AERIAL ACOUSTIC COMMUNICATION FOR LONG-RANGE APPLICATIONS BASED ON SINGLE-CARRIER FREQUENCY DOMAIN EQUALIZATION

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ABSTRACT

Conventional airborne wireless data transmission system is constrained for applications within the distance of ten meters. However, for some applications such as in electromagnetic shielding pipelines, saps, rooms or hallways, the receiver may be located in a certain distance away from the transmitter where both reverberation and background noise need to be considered. An aerial acoustic communication method using transducer arrays is presented for long-range circumstances. Signal-to-noise ratio can be enhanced by the proposed method based on single-carrier frequency domain equalization algorithm. Severe inter-symbol interferences caused by reflected waves of various boundaries can also be reduced. Experimental results carrying out along hallway demonstrated its reliability in the distance expanded to one hundred meters.

Index Terms**—**Acoustic transducers, Wireless communication, Channel estimation, Acoustic applications, Acoustic signal processing

1. INTRODUCTION

Effectiveness of electromagnetic telecommunication is limited by Faraday shielding effects in some applications such as in saps. Acoustic communication promotes the possibility of overcoming this problem. Researches can be dated back to 1956 when Rosen [1] raised the concept of "Ceramic transformer". With the development of sonar technology, underwater acoustic communication has become a matured technology field [2,3]. On the other hand, data transmission via sound waves through air is still being developed. Aerial acoustic communication can be accomplished using airborne ultrasound due to its high directivity. Its applications can generally be found in devices used for location positioning and tracking in hospital [4,5], wireless computer keyboard [6], etc. Ultrasound signals can also readily propagate through metal barriers, so it is useful in applications where through-wall data transmissions are needed in metal enclosures [7-9]. For applications on audible carrier waves, sound is used to safely transmit preauthentication information, steganography, or for entertainment [10]. Jurdak et al. implemented this system to monitor pollution indicators within a distance of ten meters [11]. These airborne wireless data transmission systems are mainly focused on short-range applications. However, for long-range applications such as those found in electromagnetic shielding pipelines, saps, rooms or hallways, the adverse effects caused by background noise and reverberation can degrade the performance of acoustic communication.

This paper presents a new acoustic data transmission system for long-range applications. Combining digital communication algorithm with custom-designed transducer arrays, the proposed system is achieved to enhance the signal-to-noise ratio (SNR), which makes the system suitable for long distance acoustic amplification. Single-carrier frequency domain equalization (SC-FDE) [12] algorithm is used to reduce inter-symbol interferences (ISI). The overall system performance is evaluated by carrying out experimental tests and the mechanism for selecting optimal parameters is discussed. The remainder of this paper is organized as follows. Section 2 describes hardware design of the acoustic data transmission system. Custom-designed transducer arrays are adopted to improve system performance for long-range applications. Section 3 presents the signal processing algorithm, while Section 4 shows the experimental results for long hallway and outdoor tests. The conclusion is drawn in Section 5.

2. SYSTEM CONFIGURATION

The main advantage of this proposed system is the distance available for data transmission. The sound wave information can be transmitted over a considerably long distance expanded to hundreds of meters. Multi-cellular horn loudspeaker array with several single horn as shown in Fig. 1 is utilized to enhance power amplification efficiency and to achieve large-scale sound reproduction. Forty-four exponential horn loudspeakers are combined to form the loudspeaker array terminal which serves as a transmitter. Contrariwise, higher sound pressure can be achieved at the throat of a horn while its wide end is struck by sound waves [13], allowing the transducer array to also act as a receiver. With signal detection and amplifying circuits, sensitivity and

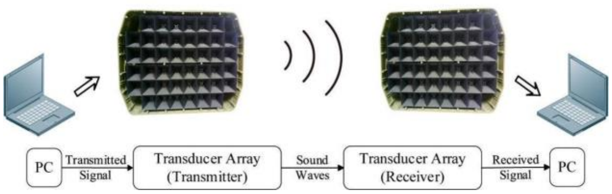

Fig. 1. System hardware configuration.

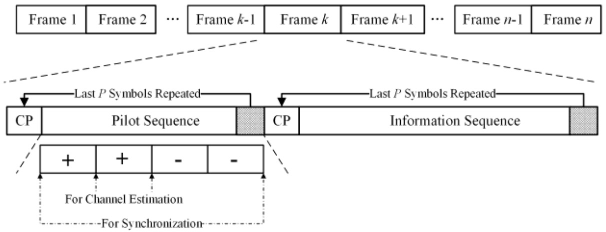

Fig. 2. The frame structure of the acoustic data transmission system.

reliability of the whole system are further enhanced.

To make the prototype of the entire system more versatile in use, system hardware complexity plays an importance role. Both the simulation signals and the off-line signals were processed on personal computers for testing the performance of the system. Fig. 1 illustrates the main structure of the acoustic data transmission system. Notebooks were used as central controllers, as well as playing and recording sound. Information was encoded and played via PC soundcard. Amplified by loudspeaker array, signals with information were sent out as audible waves. In the receiving terminal, these audible waves were recorded, and then filtered by single-carrier frequency domain equalizer. Estimation of information was obtained after decoding.

3. ALGORITHM DESIGN

In the proposed data transmission system, input binary source code word is generated by acoustic source encoder from information source in the form of voice, picture, text, etc. Quadrature amplitude modulation (QAM) is adopted for its high modulation efficiency [14]. In order to overcome the effects of noise and interference, convolutional codes with memory are added as channel encoder before the digital modulator [15]. For every single bit input in the frame, two convolutional code output bits are shifted out. With extra degree of redundancy under certain schemes, error control of the system becomes more effective. The output sequence depends on the previous data stored in the six shift registers. After the channel encoding and digital modulating, primary transmission signal is formed. Divided into several symbol blocks as information sequences, the system is more tolerant to noise variance.

The frame structure of the proposed acoustic data transmission system can be represented as in Fig. 2, each frame block contains pilot sequence for both synchronization and channel estimation. The pilot sequence defined by Minn's algorithm [16] is in the form of $[+A+A-A-A]$, whereby each quarter denoted as A is a pseudonoise (PN) sequence. Acted as a guard interval between blocks to resist ISI, cyclic prefix (CP) is added for each small fast Fourier transform (FFT) block in SC-FDE algorithm. It is inserted in the transmitter and is then removed in the receiver. A typical CP form is produced by replicating the last several symbols in each data sequence.

A root raised cosine (RRC) filter is implemented in the transmitter. The frequency characteristic of the filter is given as

$$G_T(f)=\begin{cases}\sqrt{T}, & 0\le|f|\le\dfrac{1-\alpha}{2T}\\ \sqrt{\dfrac{T}{2}[1+\cos\dfrac{\pi T}{\alpha}(|f|-\dfrac{1-\alpha}{2T})}, & \dfrac{1-\alpha}{2T}<|f|\le\dfrac{1+\alpha}{2T}\\ 0, & |f|>\dfrac{1+\alpha}{2T}\end{cases}, \qquad (1)$$

where α is the roll-off factor within the range of $0\le\alpha\le1$, and T is the symbol interval. A matched RRC filter is also adopted in the receiver in order to form a raised cosine roll-off Nyquist filter for pulse shaping [14]. Subsequently, the transmitted signal is amplified and output by the custom-designed transducer array.

Received by another custom-designed transducer array acting as the receiver, sound waves after passing the air channels are converted back into electrical signals. It is inevitable for the proposed system to encounter acoustic reflections when implemented in real environment such as those in rooms, hallways, etc. Compared with conventional time domain approaches, SC-FDE offers better ISI mitigation performance and complexity tradeoff. With efficient FFT data block and simple frequency-domain equalizer, lower complexity can be achieved [17]. The output signal of RRC filter is also divided into several frame blocks. Pilot sequence in each block is separated out after CP is removed, and equalization weights can be acquired after FFT. These optimized filter coefficients are multiplied to the subsequent information sequence to compensate for channel distortion. The equalized data in each frame achieved after inverse fast Fourier transform (IFFT) are combined and reorganized as frequency-domain equalization received signal. Finally, it is sent to the digital demodulator where Viterbi's maximum likelihood algorithm [15] is adopted for decoding.

4. EXPERIMENTS AND RESULTS

4.1. Performance Indices

Modulated audio signals were generated following the procedures mentioned above. Binary source code word had a length above 10^5 bits so as to acquire reliable bit error rate (BER) in Monte Carlo test. Part of the 64-QAM waveforms

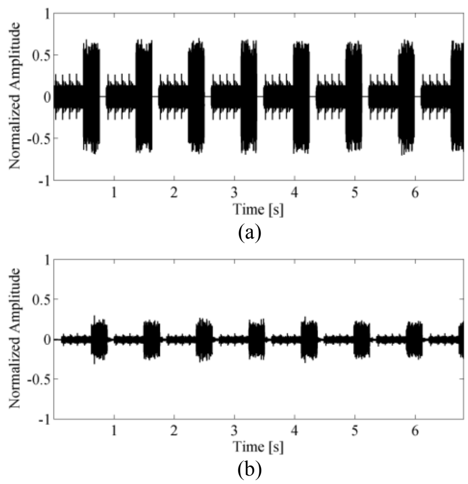

Fig. 3. Part of time-domain signal waveforms in the acoustic data transmission system. (a) The transmitter. (b) The receiver.

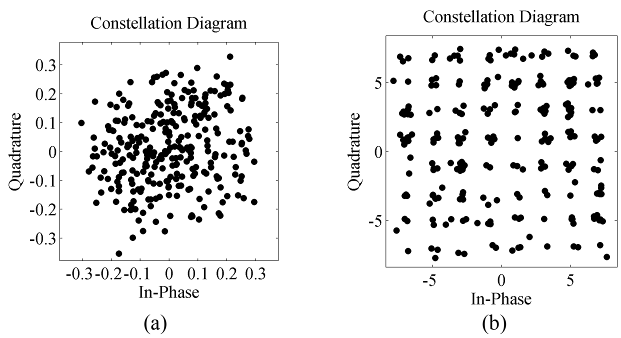

Fig. 4. Constellation diagrams for signals of the acoustic data transmission system. (a) In the receiver. (b) After equalization.

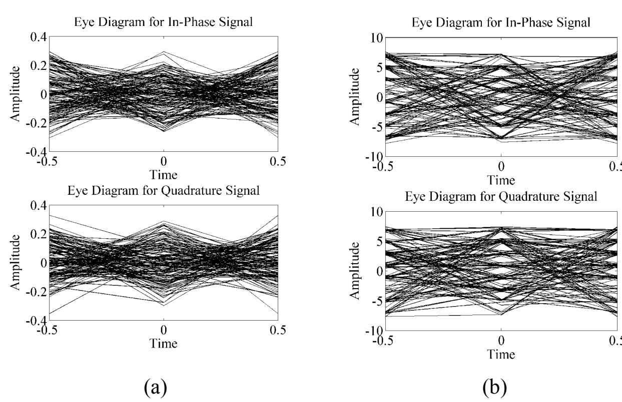

Fig. 5. Eye Diagrams for both in-phase and quadrature signals. (a) In the receiver. (b) After equalization.

of the transmitted signal and the received signal are illustrated in Fig. 3. Due to the presence of channel response signals, i.e. echo and noise, the received signal can be attenuated and distorted in time domain and the signal constellation diagram appears to be more scattered as shown in Fig. 4(a). The eye diagram also becomes closed and indistinct as shown in Fig. 5(a). Under this condition, the vertical extent of the eye for both in-phase and quadrature channels is reduced. After channel equalization, constellation points reassembled and the eye pattern open again, as represented in Fig. 4(b) and Fig. 5(b), respectively. The best time for sampling is at the point where the vertical opening of the eye reaches the largest. After QAM digital demodulator and channel decoder, the estimation of source code word can be reacquired.

We evaluated the effects of various parameters associated with the transmitted signals in a rectangular room with dimensions of 5.00 m × 3.50 m × 2.50 m on the performance of the system. The distance between the transmitter and the receiver was five meters and the normalized amplitude remained between 0.3 and 0.5. Each case was tested three times and their mean was taken to reduce discrepancies. For *M*-ary QAM modulation, system performance will be affected since the distance between the adjacent signals in signal constellations is decreased as *M* is increased. With channel coding added, Euclidean distance between transmitted signals is increased, thus reducing the BER without increasing the value of the transmitted power. This pursuit of reliability is at the expense of increasing bandwidth and the complexity of the system. The data rate is also reduced as the redundancy is added. Due to the addition of pilot sequences, the ratio between the length of the information sequence and that of the whole sequence should be multiplied to baud rate and used as the final symbol rate. We observed that if the pilot sequence length was set too small, useful information of the channel would be lost and the system BER would be increased. On the other and, if the transfer function length was stretched too far, it would introduce additional channel noise, the data would therefore be distorted and the system performance be degraded. With an increased SNR, the system performed well when the normalized amplitude of the received signal was within an acceptable parameter scope ranged between 0.2 and 0.6. Taking into account the system frequency response and its directivity characteristics, the carrier frequency should be selected between 3 kHz and 6 kHz. The optimal value for the parameters associated with the transmitter signal will involve some trade-offs and need to be adjusted through practical tests.

4.2. Long-Range Tests

We also tested long-range performances both in hallway and outdoor environments. The main conundrum limiting the transmission range of acoustic communication is the impedance mismatch with air. Sound wave intensity attenuates greatly as distance to the source is increased. Using horn loudspeakers array allows an effective acoustical match between the diaphragm at the narrow end of the horn loudspeaker and the air to be achieved [18]. Using an equivalent area, we get a Rayleigh range of the transducer array which is directly proportional to the carrier frequency.

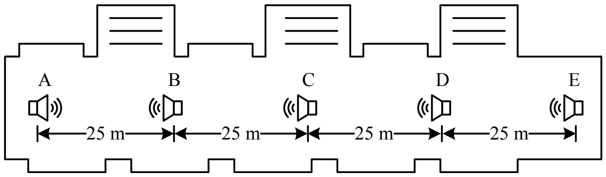

Fig. 6 Layout of long hallway test scenario

Table 1 BER of the acoustic data transmission system tested along a long hallway.

Symbol Rate (baud/s)	Distance = 25 m		Distance = 50 m	
	64-QAM	16-QAM	64-QAM	16-QAM
294	0.384534	0.000102	0.465683	0.020820
147	0.000000	0.000000	0.000217	0.000000
73.5	0.000000	0.000000	0.000000	0.000000

Symbol Rate (baud/s)	Distance = 75 m		Distance = 100 m	
	64-QAM	16-QAM	64-QAM	16-QAM
294	0.493272	0.430773	0.471788	0.016146
147	0.000596	0.000000	0.000116	0.000000
73.5	0.000000	0.000000	0.000000	0.000000

The higher the frequency is, the longer the distance between arrays becomes. As the distance increases, amplitude of the received signal decays.

Measurements were carried out along a long hallway exceeded 100 meters in Dah-You Maa Building of Institute of Acoustics, Chinese Academy of Sciences. Reflections from doors, windows, stairs, etc. will make the acoustic data transmission a much more complex circumstance. As shown in Fig. 6, A stands for the transmitted transducer array, while B to E represented the received transducer array tested for a range of distances every 25 meters. Monte Carlo test for each case was repeated three times and their average was calculated and used as the final result. Results are shown in Table 1. Since the data length is around 10^5 bits, if the BER exceeds 0.001, the system reliability is low for data transmission. Error performance of the system degrades as symbol rate is increased. As M is increased, i.e. from 16-QAM to 64-QAM modulation, BER also increases accordingly for a given value of symbol rate. We also found if we amplified the transmitted signal and ensured the received signal amplitude was large enough, BER was not increased as the distance between the transmitter and receiver was increased. Consequently, the system performed well for different distances up to a hundred meters and can reach a maximum bit rate of 0.882 kbps.

4.3. Outdoor Tests

Outdoor tests were also carried out to evaluate the system performance. Considering the results in Table 1, a 16-QAM modulation was chosen for the outdoor tests and the symbol rate equaled 147 baud/s. There are no obvious differences in the results for the different distances tested, ranged from 25 m to 100 m. The outdoor tests repeated in the same location from different time of the day, and in three days. The results differ significantly amongst various tests subjected to different wind conditions. The stronger the wind was, the higher BER would be. Tests were also carried out to test the performances for different baud rates and different signal amplitudes. Data of these different parameters obtained from three different days exhibited similar characteristics. Determining the relationship between wind conditions and system performance is our future work.

5. CONCLUSIONS

A new approach to acoustic data transmission in air is demonstrated in this paper. System prototype is implemented with custom-designed transducer arrays using the proposed quadrature modulation and single carrier frequency domain equalization algorithm. Each array, which can be set to either as a transmitter or as a receiver, plays an effective role as a node to form a network. With extra optimizations like channel coding and pulse shaping, system performance can be improved. Transmitted signal, received signal and equalized signal are analyzed and compared in the form of time-domain signal waveform, constellation diagram and eye diagram. We also analyzed the effects of signal parameters including the type of M-ary QAM modulation, pilot sequence length and audible wave amplitude on the system performance. The optimal frame structure is designed with parameters whose values involve many trade-offs between various factors. Experiments conducted along a long hallway show that the proposed acoustic data transmission system can resist reverberation effectively and achieve reliable communication under a long-range circumstance. The system performance in outdoor test is greatly affected by the characteristics of the wind, which requires further study.

6. ACKNOWLEDGEMENT

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