Design and Analysis of Acoustic Data Transmission System for Wireless Multimedia Data Transmission

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***Abstract*—Acoustic communication is transmission technology to transmit sound and data simultaneously through speaker and microphone. In this paper, we propose a parallel acoustic communication system enabling audio and multimedia data transmission at the same time. The parallel acoustic communication system is based on orthogonal frequency division multiplexing (OFDM) using acoustic wave. The sampling clock offset (SCO) occur because of clock mismatch between ADC and DAC as sound card. Therefore, we compensate sampling clock offset and do real experiment to ensure the good system performance. We measure system performance depending on the distance. The propose system is available as short distance multimedia data transmission method.**

Keywords-Acoustic communication; OFDM; SCO; ICI; STO;

I. INTRODUCTION

Recently, interest about acoustic communication has been growing because of development of mobile and PDA technology. Acoustic communication uses audio frequency band. But, human ears cannot perceive hidden information of propagated audio signal in the air. Acoustic communication is not required in additional equipment because it only uses microphone and speaker as input and output.

There are several techniques about acoustic communication such as echo hiding method that transmits data in air in very short time, spread spectrum method that hide data in sound and acoustic OFDM-based data transmission technique. Echo transmission method is easy to implement because echo data is inserted in time domain without regarding frequency domain [1], [2]. Spread spectrum is transmission method that overlaps signal using Pseudo-random-Noise (PN) lower than frequency masking threshold value [3].

This method is robust to aerial propagation and environmental noise. But its data throughput is lowed because the system has data spread. Acoustic OFDM signal removes high frequency band by low pass filter (LPF) to generate a low-frequency audio signal. Then each subcarrier of OFDM is controlled such as the spectral envelope of the original sound source. Afterward, generated OFDM signal is inserted in high frequency band of audio signal and is transmitted. Acoustic OFDM system has much better not only sound quality but also data throughput than different techniques. However, acoustic OFDM occurs degradation of sound quality and performance of communication since low pass filter is used [4]-[6].

In this paper, we like to design a parallel acoustic communication system enabling audio and multimedia data transmission at the same time. Parallel acoustic communication system can be sent with minimal quality loss of audio signal and voice signal through re-sampling process. Also, this system can transmit information data over audio signal frequency band. Therefore, information signal and audio signal can efficiently transmit simultaneously by using frequency band at least. Furthermore, we analyze effect of system performance degradation by simulation and experiment because of low cost acoustic communication devices. We compensate the system performance degradation by using least square zero forcing (LS-ZF) in training signal-based channel estimation techniques [7]-[9]. Also we compensate channel effect by frequency offset [10].

II. PARALLEL ACOUSTIC COMMUNICATION SYSTEM

A. Parallel acoustic communication system

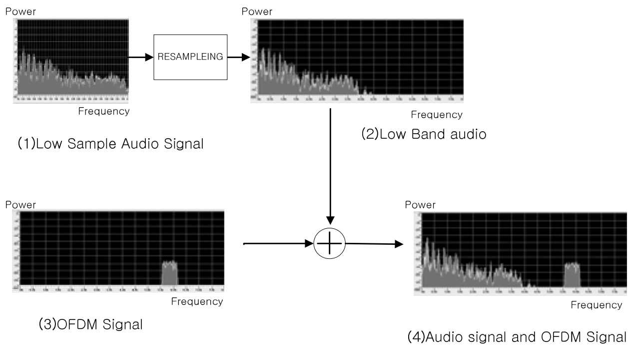
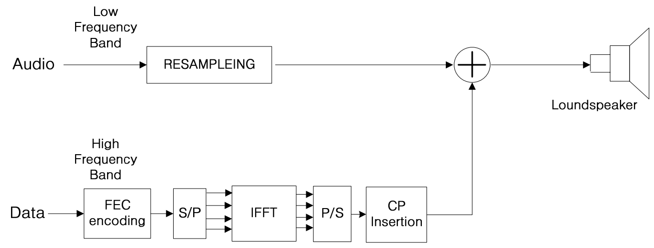


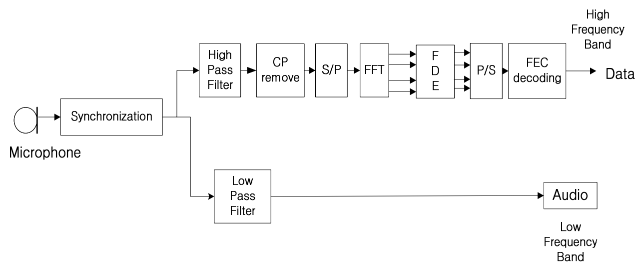
Fig. 1. Parallel acoustic communication system concept.

Figure 1 is system concept. First, low sample sound source makes a two times re-sampling. OFDM signal is transmitted to high frequency band of re-sampled audio signals. At this time, audio signal and OFDM data are transmitted simultaneously.

Figure 2 shows a parallel acoustic communications based on OFDM system. OFDM data is located in high frequency bandwidth of different audio signal. And OFDM data is not affected BER performance of audio signal because frequency bands of OFDM data and audio signal is different each others. Modulation method based on OFMD has the merit of frequency efficiency compared with single carrier method.



(a) Block diagram of transceiver.



(b) Block diagram of receiver.

Fig. 2. Parallel acoustic communication system.

OFDM signal is composed of subcarriers that are modulated as BPSK or QPSK. OFDM signal is represented as

$$x(n) = \sum_{k=0}^{N-1} X(k) e^{j\frac{2\pi}{N}kn}. \tag{1}$$

where $x(n)$ and $X(k)$ are OFDM modulated symbol as time domain and the data signal as frequency domain, k is subcarrier number.

The guard interval and cyclic prefix are inserted to a symbol in order to reduce inter-symbol interference (ISI) occurred by multi-path channel. Received OFDM signal is represented as

$$r(n) = \sum_{k=0}^{N-1} X(k)H(k)e^{j\frac{2\pi}{N}kn}. \tag{2}$$

$$\begin{aligned} R(k) &= \frac{1}{N}\sum_{k=0}^{N-1} r(n)e^{j\frac{2\pi}{N}kn} \\ &= \frac{1}{N}\sum_{k=0}^{N-1}\sum_{l=0}^{N-1} X(l)H(l)e^{j\frac{2\pi}{N}ln}e^{-i\frac{2\pi}{N}kn} \\ &= \frac{1}{N}\sum_{k=0}^{N-1}\sum_{l=0}^{N-1} X(l)H(l)e^{i\frac{2\pi}{N}(l-k)n} \\ &= \frac{1}{N}\sum_{k=0}^{N-1}\sum_{l=0}^{N-1} X(l)H(l). \end{aligned} \tag{3}$$

The acoustic communication system transmits a sound in low frequency under 4 KHz and OFDM data in high frequency over 4 KHz. First, the voice signal has about 3.6KHz bandwidth and we can do 8 KHz sampling without frequency loss. Voice signals sampled at 8 KHz according to Nyquist-Shannon sampling theorem has the maximum frequency 4KHz. Voice and low sampling audio signal do not affect to OFDM signal as interference. Because OFDM signal is transmitted over 4 KHz. OFDM signal should be transmitted after power control of subcarriers because sound of OFDM data in high frequency band hears like AWGN noise. However, We do not hear power controlled OFDM signal than generally OFDM signal.

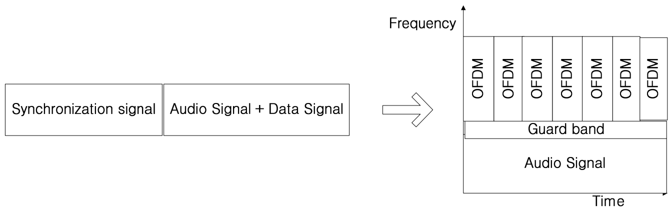


Fig. 3. Synchronization signal and data frame structure.

Figure 3 shows synchronization signal and frame structure. Synchronizations signal of one block is transmitted for searching correct starting point and afterward the 40 blocks as audio signal and data signal are transmitted.

B. *Synchronization signal*

We do not know starting point of sound signal when recode sound through microphone, Therefore we need synchronization signal $x[k]$ to know start point of receive signal. The synchronization is basically using correlation value with a known synchronization signal.

$$x[k]*x[-k] = \sum_{n=0}^{N} x[n]\cdot x[n+k] \approx NE\{x[n]x[n+k]\} = N\delta[k] \tag{4}$$

When Synchronization signal correlated, the signal has the maximum values $N\delta[k]$ as shown figure 4. Thus, we can gain stat point of signal through k-th signal with maximum values $N\delta[k]$.

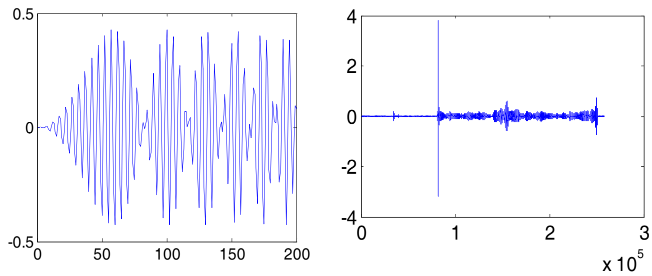


Fig. 4. Synchronization signal and correlation.

C. *sampling clock offset effect*

Frequency offset is generated between transmitter and receiver in the real acoustic communication system. The computer sound devices exists clock's difference in two sound cards between A/D converter and D/A converter. And these problems are more serious because those devices did not be made for acoustic communication.

Those problems that are made by different symbol periods between transmitter and receiver make ICI (inter-carrier interference). ICI effect is increased when two sound card more generate the sampling clock mismatch. The follow formulas express how the problems, sampling clock frequency offset, are generate in receive part.

$$\Delta = \frac{(Ts - Ts')}{Ts}, \tag{5}$$

$$\tilde{Y}[k] = Y[k]\frac{\sin(\pi\Delta k)}{\sin(\pi\Delta k / N)}e^{\frac{j\pi\Delta k(N-1)}{N}} + Z_{ICI}[k]. \tag{6}$$

Equation (6) shows that ICI effect generates to change magnitude and phase of the received signal. Therefore we compensate changed signal amplitude and phase using the normalization four comb-type pilots.

The phase rotation of received signal occurs as random values by sampling offset in frequency domain. We can compensate the phase rotation using mean values of pilots.

III. Simulation And Experiment Results

A. BER performance in simulation

Table 1 shows the system parameters. The OFDM simulation is to implement by BPSK and QPSK modulation and used frequency band 4234~8000Hz.

TABLE I. Simulation Enviroment

Parameter	Value
Sampling frequency	16KHz
Audio signal frequency	0-4000Hz
OFDM carrier frequency	4234-8000Hz
Carrier modulation	BPSK/QPSK
Number of carriers	33(+4pilots symbols)
Symbol interval	2048sample(128ms)
Guard interval	512samples(32ms)
FEC coding	1/3 convolution coding

Real acoustic communication systems decrease BER performance by the sampling timing offset and frequency offset. Therefore, we simulate to consider the effect of these two degradation factors.

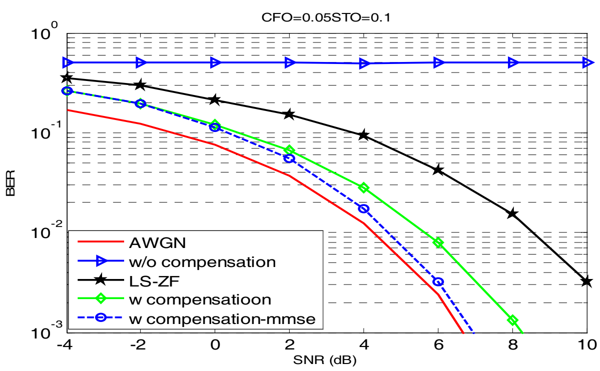


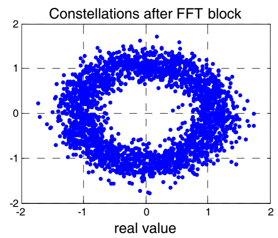
Fig. 5. BER performance

Figure 5 shows BER performance according to channel compensation methods. We compensate CFO and STO effects by using the average value of 4 pilots and get better performance than LS-ZF method.

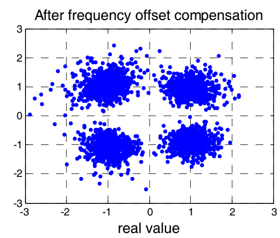
Phase rotation is compensated by using normalized pilot. But, Spreading of received signal by ICI effect is not compensated. Accordingly, we use Minimum Mean Square Error (MMSE) method to reduce ICI effect.

B. Constellation in experiment

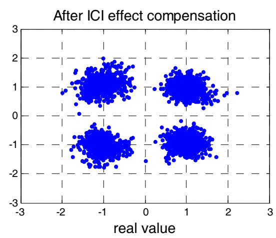
We experiment system performance about 50 dB noise level environments. This is similar to noise level in office environment.



(a) Constellation of received signal



(b) Compensation offset



(c) Compensation ICI effect

Fig. 6. Experiment constellations

Figure 6 shows different constellation when distance difference of transmitter and receiver is 1 meter. Figure 6 (a) shows phase rotation and magnitude decrease of received signal in frequency domain because of frequency offset effect. Figure 6 (b) shows constellation after offset compensation by using mean value of comb type pilots. Figure 6 (c) shows constellation after ICI effect compensation.

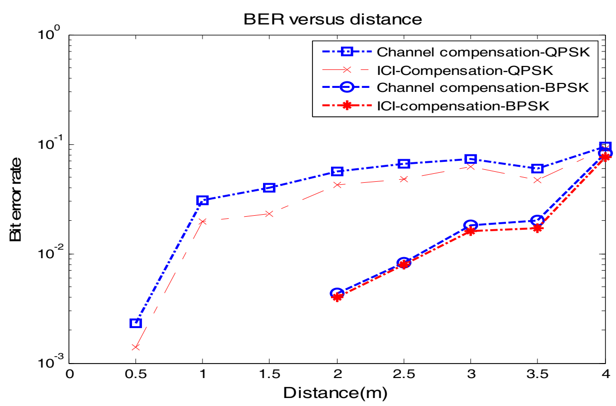


Fig. 7. BER performance

C. BER performance in experiment

Figure 7 shows BER performance according to distance measured. The acoustic communication system using both channel compensation and MMSE method has better performance than using only channel compensation method. The system performance is improved by removing the ICI through the process of MMSE.

D. System performance in experiment

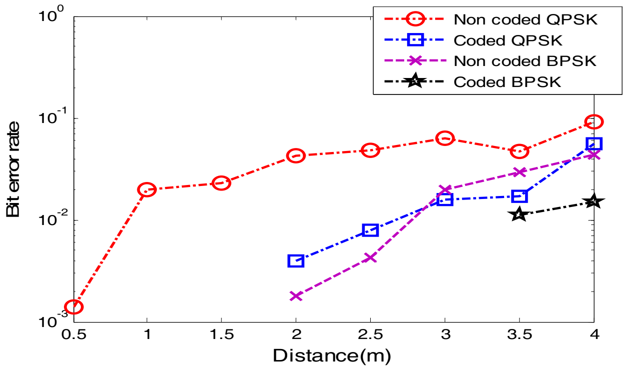
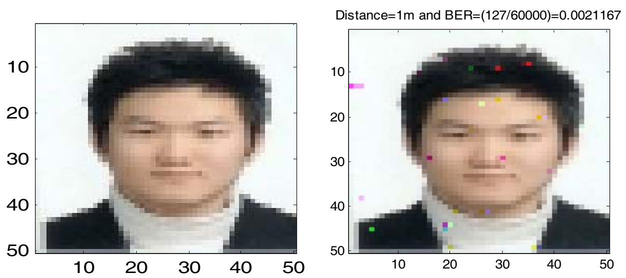


Fig. 8. BER versus distance

Figure 8 shows the result of bit error rate (BER) according to propagation distance and when we transmit only short text message. We consider the two cases using channel coding or not. The signal power loss will affect the BER performance.

E. Image transmission performance



(a) Sending image (b) Received image

Fig. 9. The results of channel and ICI compensation

Figure 9 shows the results of image transmission in experiment. We measure image transmission performance when distance difference of transmitter and receiver is 1 meter and microphone is positioned about 5 degrees from speaker. The results of our experiment show that a parallel acoustic communication system is available as short distance communication for transmitting image and text.

IV. Conclusion

In this paper, we implement simulation and experiment about performance of acoustic communication system based on OFDM in various environments. So we do experiment about 50dB noise level environment by using two computers as transceiver. When we receive the signal in the receiver sampling offset and phase rotation is included by different clock speed of different sound card devices. Therefore, we compensate phase rotation through channel compensation techniques in frequency domain. And also, we measure the BER performance according to the distance. We confirm that simulation results and experiment results are similar. We transmit image through acoustic communication system and ensure that received system performance is improved through experiment.

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