messages in AM signals. The proposed scheme modulates a carrier signal with both original and watermarked signals as lower and upper sidebands by using the novel double-modulation and then transmits the modulated signal to the receivers. Particular receivers in the proposed scheme demodulate the received signals to get both original and watermarked signals by using the double-demodulation and then extract messages from the watermarked signal and the original signal using CD-based watermarking. The results we obtained from computer simulations revealed that the proposed scheme can transmit messages as watermarks in AM signals and then correctly extract the messages from observed AM signals. The results also indicated that the sound quality of the demodulated signals could be kept high not only with the proposed scheme but also in traditional AM radio systems. This means that the proposed scheme has the possibility of acting as a hidden-message transmitter as well as having low-level compatibility with AM radio systems. The proposed scheme could be applied in emergency alert systems and high utility AM radio services.*
Keywords: data hiding, cochlear delay, amplitude modulation, emergency communications.

1. **Introduction.** Radio is a simple and typical technique for the wireless communication of signals through free space. The baseband signal, which represents text, images, or audio, is conveyed by a carrier signal. The amplitude, frequency, or phase of carrier signals is modified in relation to baseband signals by a process called modulation. The two most typical kinds of analog modulation technique are amplitude modulation (AM) and frequency modulation (FM), and we have corresponding radio systems, i.e., AM and FM radios. AM radio has some advantages over FM radio, such as its use of narrow channel bandwidth and wider coverage area, while FM radio provides a better sound quality of the conveyed sound at the expense of a using larger channel bandwidth [1]. AM and FM radios have been used popularly in our daily life to receive radio services

broadcasting news, entertainment, and educational programs. When radio users listen to radio programs, all the information is usually provided by audio signals only. However, if additional information such as programs, weather forecasting, news, advertisements, etc. was also provided digitally, as in recently introduced smart TV broadcasting, such information could improve the utility of radio services as well.

As radio transmitters and receivers for radio are simple in construction, radio is a robust communication technology and thus useful in the event of natural disasters (e.g., earthquakes or typhoons). Radio receivers can operate for many hours on just a few batteries while other devices such as televisions and PCs cannot. Radio is a particularly useful device during and after disasters in which power suppliers can easily be cut off. Emergency Alert Systems (EAS) [2] and Earthquake Early Warning systems (EEW) [3, 4] have been working on broadcasting warnings via radio services. Audible emergency warnings are used to attract the attention of people engaged in everyday activities (e.g., office workers and car drivers). These emergency warnings allow workers to take protective actions such as slowing and stopping trains or taking steps to protect important infrastructure, and provide people a few seconds to take cover [5, 6]. The warning messages need to be accurately and quickly understood by everyone during emergencies, though, so complementing audible messages with additional digital information expressed in characters is needed to more effectively broadcast warnings during and even after disasters. The application of multimedia to conventional radio will therefore be useful particularly in the event of emergencies where the ordinary Internet does not work well.

In a nutshell, supplementing the sound (speech) information conveyed by radio waves with additional digital information is likely to be very useful in emergency situations. Data hiding for radio signals is therefore important for public safety. Some advanced radio systems such as Radio Data System (RDS) [7] and AM Signalling System (AMSS) [8, 9, 10] have been proposed to embed a certain amount of digital information within broadcast signals. RDS embeds digital information into a subcarrier of the broadcast signal in FM radio broadcasts. The data rate of the RDS is 1187.5 bits per second (bps) [7]. RDS allows some functions to be implemented in FM radio services, such as Program Identification, Program Service name, and Alternative Frequency list. AMSS provides broadly similar functionality to that offered by RDS, and uses low bit-rate phase modulation of the AM carrier to add a small amount (about 47 bps [9]) of digital information to the broadcast signal. The envelope detector used in conventional AM receivers, however, does not respond to AMSS. While AMSS has the limitations of a low data rate and incompatibility with conventional radio receivers, RDS can be effectively used to embed digital information with a high data rate into radio signals in FM radio broadcasts. This drawback motivated us to search for a new method of information hiding for AM signals.

Data-hiding techniques such as audio watermarking have been proposed in recent years to protect the copyright information of public digital-audio content and to transmit digital information in the same channel [11]. Many approaches have been taken towards watermarking for digital audio content, which can embed and then precisely detect data. These approaches have been implemented in computer and the results suggest that digital information can be embedded into digital audio with little or no effect on the listener's perception of the target audio. It has also been reported that the bit rate of embedded data is reasonably high, in the hundreds of bits per second (bps) range [11, 12]. We can take these advantages of watermarking techniques and apply them to an AM radio system to create a data hiding scheme for AM radio signals.

This paper proposes a novel data-hiding scheme for AM radio signals. Digital information is embedded into audio signals instead of modulated signals as is done in RDS and AMSS systems. The carrier signal is then modulated by the audio signals and transmitted to receivers. Although methods of embedding data and detecting data have been taken care of by watermarking techniques, some issues might arise due to the modulation and demodulation processes in an AM radio system. For instance, the inaudibility of watermarked signals and the accuracy of embedded data detection could be reduced. Additionally, the vast majority of users listen to AM radio using conventional receivers. The hiding system should have downward compatibility, i.e. it should yield AM signals that can be detected by conventional receivers in order to be widely applied in practice. Suitably-equipped receivers will be able to extract embedded data in addition to audio signals while conventional receivers will be able to extract only an audio signal. The proposed scheme could be used to broadcast audio signals and embedded data to receivers over an AM radio link, and applied to construct a high utility AM radio service able to multi-modally distribute essential information during emergencies.

The rest of this paper is organized as follows. Section II provides a basic explanation of the amplitude modulation technique and the concept underlying the data-hiding scheme. In section III, we explain the proposed data-hiding scheme. Section IV gives the results from an evaluation of the proposed scheme. Discussion and conclusion are presented in section V.

2. Amplitude Modulation and Concept of the Data Hiding Scheme.

2.1. Amplitude modulation in radio broadcast. Modulation is a typical technique used to transmit information (e.g., text, image, and sound) in telecommunication. It is used to gain certain advantages such as far-distance communication and transmission of signals over radio waves. The baseband signal, which carries the information, cannot be directly transmitted over a radio wave because it has sizable power at low frequencies. The size of antennas that radiate the waveform signal is directly proportional to the signal wavelength. Long-haul communication over a radio wave requires modulation to enable efficient power radiation using antennas of reasonable dimensions [13].

AM varies the amplitude of a carrier signal $c(t)$ which is usually a sinusoidal signal of high frequency in proportion to the message signal $m(t)$ according to the formula,

$$u(t) = [A + m(t)]c(t) = [A + m(t)]\cos(\omega_c t), \tag{1}$$

where ω_c is angular frequency (rad/s) of the carrier signal and $u(t)$ is referred to as an AM signal. The modulation depth is defined as $\frac{\max(|m(t)|)}{A}$.

Equation (1) represents the basic AM scheme, namely a double-sideband with carrier (DSB-WC). There are other types of AM such as double-sideband suppressed carrier (DSB-SC) and single-sideband modulation (SSB) which can be referred to in [13].

The waveform and spectrum of the message signal and the modulated signal are depicted in Figs. 1(a) and 1(b), respectively. AM simply shifts the spectrum of $m(t)$ to the carrier frequency. Suppose $M(\omega)$ is the spectrum of the message signal $m(t)$; the spectrum of the AM signal is then expressed by

$$U(\omega) = \pi A\delta(\omega + \omega_c) + \pi A\delta(\omega - \omega_c) + \frac{1}{2}M(\omega + \omega_c) + \frac{1}{2}M(\omega - \omega_c). \tag{2}$$

We observe that if the bandwidth of $m(t)$ is B Hz, then the bandwidth of the modulated signal $u(t)$ is $2B$ Hz. The spectrum of the modulated signal centered at ω_c is composed of two parts: a portion called the lower sideband (LSB) lying below ω_c and a portion called the upper sideband (USB) lying above ω_c. Similarly, the spectrum centered at $-\omega_c$ also has an LSB and USB.

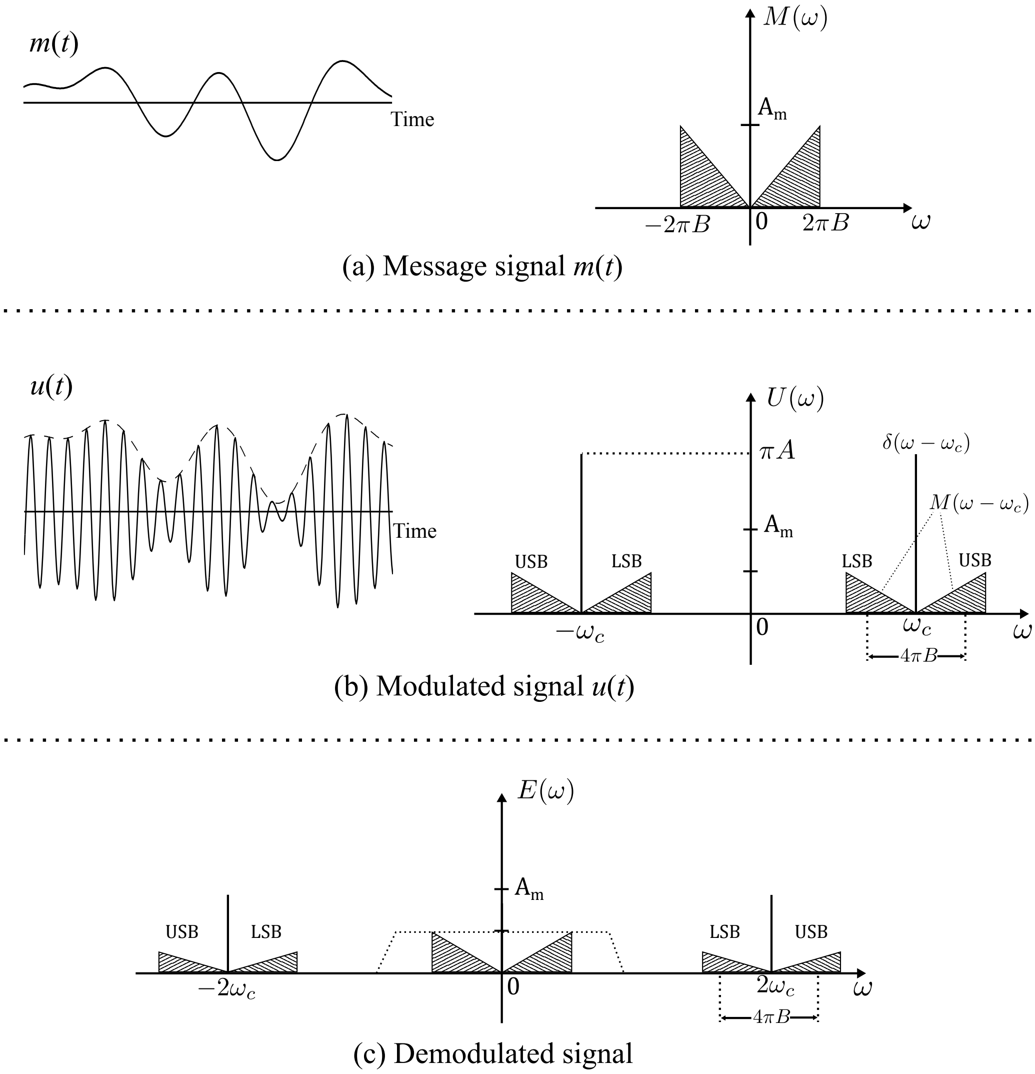

FIGURE 1. Waveform and spectrum of signals in DSB-WC method: (a) message signal, (b) modulated signal, and (c) demodulated signal.

2.2. **Demodulation of the AM signal.** There are two types of demodulation: synchronous demodulation and asynchronous demodulation. They differ in the use of a carrier signal in demodulation processes. The receiver must generate a carrier signal synchronized in phase and frequency for synchronous demodulation, but the carrier signal is not needed in asynchronous demodulation.

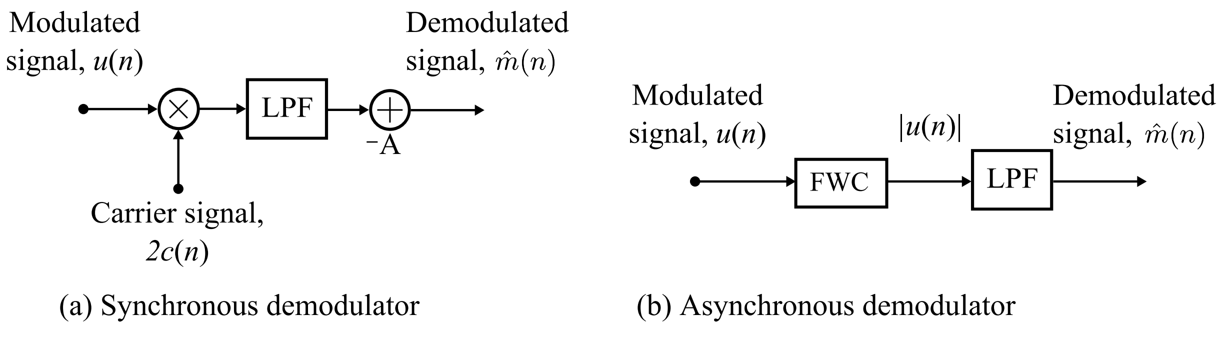

FIGURE 2. Block diagram of demodulation methods: (a) synchronous demodulator and (b) asynchronous demodulator for DSB-WC.

Synchronous Demodulation. This kind of demodulation can be referred to as a coherent or product detector. Figure 2(a) shows a flow chart of a synchronous demodulator. At the receiver, we multiply the incoming modulated signal by a local carrier of frequency and phase in synchronism with the carrier used at the transmitter.

$$\begin{aligned} e(t) &= u(t)c(t) = [A + m(t)]\cos^2(\omega_c t) \\ &= \frac{1}{2}[A + m(t)] + \frac{1}{2}[A + m(t)]\cos(2\omega_c t) \end{aligned} \tag{3}$$

Let us denote $m_a(t) = A + m(t)$. Then,

$$e(t) = \frac{1}{2}m_a(t) + \frac{1}{2}m_a(t)\cos(2\omega_c t) \tag{4}$$

The Fourier transform of the signal $e(t)$ is

$$E(\omega) = \frac{1}{2}M_a(\omega) + \frac{1}{4}[M_a(\omega + 2\omega_c) + M_a(\omega - 2\omega_c)] \tag{5}$$

The spectrum $E(\omega)$ consists of three components as shown in Fig. 1(c). The first component is the message spectrum. The two other components, which are the modulated signal of $m(t)$ with carrier frequency $2\omega_c$, are centered at $\pm 2\omega_c$.

The signal $e(t)$ is then filtered by a lowpass filter (LPF) with a cut-off frequency of f_c to yield $\frac{1}{2}m_a(t)$. We can fully get $m_a(t)$ by multiplying the output by two. We can also get rid of the inconvenient fraction $\frac{1}{2}$ from the output by using the carrier $2\cos(\omega_c)$ instead of $\cos(\omega_c)$. Finally, the message signal $m(t)$ can be recovered by $\hat{m}(t) = m_a(t) - A$.

Asynchronous Demodulation. An asynchronous demodulator can be referred to as an envelope detector. For an envelope detector, the modulated signal $u(t)$ must satisfy the requirement that $A + m(t) \geq 0, \forall t$. A block diagram of the asynchronous demodulator is shown in Fig. 2(b). The incoming modulated signal, $u(t)$, is passed through a full wave rectifier (FWR) which acts as an absolute function. The FWR output which is the absolute value of $u(t)$, $|u(t)|$, is then filtered by a low-pass filter resulting in the demodulated signal, $\hat{m}(t)$.

A synchronous demodulator can decode over-modulated signals and the modulated signals produced by DSB-SC and SSB. A signal demodulated with a synchronous demodulator should have signal to noise ratio higher than that of the same signal demodulated with an asynchronous demodulator. However, the frequency of the local oscillator must be exactly the same as the frequency of the carrier at the transmitter, or else the output message will fade in and out in the case of AM, or be frequency shifted in the case of SSB. Once the frequency is matched, the phase of the carrier must be obtained, or else the demodulated message will be attenuated. On the other hand, an asynchronous demodulator does not need to generate the carrier signal at the receiver, thus it is simple to implement in practical applications.

2.3. **Concept of the data hiding scheme.** Telecommunication systems, such as AM radio broadcasting systems, modulate the carrier signal with the audio signal for transmission of the audio signal to receivers. Modulation is advantageous because the baseband signals have sizable power at low frequencies while modulated signals have sufficiently strong power to be transmitted over radio links [13]. At the receivers, the modulated signals are demodulated to extract the audio signals. Based on this principle of modulation, two approaches to embedding data into AM radio signals seem feasible. The first approach is to directly modify modulated signals to embed data. With this approach, however, it is difficult to ensure downward compatibility. Another approach is to embed data into audio signals and then modulate the carrier signal with the embedded audio signals. This

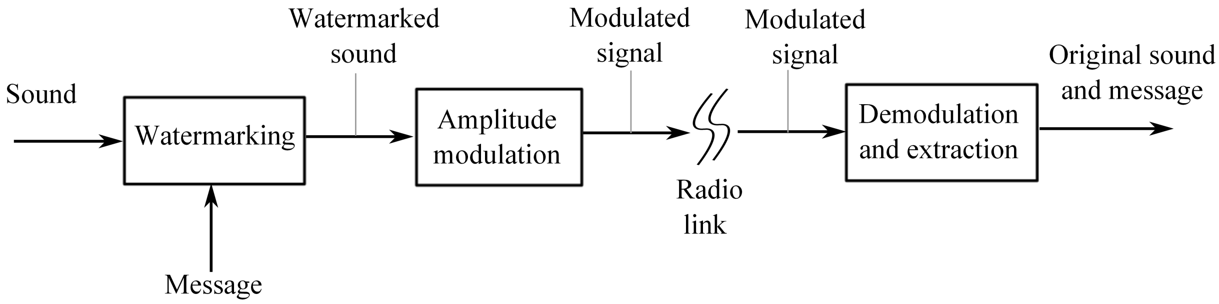

FIGURE 3. Approach to a data-hiding scheme for AM radio signals.

approach could be implemented by employing a proposed audio watermarking scheme. Our work focuses on this approach to construct data hiding scheme for digital-audio in AM domain.

Figure 3 shows the approach to embedding data into AM radio signals. Audio signals are embedded with data by using an available method of audio watermarking before they are modulated for transmitting. Many methods of audio watermarking have been developed in recent years. The method of audio watermarking used to embed data into audio signals in this scheme should satisfy the requirements of inaudibility, robustness, and high capacity. After the audio signal is embedded with data, the embedded signal is modulated by a modulation process. The modulated signal is then sent to receivers through an AM radio link. When the receivers in this scheme receive the modulated signal, they can demodulate the received signal and extract the embedded message from the demodulated signal by using a demodulation process and a data extractor. The extracted audio signal and the detected data are then used for the desired purposes.

We employed a method of audio watermarking based on cochlear delay (CD) characteristic because it enable a high embedding capacity, provides high quality of watermarked audio, is robust against signal modification, and conceals the embedded watermark [14]. CD-based watermarking method can be used to embed data into audio signals inaudibly and to detect data from watermarked signals precisely. The method we used applies a non-blind detection scheme to extract embedded data in the detection process. Using a non-blind scheme to extract embedded data could result in a lower detection error rate as well as higher embedding capacity. However, it would make it necessary to use two broadcasters to transmit both the original and the watermarked signals. That would increase the system cost and complexity.

To efficiently apply the CD-based watermarking method in the proposed scheme, we propose novel double-modulation and double-demodulation algorithms to modulate a carrier signal with both the original and watermarked signals. The double-modulation algorithm produces an AM signal that can carry both the original and watermarked signals simultaneously, and the double-demodulation algorithm can be used to extract the two original and watermarked signals from the AM signal. The double-modulation algorithm was also designed to yield AM signals that can be reacted to by conventional radio devices. As a result, the drawback of blind detection in a CD-based watermarking method could be overcome by the proposed double-modulation and double-demodulation algorithms. The proposed scheme combined the advantages of a CD-based watermarking method and those of the double-modulation/double-demodulation algorithms to enable an efficient data hiding for AM radio signals.

3. **Data Hiding Scheme for AM Radio Broadcast.**

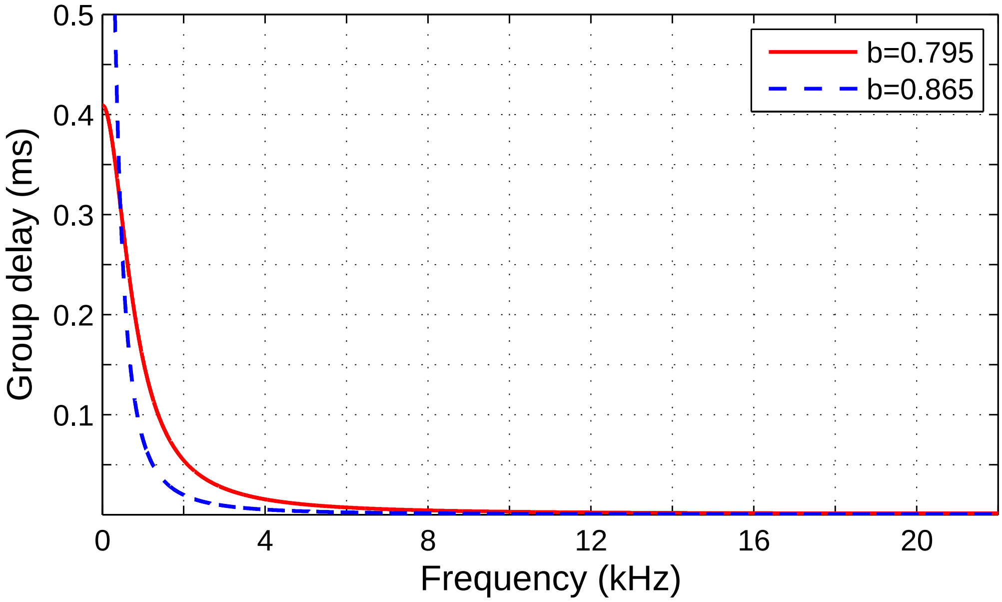

FIGURE 4. Group delay characteristics of 1^{st} order IIR all-pass filters.

3.1. **Digital audio watermarking.** Digital-audio watermarking has been proposed for copyright protection of digital-audio material and to transmit hidden messages in the host signal [11, 12, 15, 16, 17]. Many approaches have been proposed based on five requirements: (a) inaudibility (watermarks should be imperceptible to listeners), (b) confidentiality (secure and accurate concealment of embedded data), (c) robustness (watermarking methods should be robust against various attacks, e.g., resampling and compression), (d) blind detection (the ability to detect watermark without original signals), and (e) high watermarking capacity (the ability to conceal a lot of information). It should be noted that these requirements depend on each class of application. For example, a watermarking method with high capacity is necessary for covert communication applications, while copyright protection applications need confidentiality and robustness. Different applications demand different types of watermarking schemes with different requirements.

The most basic and simplest audio watermarking techniques have been based on the least significant bit (LSB-shifts) [11]. Watermarks can be embedded with this approach at a high data rate but at the cost of being vulnerable to various manipulations (e.g., resampling and compression). Other methods of digital-audio watermarking are based on characteristics of the human auditory system (HAS), such as the direct spread spectrum method (DSS) [12] and the secure spread spectrum [18]. Methods based on a spread spectrum are relatively robust against attacks since watermarks are spread throughout whole frequencies. However, they do not satisfy the other requirements, especially the inaudibility requirement [14]. There are a variety of watermarking schemes based on manipulating the phase spectra of signals, such as the echo-hiding approach [19] and a method based on periodical phase modulation (PPM) [20, 21]. Although these methods can partially satisfy the five requirements, it is difficult to achieve an audio watermarking scheme that can simultaneously satisfy all of these requirements, especially the inaudibility and robustness requirements [14].

Characterizing mechanism of HAS, a method of digital-audio watermarking based on the cochlear delay (CD) characteristic could satisfy most of the requirements: inaudibility, confidentiality, robustness, and high capacity [14]. This method is based on properties of the human cochlea, a fluid-filled cavity that receives vibrations caused by sound signals. The cochlea and other parts of the auditory system help us perceive sound. Researchers

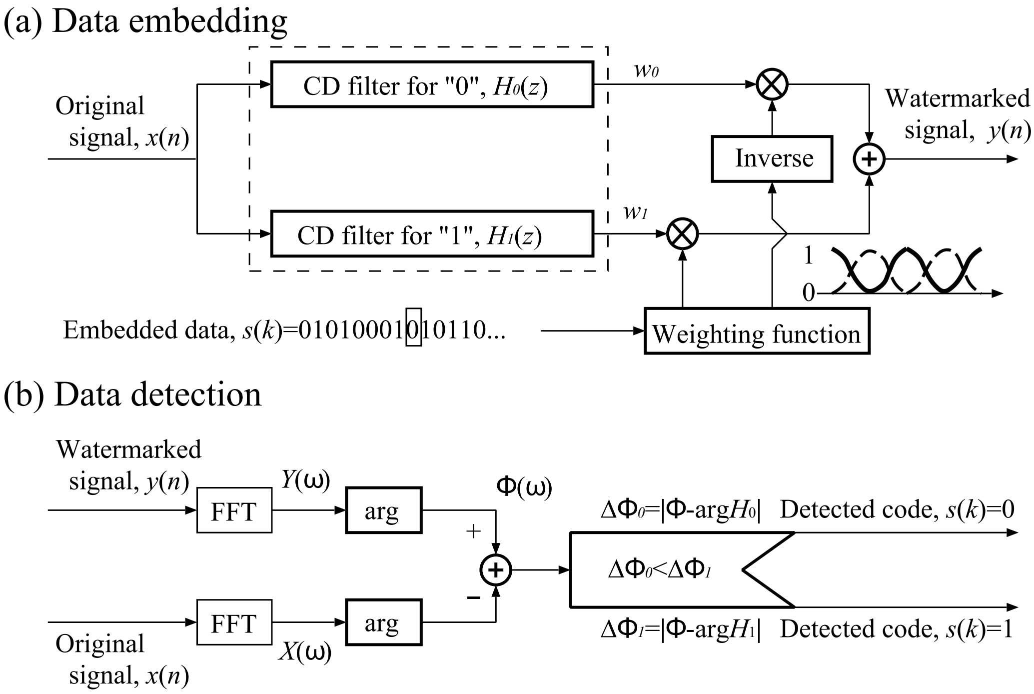

Figure 5. Method of digital audio watermarking based on the cochlear delay characteristic: (a) data-embedding and (b) data-detection process [22, 23].

have shown that different frequency components of sound signals excite different positions in the cochlea (see [14] and references therein). Low frequency components require more time to reach the corresponding places near the apex of the cochlear while high frequency components excite places near the base of the cochlea. The difference in travel time through the cochlea for low frequency components compared to high frequency components is referred to as "cochlear delay." Studies on human perception with regard to cochlear delay by Aiba et al. [22, 23] suggest that the human auditory system cannot distinguish sound with enhanced delay and non-processing sound.

The key idea behind this method is that enhancing group delays related to CD does not affect human perception of the target sound. The method of audio watermarking based on CD embeds data $s(k)$ into sound $x(n)$ by controlling the two different group delays of CD (phase information) of the original signal in relation to bit data being embedded ("1" and "0"). Figure 5 is a block diagram of the (a) embedding and (b) detection process for the CD-based method.

This method uses two first order IIR all-pass filters, $H_0(z)$ and $H_1(z)$, to control the group delay of the original signal. The phase characteristics of these two filters are modeled as cochlear delay as shown in Fig. 4. The outputs of $H_0(z)$ and $H_1(z)$ are $w_0(n)$ and $w_1(n)$ which are not affected in terms of human perception. Then, $w_0(n)$ and $w_1(n)$ are decomposed into segments. Finally, these signal segments are merged together in relation to watermark $s(k)$ (e.g., "010010101100110") as in Eq. (6), which results in watermarked signal $y(n)$. A weighting ramped cosine function was used to avoid discontinuity with this method between the marked segments in the watermarked signal.

$$y(n) = \begin{cases} w_0(n), & s(k) = 0 \\ w_1(n), & s(k) = 1, \end{cases} \tag{6}$$

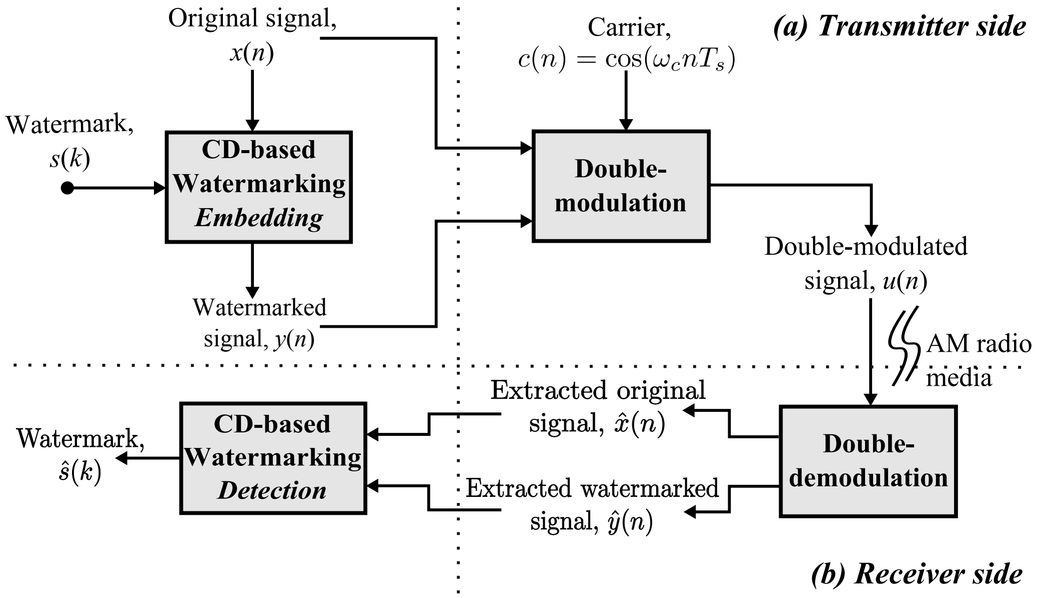

Figure 6. General scheme for digital audio data hiding in the AM domain.

where $(k-1)\Delta W \leq n < k\Delta W$. n is the sample index, k is the frame index, and $\Delta W = f_s/N_{bit}$ is the frame length. f_s is the sampling frequency of the original signal and N_{bit} is the bit rate per second (bps) of embedded data.

The detection process of this method is a non-blind process. The original and watermarked signals are first decomposed into segments using the same window function used in the embedding process as shown in Fig. 5(b). The phase differences, $\phi(\omega)$s, are calculated between the segments of the original signal and those of the watermarked signal as in Eq. (7). FFT[$\cdot$] is the fast Fourier transform (FFT). The phase difference of each segment, $\phi(\omega)$, is then compared to the group delays of $H_0(z)$ and $H_1(z)$ to detect bits of "0" or "1" as in Eqs. (8), (9), and (10). If $\phi(\omega)$ is closer to the group delay of $H_0(z)$ than that of $H_1(z)$, bit "0" is detected. Otherwise, bit "1" is detected.

$$\phi(\omega_m) = \arg(\text{FFT}[y(n)]) - \arg(\text{FFT}[x(n)]), \tag{7}$$

$$\Delta\Phi_0 = \sum_m \left|\phi(\omega_m) - \arg(H_0(e^{j\omega_m}))\right|, \tag{8}$$

$$\Delta\Phi_1 = \sum_m \left|\phi(\omega_m) - \arg(H_1(e^{j\omega_m}))\right|, \tag{9}$$

$$\hat{s}(k) = \begin{cases} 0, & \Delta\Phi_0 < \Delta\Phi_1 \\ 1, & \text{otherwise} \end{cases} \tag{10}$$

3.2. **System architecture.** The proposed data-hiding scheme for AM radio signals is shown in Fig. 6. There are two main phases in this scheme: (a) data-embedding and a double-modulation process on the transmitter side and (b) a double-demodulation and data-detection process on the receiver side.

On the transmitter side (Fig. 6(a)), the CD-based method of watermarking is used to embed data, $s(k)$, into audio signal $x(n)$. The output of this step is the watermarked signal, $y(n)$. The original and watermarked signals, $x(n)$ and $y(n)$, are modulated as LSB and USB, respectively, with the carrier, $c(n)$ by the proposed double-modulation process. The proposed double-modulation process is based on AM with the method of DSB-WC, but there is a difference between DSB-WC and the proposed scheme: LSB and USB are different in the proposed double-modulation process while LSB and USB in DSB-WC are the same. The double-modulated signal, $u(n)$, which conveys both the original and watermarked signals, $x(n)$ and $y(n)$, is used for broadcasting to receivers.

For the particular receivers that are suitably-equipped with the double-demodulator and the watermark detector as shown in Fig. 6(b), they can extract the conveyed signals from the double-modulated signals and the embedded data therein. The double-modulated signal, $u(n)$, is first demodulated to extract the conveyed signals, $\hat{x}(n)$ and $\hat{y}(n)$, by the proposed double-demodulation process. The original signal is extracted from LSB and the watermarked signal is extracted from USB . The extracted signals, $\hat{x}(n)$ and $\hat{y}(n)$, are then used to detect the embedded data $\hat{s}(k)$ by using the detection process of CD-based watermarking.

For conventional receivers that use an envelope detector to extract signals, they are able to extract the conveyed signals as the original signal from the double-modulated signal. Although LSB and USB in double-modulated signals are not the same, the differences between them are very slight, much like the phase modulation related to CD patterns. As a result, the extracted signals seem to be a mixture of the original and watermarked signals which are slightly distorted.

The next two sections describe the implementation of the double-modulation and double-demodulation processes used in this scheme. The implementation of the audio watermarking method based on CD (the embedding and detection processes) is described in Section 3.1.

3.3. **Double-modulation process.** The block diagram for the double-modulation process is shown in Fig. 7(a). First, the original signal $x(n)$, the watermarked signal $w(n)$, and the carrier signal $c(n)$ are split into successive frames. The frames are sequentially processed and then the outputs of all frames are merged together to form the modulated signal $u(t)$. Each frame of $x(n)$, $w(n)$, and $c(n)$ is processed in four steps:

Step 1: Original signal $x(n)$ and watermarked signal $y(n)$ are modulated with the DSB-WC method. The standard-modulated signals $u_1(n)$ and $u_2(n)$ are obtained as the outputs of standard modulation processes.

Step 2: The standard-modulated signals $u_1(n)$ and $u_2(n)$ are then transformed into a Fourier domain by FFT. The outputs correspond to frequency spectra $U_1(\omega)$ and $U_2(\omega)$ of $u_1(n)$ and $u_2(n)$, respectively. Each spectrum contains three parts, i.e., LSB, carrier component $C(\omega)$, and USB.

Step 3: The LSB and carrier component of $U_1(\omega)$ and the USB of $U_2(\omega)$ are merged into $U(\omega)$. $U(\omega)$ then contains three parts: LSB of $U_1(\omega)$, carrier component $C(\omega)$, and USB of $U_2(\omega)$.

Step 4: Finally, $U(\omega)$ is transformed into a time domain by using inverse FFT (IFFT). The double-modulated signal $u(n)$, which is the output of the IFFT, carries both signals $x(n)$ and $y(n)$.

Figure 8 shows the differences between the DSB-WC method and the double-modulation method. The original signal is modulated as LSB and USB in DSB-WC, where LSB and USB are the same. On the other hand, original and watermarked signals are modulated as LSB and USB in double modulation, where LSB and USB are different. The double-modulated signal which is the output from double-modulation process simultaneously carries two signals while the modulated signal which is the output from DSB-WC carries only one signal.

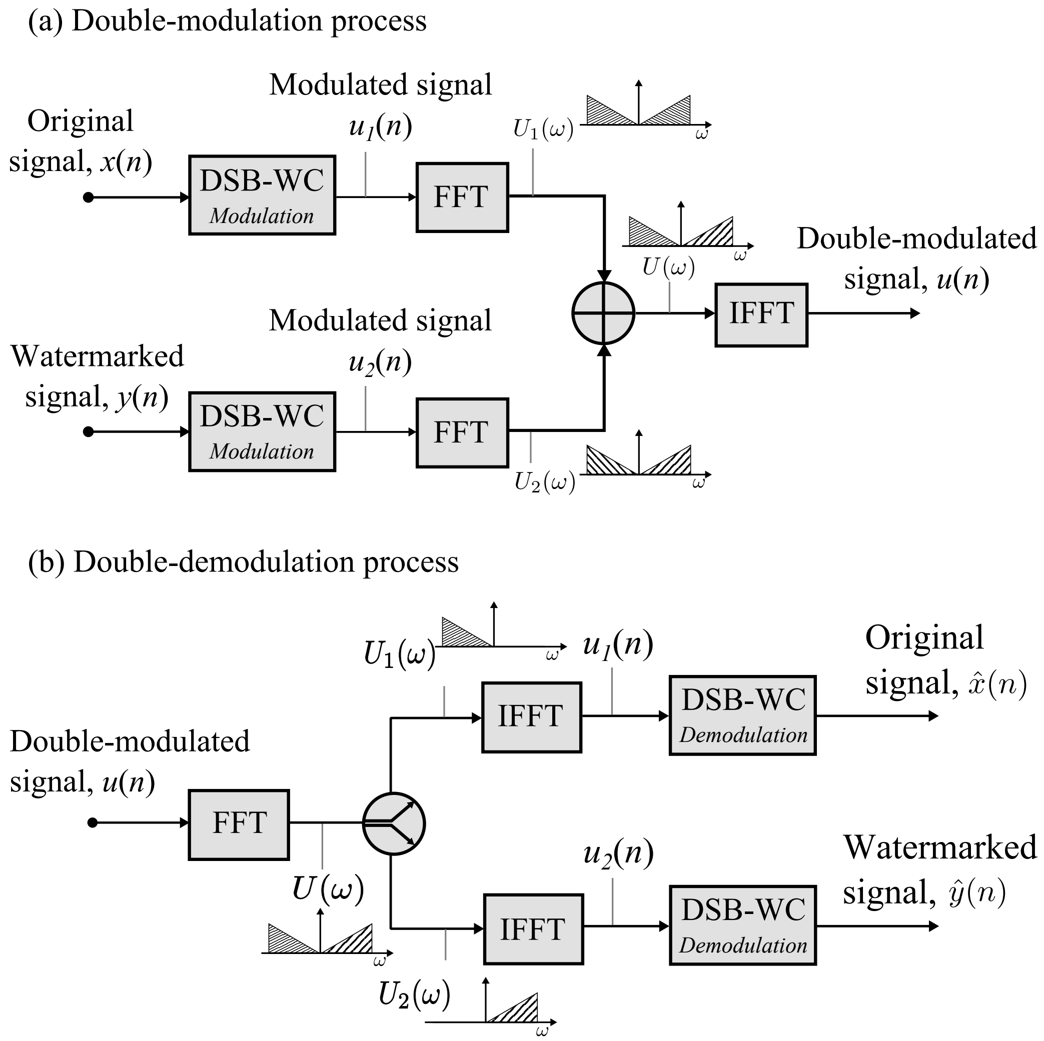

FIGURE 7. Block diagram of (a) double-modulation process and (b) double-demodulation process.

3.4. **Double-demodulation process.** The double-demodulation process extracts the original signal $x(n)$ and the watermarked signal $y(n)$ from the double-modulated signal, $u(n)$. The block diagram of the double-demodulation process is shown in Fig. 7(b). First, the double-modulated signal $u(n)$ is split into successive frames. Each frame of $u(n)$ is processed in four steps:

Step 1: The double-modulated signal, $u(n)$, is transformed into frequency spectrum $U(\omega)$ by FFT. $U(\omega)$ has three parts: LSB, carrier component $C(\omega)$, and USB.

Step 2: $U(\omega)$ is decomposed into $U_1(\omega)$ and $U_2(\omega)$ where $U_1(\omega)$ contains LSB and $C(\omega)$ and $U_2(\omega)$ contains $C(\omega)$ and USB. USB of $U_1(\omega)$ and LSB of $U_2(\omega)$ are equal to zero.

Step 3: $U_1(\omega)$ is then transformed into $u_1(n)$ and $U_2(\omega)$ into $u_2(n)$ by IFFT.

Step 4: Finally, $u_1(n)$ is demodulated by a product detector to extract $\hat{x}(n)$. Since $u_1(n)$ only contains one sideband, the standard-demodulated signal of $u_1(n)$ is multiplied by two to fully recover the original signal. Similarly, the extracted watermarked signal $\hat{y}(n)$ is obtained by multiplying the standard-demodulated signal of $u_2(n)$ by two.

4. **Evaluation.** We conducted computer simulations to evaluate the proposed data hiding scheme for AM radio signals. We evaluated the sound quality of original signals and

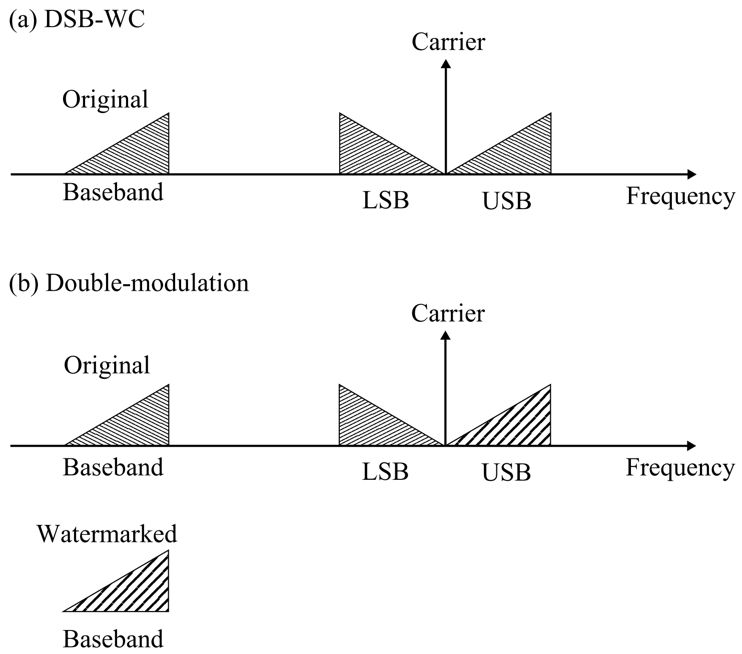

FIGURE 8. Differences between DSB-WC and double-modulation methods.

watermarked signals that were extracted from double-demodulation process. We investigated the sound quality of extracted watermarked signals with regard to the inaudibility and accuracy of watermark detection to confirm feasibility of applying a watermarking method based on CD to AM radio signals. In addition, we investigated the sound quality of demodulated signals that were output from of a standard demodulator to confirm the low-level compatibility of the proposed scheme with standard AM radio receivers. We used all 102 tracks of the RWC music database [24] as the original signals. These music tracks had a sampling frequency of 44.1-kHz, were 16-bit quantized, and had two channels. The carrier frequency was 250 kHz. The sampling frequency was 1000 kHz. The same watermark "JAIST-AIS" was embedded into the original signal. The data rates were from 4 to 1024 bps.

We used objective evaluations: the signal-to-error ratio (SER), log spectrum distortion (LSD) [25], and perceptual evaluation of audio quality (PEAQ) [26] to measure the sound quality of the target signals. SER was used to compare the level of a clean signal to the level of error. A higher SER signal indicated better sound quality. SER is defined in dB by

$$\mathrm{SER}(c,o) = 10\log_{10}\left(\frac{\sum_{n=1}^{N} c^2(n)}{\sum_{n=1}^{N}(o(n)-c(n))^2}\right) \qquad (\mathrm{dB}), \tag{11}$$

where $c(n)$ is the clean signal, $o(n)$ is the observed signal, and N is the number of samples.

LSD was used to measure the distance or distortion between two spectra. A lower LSD value indicates a better result. LSD is defined by

$$\mathrm{LSD}(C,O) = \sqrt{\frac{1}{K}\sum_{k=1}^{K}\left(10\log_{10}\frac{|O(\omega,k)|^2}{|C(\omega,k)|^2}\right)^2} \qquad (\mathrm{dB}), \tag{12}$$

where $C(\omega,k)$ and $O(\omega,k)$ are the short-time Fourier transform of the clean and observed signals, respectively, in which an overlap rate of 0.6 was used for this evaluation. k is the frame index and K is the number of frames.

TABLE 1. Quality degradation of sound and PEAQ (ODG).

Quality degradation	ODG
Imperceptible	0
Perceptible, but not annoying	-1
Slightly annoying	-2
Annoying	-3
Very annoying	-4

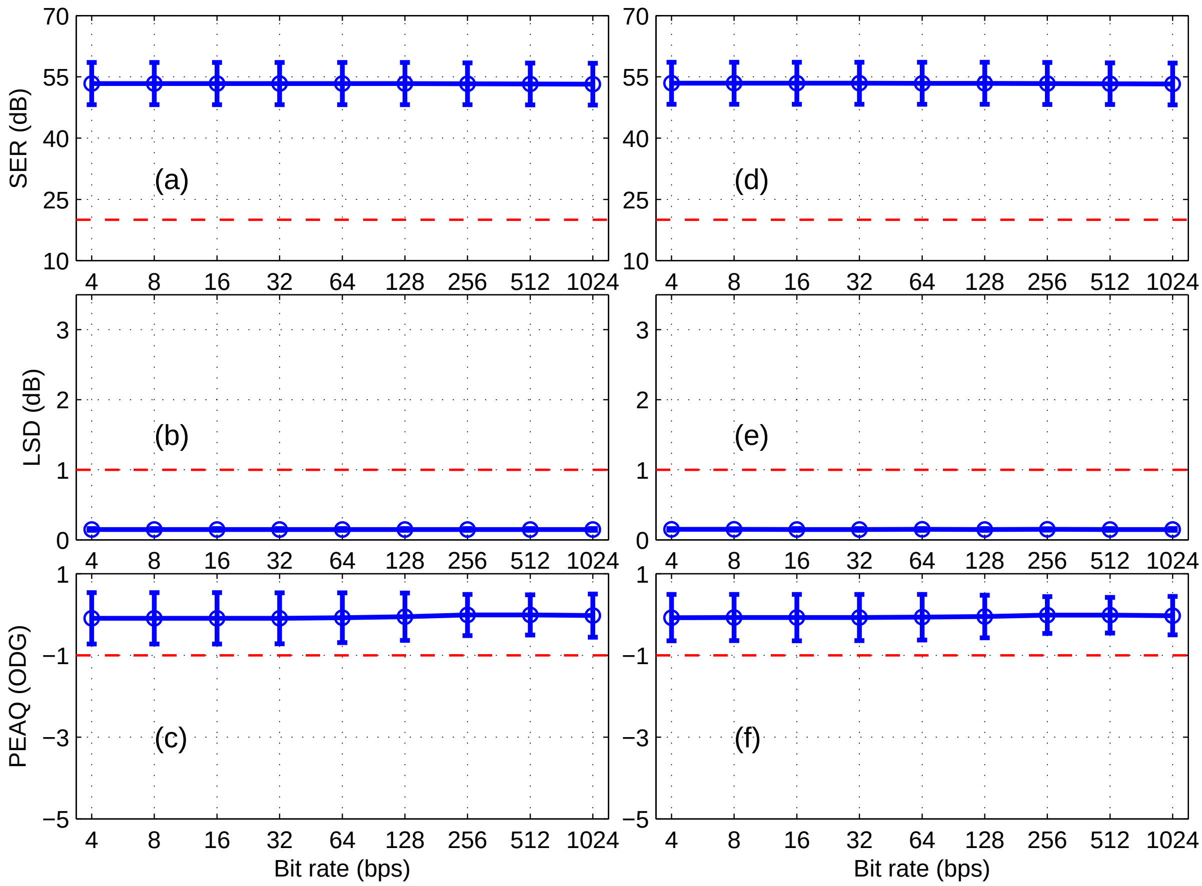

FIGURE 9. Results from objective evaluations of extracted signals as a function of bit rate: (a), (b), and (c) provide results for extracted original signals and (d), (e), and (f) provide results for extracted watermarked signals.

PEAQ is used to measure quality degradation in audio according to the objective difference grade (ODG) which ranges from -4 to 0. ODG indicates the sound quality of target signals as shown in Table 1.

Evaluation thresholds for the SER, LSD, and PEAQ corresponding to 20 dB, 1 dB, and -1 ODG, respectively, were chosen to evaluate the sound quality of the signals in these simulations.

The accuracy of watermark detection was measured by the bit detection rate, which is defined as the ratio between the number of correct bits and the total number of bits of the detected watermark. The evaluation threshold for the bit detection rate was 75%.

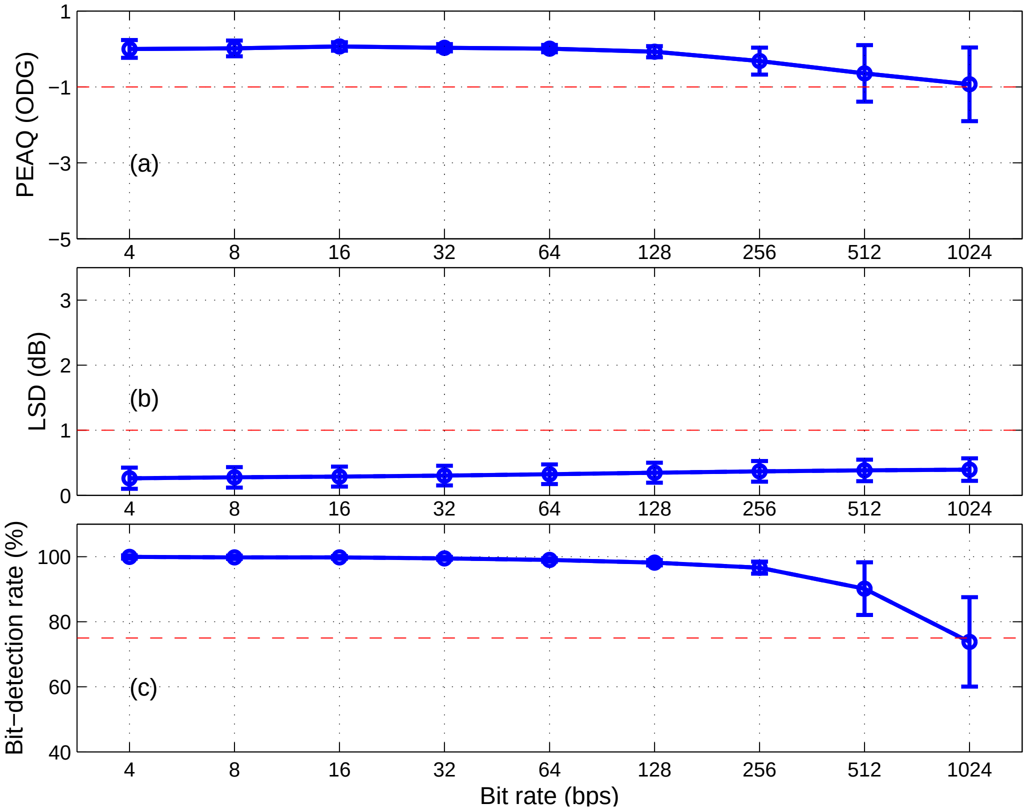

FIGURE 10. Results from objective evaluations of the sound quality of watermarked signals and bit detection rate as functions of the bit rate.

4.1. **Performance of hiding system.** The sound quality results for the signals extracted from the double-modulated signals are plotted in Fig. 9 as a function of bit-rate. When the bit-rate was 4 bps, which is a critical condition for the CD-based method of watermarking [14], the SER, LSD, and PEAQ of the extracted original and extracted watermarked signals corresponded to approximately 53.36 dB, 0.15 dB, and −0.09 ODG.

The SER and LSD had significantly high values in practice. The PEAQ was about −0.09 ODG, which is imperceptible, i.e., no different in the signals could be perceived. These results indicate that the extracted signals were not distorted by the double-modulation and double-demodulation process. When the bit-rate increased from 4 to 1024 bps, the SER, LSD, and PEAQ remained relatively unchanged. This confirms that the quality of original signals and watermarked signals conveyed in double-modulated signals is independent on the bit rate.

The sound quality with regard to the inaudibility of the watermarked signals extracted from the double-modulated signals and the accuracy of watermark detection with the signals extracted from the double-modulated signals are shown in Fig. 10. The bit-detection rate was greater than 99.5% when the bit-rate increased from 4 to 256 bps. It decreased dramatically when the bit-rates were 512 and 1024 bps. The PEAQs of the extracted watermarked signal with the extracted original signal were greater than −1 ODG and the LSDs were less than 0.5 dB. These results demonstrate that the bit-detection rate and inaudibility of watermarked signals with this scheme were the same as with the watermarking method. This indicates that the CD-based method of watermarking could be applied to the AM domain without any distortion and that our proposed scheme could be used to embed data into the AM audio signal and could precisely and robustly detect embedded data.

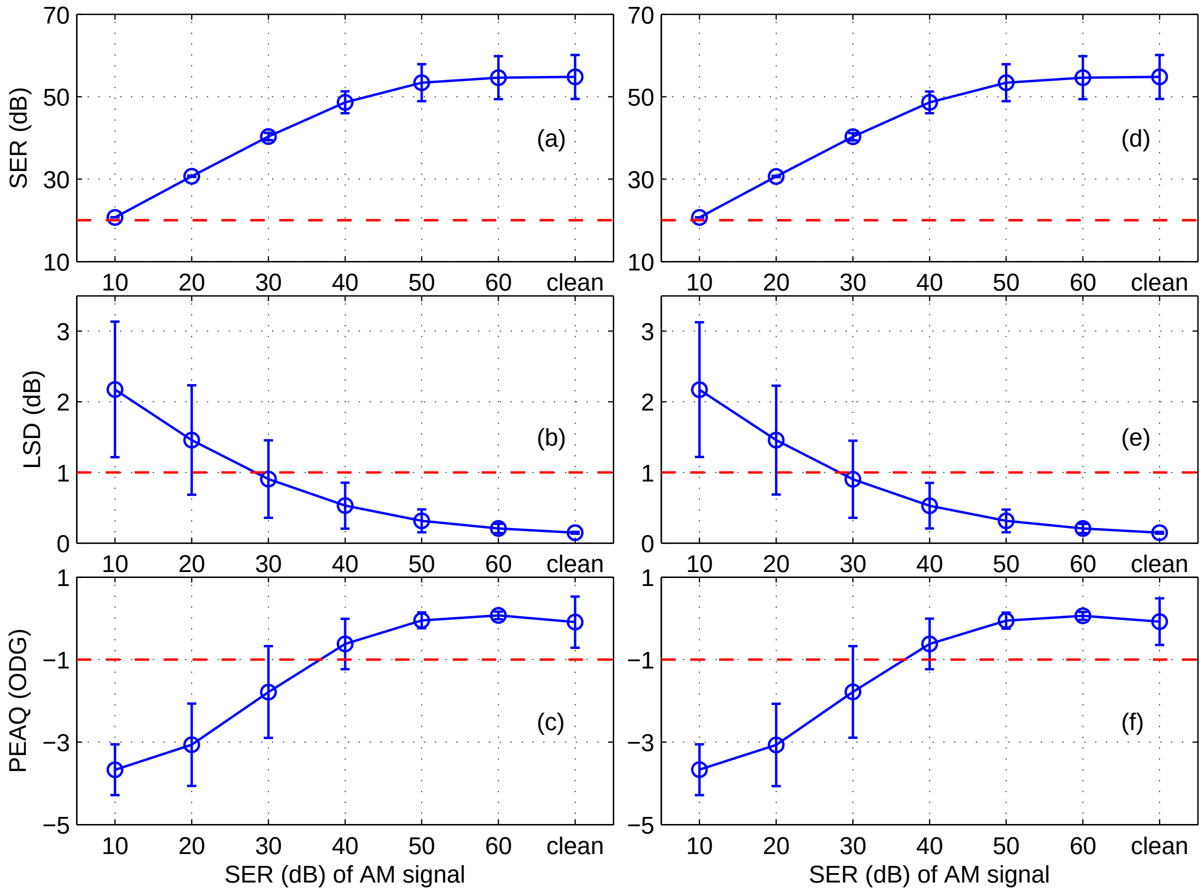

Figure 11. Results from objective evaluations of extracted signals against external noise: (a), (b), and (c) provide results for extracted original signals and (d), (e), and (f) provide results for extracted watermarked signals.

The double-modulated signal was transmitted through the air and may have been affected by external white noise. We examined the distortion of extracted signals when the double-modulated signal was subjected to white noise. Figure 11 plots the results from the objective tests of the extracted original and the extracted watermarked signals. The horizontal axis shows the SER of the double-modulated signal which indicates the level of noise. The extracted signals were most distorted when the noise level was high (SER $<$ 30 dB). However, when the noise level decreased (SER $\geq$ 30 dB), the SERs, LSDs, and PEAQs were significantly better ($\geq$ 40 dB, $\leq$ 0.9 dB, and ≥ -1.8 ODG). The bit-detection rates for high-level noise were less than 98.2% and for low-level noise (SER $\geq$ 30 dB) they were greater than 99.2%. These results indicated that the proposed scheme can robustly extract signals from a double-modulated signal that is affected by low-level noise.

4.2. **Low level compatibility.** A vast majority of AM radio receivers extract audio signals from AM radio signals by using standard AM techniques (the envelope detector and the product detector). The data-hiding scheme should produce a modulated signal that can be demodulated by standard AM radio devices. The difference between the two sidebands of the double-modulated signal is phase shift according to the CD characteristics. Therefore, if the difference in phase of the original and watermarked signals can be appropriately reduced, the double-modulated signal could be demodulated with the

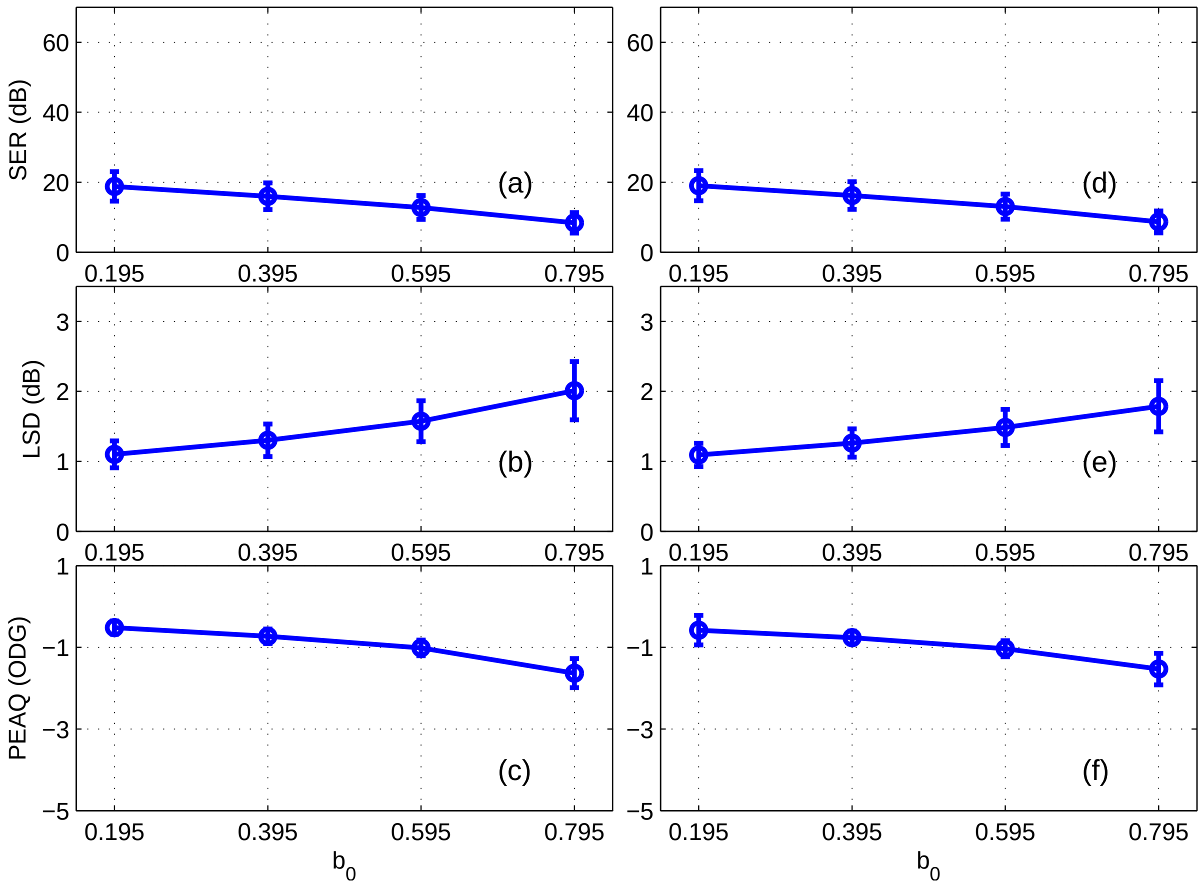

FIGURE 12. Sound quality of signals extracted using standard demodulators with respect to b_0: (a), (b), and (c) show results for a product demodulator and (d), (e), and (f) show results for an envelope demodulator.

standard technique with less distortion. Of course, the quality of the double-demodulated signal should not be degraded.

We utilized b_0 and b_1 with the CD-based method of watermarking to find the most suitable values for low-level compatibility with our proposed scheme. Figure 12 shows the sound quality of signals extracted by using a product demodulator and an envelope demodulator with respect to b_0, where $b_1 - b_0 = 0.07$ (the sufficient difference between b_0 and b_1 [14]). The results demonstrate that the sound quality of the standard-demodulated signals decreases as b_0 increases. The sound quality of demodulated signals using double-demodulation remains unchanged under these conditions. Thus, smaller values for b_0 and b_1 should be chosen. This experiment showed that $b_0 = 0.195$ and $b_1 = 0.265$ are the most suitable for the proposed scheme.

The proposed double-modulation modulated the carrier signal with the original signal and the watermarked signal as LSB and USB, respectively. In addition, it can modulate the carrier signal in reserve with the original signal and the watermarked signal as USB and LSB without affecting the performance. We also carried out experiments under the reverse condition and the results were the same as those under the normal condition.

5. **Conclusions.** To enable a more efficient emergency alert system as well as high utility AM radio service, this paper proposed a data-hiding scheme for AM radio broadcasting systems. The proposed scheme can be used to transmit additional digital information along side audio content in AM radio signals. Playing an important role in this scheme,

the digital audio watermarking method based on CD was used to embed an inaudible message into audio content before the audio was further processed for long-distance transmission over a radio link. The CD-based method is a non-blind watermarking scheme which requires a double transmission bandwidth, but we overcame this problem by developing novel double-modulation and double-demodulation algorithms. The double-modulation modulates the carrier signal with both the original and watermarked signals as LSB and USB, respectively. Although the proposed scheme generates AM radio signals having different sidebands in their frequency spectra, standard receivers can still extract audio content from the AM signals using standard demodulation techniques (the product demodulator and envelope demodulator).

We conducted computer simulations to evaluate the effectiveness of the proposed scheme. We first confirmed the effectiveness of the double modulation and double demodulation processes by measuring the sound quality of audio signals that were extracted from the AM signals. The SER, LSD, and PEAQ results showed that these signals could be properly extracted. These results confirmed the feasibility of the proposed double-modulation and double-demodulation by showing that we can precisely extract both the original and the watermarked signals from LSB and USB. Second, we evaluated the inaudibility of watermarked signals and the accuracy ofthe watermark detection process. The LSD and PEAQ results confirmed that the CD-method could be used with the proposed scheme to embed inaudible message. The bit-detection rate results revealed that the watermark detection accuracy could be kept as high as that in the CD-based watermarking system for digital-audio. Third, we evaluated the sound quality of the signal extracted from the AM signal using standard demodulators to check the compatibility of the proposed data-hiding scheme. We found that the standard receivers could acquire audio content at a reasonable level of distortion. Finally, we looked at the distortion of the extracted signals caused by external white noise. The scores showed that the sound quality of the extracted signals was degraded when the the external noise level was relatively high (SER < 30 dB).

Compared to conventional techniques, such as AMSS, for embedding additional digital information into AM radio signals, the proposed method offers better embedding capacity and compatibility. The embedding capacity of the proposed scheme is about 512 bps while that of AMSS is 47 bps. Conventional radio devices can demodulate modulated signals to extract audio signals in the proposed scheme, but will not respond to the modulated signals in AMSS.

The proposed scheme can be used to develop a hidden-message AM radio system. Such a system can be used as an emergency alert system in the event of natural disasters. Moreover, it can broadcast additional digital information as part of radio services such as programs, weather forecasting, news, advertisements, etc., thus providing high utility AM radio service.

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