Research Article

Noncoherent Low-Frequency Ultrasonic Communication System with Optimum Symbol Length

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A noncoherent low-frequency ultrasonic (LFU) communication system is proposed for near-field communication using commercial off-the-shelf (COTS) speakers and microphones. Since the LFU communication channel is known to be a frequency-selective characteristic, the proposed system is basically designed by differential phase-shift keying (DPSK) modulation with forward error correction. In addition, automatic gain control of the carrier frequency band over the LFU communication channel is proposed. Then, in order to optimize the symbol length of the proposed LFU communication system under a realistic aerial acoustic channel, a propagation model of the LFU communication channel is proposed by incorporating aerial acoustic attenuation. The performance of the proposed LFU communication system is demonstrated on two different tasks: bit error rate (BER) measurement and successful transmission rate (STR) comparison with Google Tone for various distances between the transmitter and the receiver. Consequently, the proposed method can operate without a bit error at a distance of 8 m under various noise conditions with sound pressure level of 80 dB. Moreover, the proposed method achieves higher STR than Google Tone on a task of URL transmission using two laptops.

1. Introduction

Nowadays, everyday devices are evolving to have connectivity owing to the advancement of short-range wireless communication technology, such as near-field communication (NFC) [1] and Bluetooth low energy (BLE) [2]. Both NFC and BLE have been successfully applied to short-range communication-based services, including electronic payment or location-based advertisement via smartphones. However, compatibility issues exist in NFC and BLE, since they require corresponding hardware transceiver modules, which are rarely embedded in most devices except recent smartphones or tablets. For this reason, alternative short-range communication technology that does not rely on a hardware transceiver is in demand for the improved connectivity between various devices. Considering the improved connectivity of short-range wireless communication, acoustic communication can be a suitable alternative to the conventional systems, since acoustic communication transmits and receives data through a speaker and a microphone, which are typically implemented or easily attached to most digital devices.

Acoustic communication can be separated into underwater and aerial acoustic communication. First, underwater acoustic communication has been developed for autonomous underwater vehicles, such as submarines or underwater sensors, in order to monitor marine ecosystems or underwater military action [3, 4]. Since underwater acoustic communication delivers sound waves through water, it requires a special form of microphone and speaker that operate underwater (i.e., hydrophones and underwater speakers). In contrast to underwater acoustic communication, aerial acoustic communication utilizes airborne sound waves to transmit or receive data [5–8]. The operation frequency for aerial acoustic communication can be included in an audible range (50 Hz–18 kHz) [6–8] or within a low-frequency ultrasonic (LFU)
range (above 18 kHz) [5, 8], which can be covered by commercial off-the-shelf (COTS) speakers and microphones. In other words, aerial acoustic communication can be applied to any audio interface-equipped devices without consideration of the type of hardware or operating system. For this reason, aerial acoustic communication is compatible with most devices with either a speaker or a microphone, and such a characteristic is highly favorable for short-range wireless communication that requires connectivity between various devices.

Recently developed aerial acoustic communications can be categorized into acoustic communication with data hiding and LFU communication followed by their data transmission channels, which are the audible frequency channel (50 Hz–18 kHz) and LFU channel (above 18 kHz), respectively. Owing to the fact that the audible frequency channel is much less frequency-selective than the inaudible one, higher communication performance in terms of bandwidth capacity can be expected compared with LFU communication [6, 7]. However, modulation signals that have carrier frequencies within the audible frequency band can be perceived as disturbing noise to users who are in the communication zone. To avoid such unwanted noise, the data carrier signals need to be masked with audio signals through data hiding techniques based on acoustic orthogonal frequency division multiplexing (AOFDM) or modulated complex lapped transform (MCLT) [6, 7]. In contrast, LFU communication has a vital advantage over audible frequency channel-based communication, since it does not require data hiding techniques and its modulated signals with the LFU band are barely noticeable even with excessive loudness [5]. Nonetheless, the LFU channel is considered a highly frequency-selective channel because most COTS microphones and speakers have severely selective frequency response over 21 kHz [5]. Thus, LFU communication requires a careful modulation scheme to cope with the limited performance.

Some LFU communication systems have been developed to meet their own communication performance considering COTS microphones and speakers. First, an amplitude shift keying- (ASK-) based LFU communication system, which was named Sonicom, was introduced as a concept of a wireless dial-up modem [8]. Sonicom simply modulated binary 1 into a sinusoid with the carrier frequency of 18.3 kHz with a certain energy level or just sent silence for binary 0. In addition, a preamble was attached prior to a data sequence to synchronize the transmitter with the receiver using a matched filter. The transmission speed of Sonicom was 1.47 kbps, which was satisfactory for applications regarding short-range communication, such as sending the uniform resource locator (URL) of a web page or the service set identifier (SSID) of Wi-Fi. However, Sonicom was highly vulnerable to attenuation and channel noise because both synchronization and modulation relied on the energy level of the sinusoids at a certain carrier frequency; thus, its maximum transmission range was limited to 2 m. To improve the robustness of LFU communication, a recent approach applied chirp signal as a modulation symbol. Specifically, the chirp signal-based LFU communication utilized the LFU band of 19.5–22.2 kHz to generate a single chirp symbol, where binary 1 was represented with increasing carrier frequencies and binary 0 was with decreasing ones. The preamble consisted of up and down chirp symbols and it was then attached prior to a data sequence. In addition, both synchronization and demodulation were conducted by matched filtering using the fast Fourier transform (FFT). Owing to the good autocorrelation characteristics of the chirp symbol, the chirp signal-based communication system could transmit data up to 25 m in a realistic indoor environment, but its transmission speed was measured as 16 bps [5]. Such an exceedingly slow speed was due to the poor modulation efficiency of the chirp symbol in terms of both time and frequency. Therefore, further improvement of the LFU communication system is required to support robust data transmission in a realistic indoor environment with an acceptable speed and communication range.

Recently, Google released an end-user service named “Google Tone,” which broadcasted URLs from a device to others using the LFU communication method [9]. Owing to the attenuation of LFU through typical building barriers such as walls, doors, and windows, Google Tone can easily share URLs between devices located in the same room without manual exchange of password for connecting a common network through Wi-Fi or Bluetooth [10]. Such connection between near devices can be vastly utilized not only for data sharing, but also for home automation [10] or indoor positioning systems [11], if the LFU communication performance would be reliable according to various characteristics of COTS microphones and speakers as well as operating environments.

In this paper, a noncoherent LFU communication system is proposed for near-field communication using COTS microphones and speakers. The proposed system is based on a noncoherent communication framework without using the synchronization that often fails to perform under the highly frequency-selective channel [12]. This is because the LFU channel is highly frequency-selective, caused by the limited performance of COTS microphones and speakers in realistic noisy environments. Owing to the noncoherent framework, the proposed system simply consists of forward error correction (FEC), modulation, and automatic gain control (AGC) modules. To be specific, the perfect binary Golay code, which has the capability of 3-bit error correction and 7-bit error detection [13], is applied as the FEC module. Moreover, differential phase-shift keying (DPSK) modulation/demodulation is used for the transmission and detection of the symbol in a noncoherent way. Finally, AGC is applied to both the modulation and the demodulation to make the proposed system robust against the noisy and frequency-selective channel. In addition, a propagation model for the LFU channel is defined and utilized to find the optimum symbol length of the proposed system.

The rest of this paper is organized as follows. Following this introduction, Section 2 proposes a DPSK-based noncoherent LFU communication system. Next, Section 3 defines a propagation model of the LFU channel in order to find the optimum symbol length of the proposed LFU communication system. Section 4 evaluates the performance of the proposed system and compares it with Google Tone. Finally, the paper is concluded in Section 5.
2. The Proposed Noncoherent LFU Communication

In this section, the noncoherent LFU communication system is proposed for robust communication with COTS microphones and speakers in a noisy environment. Figure 1 shows the proposed noncoherent LFU communication system that is composed of the LFU transmitter and the LFU receiver.

2.1. LFU Transmitter. The proposed system transmits binary data through the LFU wave with three processing steps, which are FEC encoding, DPSK modulation, and the AGC of the carrier frequency band. To this end, \( P \) binary bits to be transmitted at the \( i \)th symbol are grouped as \( \mathbf{b}_i = [b^{(1)}_i \cdots b^{(P)}_i] \), where \( P \) is set to 24 in this paper. Next, \( \mathbf{b}_i \) is encoded by the FEC algorithm, and additional error correction bits are attached. Specifically, a perfect binary Golay code, \( G_{23} \), is applied for the FEC encoding [13]. In other words, \( \mathbf{b}_i \) is reshaped into an \( O \times 12 \) matrix, and it is encoded by the \( (23, 12) \) Golay block code; thus, the \( O \times 23 \) encoded block for the \( i \)th symbol, \( \mathbf{b}_n(i) \), is generated. Notice here that \( O = 2 \) and the coding rate, \( R_c \), of \( G_{23} \) is 52.17\%. In fact, \( G_{23} \) has 3-bit error correction and 7-bit error detection, and this means that the error bits of maximally 12.5\% per symbol can be neglected in the proposed system. It is known that such error correction in radio frequency communication is highly beneficial for robust communication in a highly frequency-selective channel [14].

After the FEC encoding is completed, each encoded bit of \( \tilde{b}_i \) is modulated by a DPSK scheme. To this end, \( \tilde{b}_i \) is reshaped into a vector as \( \tilde{\mathbf{b}}_i = [\tilde{b}^{(1)}_i \cdots \tilde{b}^{(Q)}_i] \), where \( Q \) is 46 in this paper. Next, differential encoding is conducted on each \( \tilde{b}^{(q)}_i \) as [15]

\[
\tilde{d}^{(q)}_i = \tilde{b}^{(q)}_i \oplus \tilde{d}^{(q)}_{i-1},
\]

where \( d^{(q)}_i \) is the \( q \)th differentially encoded bit at the \( i \)th symbol. Moreover, \( \oplus \) indicates the bitwise exclusive operation. \( d^{(q)}_i \) is then modulated in the form of sinusoidal signals by the windowed DPSK, \( c_i(n) \), as

\[
c_i(n) = w(n) \cos \left( \frac{2\pi f_c n}{f_s} + \pi (1 - d^{(q)}_{i}) \right),
\]

where \( n \) indicates a sample index from 1 to a symbol length, \( T_b \). Moreover, \( f_c \) and \( f_s \) are carrier and sampling frequencies, respectively. Here, \( f_s \) is set to 48 kHz to support the frequency range of most COTS microphones and speakers. Accordingly, \( f_c \) can be set from 18 to 24 kHz, and it is set to 20 kHz in this paper. In addition, \( w(n) \) is a squared Hanning window, which smooths the discontinuity between subsequent symbols to avoid unwanted impulsive noise at the modulated signals. Note that \( T_b \) must be chosen carefully to maintain phase continuity between subsequent symbols. To this end, \( T_b \) is obtained by the following equation:

\[
T_b = N \frac{f_s}{\gcd(f_c, f_s)},
\]

where \( N \) is an arbitrary natural number and \( \gcd(f_c, f_s) \) indicates the greatest common divisor between \( f_c \) and \( f_s \). It should be mentioned that \( T_b \) is closely related to the communication performance of the proposed system, since \( T_b \) controls the data transmission rate and communication robustness for various channel effects [15]. For this reason, it is important to find the optimum \( T_b \) of the proposed communication system. A strategy for finding the optimum \( T_b \) with the LFU propagation model is described in the next section.

Followed by the modulation, gain normalization of the carrier frequency band is conducted. First, since \( c_i(n) \) might have unwanted nonlinear harmonic components due to
discontinuity between adjacent time samples, an \( M \)-tap bandpass filter, \( g(n) \), is applied to \( c_i(n) \), such as

\[
\tilde{c}_i(n) = c_i(n) * g(n),
\]

where \( g(n) \) is designed to have the lower and upper cutoff frequencies of \( f_c - f_b \) and \( f_c + f_b \), respectively. Here, \( f_b \) corresponds to the half of the carrier frequency bandwidth, and it is obtained as

\[
f_b = \frac{f_c}{T_b}.
\]

Next, the filtered modulated signal, \( \tilde{c}_i(n) \), is normalized by using the gain on the carrier frequency band as

\[
s^T_b(i) = \frac{G\tilde{c}_i(n)}{(1/T_b)\sum_{n=1}^{T_b} |\tilde{c}_i(n)|},
\]

where \( G \) is a normalization scale and it is set as 2\(^{14} \) in this paper, which corresponds to the half of the maximum value of the sample with 16-bit resolution. Finally, the modulated LFU signals of the \( i \)th symbol, \( s^T_b(i) \), are transformed into analogue signals by a digital-to-analogue converter (DAC), and then they are spread into the aerial medium by a COTS loudspeaker.

2.2. LFU Receiver. In the LFU receiver, the input signal recorded by the COTS microphone, \( y_i(n) \), is the superposition of \( s^T_b(i) \) in (6) and additive environmental noise, \( z_i(n) \), as

\[
y_i(n) = h(n) * s^T_b(i) + z_i(n),
\]

where \( h(n) \) indicates the impulse response of the propagation channel from the LFU transmitter to the receiver. To estimate \( s^T_b(i) \) from \( y_i(n) \), the carrier frequency bandpass filter, \( g(n) \), is again applied to \( y_i(n) \) as

\[
\tilde{y}_i(n) = y_i(n) * g(n).
\]

In (8), \( z_i(n) \) in \( y_i(n) \) is filtered out, owing to the fact that \( z_i(n) \) rarely overlaps with \( s^T_b(i) \) at the carrier frequency band [5, 8]. Similar to (6), the filtered receiver signal, \( \tilde{y}_i(n) \), is normalized to compensate for energy loss due to the frequency-selective characteristics of \( h(n) \), as

\[
\tilde{c}_i(n) = \frac{G\widetilde{c}_i(n)}{(1/T_b)\sum_{n=1}^{T_b} |\widetilde{c}_i(n)|},
\]

where \( \widetilde{c}_i(n) \) is the estimate of \( b^T_i \) at the \( i \)th symbol. Next, the DPSK demodulation is performed by using noncoherent detection [15] as

\[
\hat{b}^T_i(j) = \begin{cases} 1, & \text{if } \sum_{n=1}^{T_b} \tilde{c}_i(n) \tilde{c}_{i-1}(n) > 0 \\ 0, & \text{otherwise} \end{cases}
\]

Finally, the \( p \)th binary bit at the \( i \)th symbol, \( \hat{b}^T_i(j) \), is obtained by conducting FEC decoding with \( G_{23} \) on \( \hat{b}^T_i(j) \) [13].

3. Optimization of the Proposed LFU Communication System with Aerial Acoustic Attenuation

This section describes how to optimize the proposed system by incorporating the propagation model. In this work, the propagation model of the LFU channel is defined by using aerial acoustic attenuation and background noise. The following subsections discuss the propagation model and the optimization procedure in detail.

3.1. Aerial Acoustic Attenuation Model of LFU Channel. The propagation channel at the \( k \)th frequency bin for the proposed system, \( H(k) \), is based on the aerial acoustic attenuation of the LFU channel. First, the acoustic attenuation of the LFU channel, \( A(d, k) \), is defined by the distance between the transmitter and the receiver, \( d \), and its \( k \)th frequency bin corresponding to the carrier frequency band of the LFU channel. In other words, the propagation channel at the \( k \)th frequency bin is defined as [16]

\[
H(k) \equiv A(d, k) = A(d) \exp\left(\eta \alpha d^\eta\right),
\]

where \( g \) is a spreading factor reflecting the geometry of propagation. Here, \( g \) is set to \( 2 \) in this paper since it represents spherical wave propagation from a point source [17]. In order to set the parameters in (11) such as \( A_\alpha, \alpha_0, \) and \( \eta \), a nonlinear least-square curve-fitting technique [18] is applied to pairs of \( s^T_b(i) \) in (6) and \( H(k) \) in (11) according to the different distance, \( d \), from 0 m to 20 m. Consequently, we have \( A_\alpha = 10.6264, \alpha_0 = 0.0007, \) and \( \eta = 0.5656 \). Note that the model of the COTS microphone was the microphone built in a Roland R-05 portable recorder and the model of the COTS speaker was Genelec 6010B. Next, for given \( d \) and \( T_b \), the performance of the attenuation model in (11) is measured using four different measures such as signal-to-noise ratio, SNR\((d)\), capacity, \( C(d) \), energy per bit, \( E_b(d) \), and noise power spectral density, \( N_0 \), which are defined as [19]

\[
\text{SNR}(d) = \frac{1}{I} \sum_{i=1}^{I} \left( \frac{T_b}{\sum_{k=f_c-f_b}^{f_c+f_b} \left( \frac{A(d, k)+1}{Z_i(k)} \right)^2 \left( A(d, k) \right)^2} \right)^2,
\]

\[
C(d) = \frac{1}{I} \sum_{i=1}^{I} \left( \frac{T_b}{\sum_{k=f_c-f_b}^{f_c+f_b} \left( \frac{A(d, k)+1}{Z_i(k)} \right)^2 + 1} \right),
\]

\[
E_b(d) = \frac{1}{C(d)} I \sum_{i=1}^{I} \left( \frac{T_b}{\sum_{k=f_c-f_b}^{f_c+f_b} \left( A(d, k) \right)^2} \right),
\]

\[
N_0 = \frac{1}{2f_b} I \sum_{i=1}^{I} \sum_{k=f_c-f_b}^{f_c+f_b} |Z_i(k)|^2,
\]

where \( |S^T_b(k)|^2 \) and \( |Z_i(k)|^2 \) are the power spectra of \( s^T_b(i) \) and \( z_i(n) \), respectively. In addition, \( I \) is the total number of
symbols evaluated in this performance analysis. As shown in (12), the performance of the aerial acoustic attenuation model for the proposed LFU communication system is evaluated by varying distances and SNRs.

3.2. Symbol Length Optimization Incorporating Attenuation Model. This subsection discusses a strategy for finding the optimum \( T_b \) for a given \( d \). As described in the previous subsection, aerial acoustic attenuation is modelled by \( A(d, k) \) in the frequency domain. Thus, the recorded signal at the LFU receiver can be represented by

\[
y_n = a(d, n) * S_n + z_n,
\]

where \( a(d, n) \) is the impulse response of \( A(d, k) \) and \( y_n \) is the recorded signal for a given distance \( d \). A strategy for finding the optimum \( T_b \) is to select \( T_b \) in which the lowest bit error rate (BER) is provided. To this end, \( \tilde{y}_i^{d,T_b} \) for each \( T_b \) is first demodulated into \( \tilde{x}_i^{d,T_b} \), as described in Section 2.2.

Then, BER for \( \tilde{x}_i^{d,T_b} \) is measured as

\[
\text{BER}_{i,d} = \frac{1}{\text{#symbol}} \sum_{i=1}^{\text{symbol}} e^{-d,T_b} \text{BER}_{i,d} = \frac{1}{\text{#symbol}} \sum_{i=1}^{\text{symbol}} e^{-d,T_b} \frac{1}{\text{#symbol}} \sum_{i=1}^{\text{symbol}} e^{x_i^{d,T_b}dx},
\]

where \( \text{BER}_{i,d} \) indicates the BER averaged over all the \( I \) symbols. Next, the optimum \( T_{b,\text{opt}} \), \( T_{b,\text{opt}} \), is selected from all the candidates of \( T_b \) in (3), where average BER for all possible distances from 0 to \( D \) becomes the lowest, such as

\[
\tilde{T}_b = \arg \min_{T_b} \frac{1}{D} \int_0^D e^{-x,T_b}dx,
\]

where \( D \) is the maximum distance.

4. Performance Evaluation

In order to evaluate the performance of the proposed LFU communication system, a BER and a successful transmission rate (STR) were measured. In particular, the BERs for different aerial acoustic channels were measured by varying the distance, \( d \), and noise, \( z(n) \). In addition, an STR from the proposed LFU communication system was compared with that from Google Tone by counting the number of successfully received URLs from all the transmitted URLs, where two laptops were used for the COTS microphone and speaker.

4.1. Selection of Optimum Symbol Length. In order to select the optimum symbol length, \( \tilde{T}_b \), the proposed LFU communication system was first implemented by setting \( D = 20 \) m and discretizing \( d \) at a step of 0.2 m for computing (15). Then, the candidates of \( \tilde{T}_b \) were obtained from (3) such that they were \( \{12, 24, 36, \ldots \} \) since \( f_s = 48 \) kHz and \( f_{c} = 20 \) kHz. In this experiment, noise, \( z(n) \), was recorded at an office by using a microelectromechanical system (MEMS) microphone embedded in a smartphone (Galaxy S6), where the average sound pressure level (SPL) of noise was measured as 50 dB. Note here that noise was also sampled at a rate of 48kHz so that it overlapped with the carrier frequency of the proposed system. Figure 2 shows the BER according to different \( \tilde{T}_b \) from 12 to 288. As shown in the figure, the BER was the lowest when \( \tilde{T}_b = 96 \). Thus, the optimum symbol length was fixed as 96 in the following experiments.

4.2. BER Performance. In order to measure BER performance of the proposed system, 50 randomly chosen characters were first encoded by the LFU transmitter with \( \tilde{T}_b = 96 \) as described in Section 4.1. Next, the encoded characters were passed through the aerial acoustic channel modelled by (13). Here, the distance, \( d \), was also discretized at a step of 0.2 m. In addition, three different background noises for \( z(n) \) were recorded using a smartphone (Galaxy S6) at bus stop, cafe, and street with average SPL of 90, 80, and 85 dB, respectively. Figure 3 shows power spectral densities (PSDs) of three different noises and LFU signal at 1 m distance. As shown in the figure, noises interfered with all frequency bands including the LFU band; however, the PSD of the LFU signal had around 40–60 dB higher spectral peak than that of noise.

Next, the received signal was decoded back by the LFU receiver, and the BER was measured by comparing the URLs at the transmitter with those at the receiver. Note that, for a given noise and distance, 20 times of LFU communications were repeated to get a statistically reliable BER. Figure 4 shows the BER according to the distance under three different noise conditions. As shown in the figure, the BER went lower as the distance was longer, as we expected. However, the BER was lower than \( 10^{-4} \) until the distance was shorter than 8, 10, and 12, for bus stop, cafe, and street noise, respectively. This implies that the proposed LFU system can be used for transmission in a room area network.

4.3. STR Comparison with Google Tone. The performance of the proposed LFU communication system was compared...
with that of a state-of-the-art system, Google Tone [9]. In this experiment, two laptops having both embedded microphone and speaker (Dell XPS 15 9550 and Dell Inspiron 15 7559) were placed in a meeting room with distances of 1, 2, and 4 m. Next, 100 different URLs were sent from one laptop to the other at each distance by using Google Tone and the proposed system. While sending the URLs, the number of successful transmissions was counted. This experiment was performed under office and TV noise condition to observe the effect of background noise on the STR of Google Tone and the proposed system. Here, TV noise was generated by playing video contents through a TV speaker in the meeting room with three different SPLs of 60, 70, and 80 dB.

Figure 5 compares the STR of the proposed LFU communication system with that of Google Tone under office and TV noise condition, where the distance was set to 1, 2, and 4 m. As shown in the figure, the proposed system had higher STR than Google Tone for all the noise conditions and the three distances. Specifically, both Google Tone and the proposed system had similar STRs at 1 m for office and all the TV noise conditions. In addition, the STR of Google Tone was drastically lowered at 2 m for TV noise conditions, whereas the STR reduction of the proposed system was marginal at 2 m, compared to that at 1 m. However, both STRs of Google Tone and the proposed method were pretty low at 4 m. This was because the frequency response of the microphone and/or speaker inside the laptops was frequency-selective.

5. Conclusion

This paper proposed a noncoherent LFU communication system for short-range communication with COTS
The proposed system consisted of FEC coding, windowed DPSK modulation, and gain normalization of the carrier frequency band in order to make the proposed system robust over the LFU channel. In particular, a propagation model of the LFU channel was proposed by incorporating aerial acoustic attenuation, and it was used for symbol length optimization. The BER performance of the proposed LFU communication system was measured according to the distance under different noise conditions. As a result, the proposed LFU communication system could operate up to around 8 m without bit error even at high ambient noise of 80 dB or above. Finally, the STR performance of the proposed system was compared with that of Google Tone. Consequently, it was shown that the proposed system achieved higher STR on a task of URL transmission under TV noise conditions.

**Competing Interests**

The authors declare that they have no competing interests.

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