Proposal and Evaluation of a Data Transmission Method for Using Sound in Accurate Indoor Positioning

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Abstract— **This paper proposes a method to process the data representing the sound that is used to determine the indoor position of objects. A scheme is proposed and its theoretical validity is confirmed by numerical simulation using MATLAB. The scheme is to modulate a sound source based on information or data obtained before diffusion (spread spectrum) by an M sequence code. The experimental setup includes a speaker and a microphone, and the bit error rate is evaluated by changing experimental conditions such as the transmitting frequency and the order of the M sequence code. This fact leads to the realization of the system by sound, to provide both indoor positioning and sensor data transmission functions.**

Keywords-Sound; Spread Spectrum; Data Transmission; Inaudible Sound; Indoor Positionng.

I. INTRODUCTION

Location information is one of the most important information, whether outdoors or indoors. Global Positioning System (GPS)/Global Navigation Satellite System (GNSS) has become a standard positioning system and its hardware and uses now involve large market. Although many kinds of positioning methods and systems have been studied and developed [1][2], there is no standard system to determine the position of objects or people in an indoor environment. At present, several systems may be selected, depending on the specific requirements of particular situation.

The authors have been studying a highly accurate indoor positioning system that uses sound. The velocity of sound is quite low compared with that of radio waves, so the problem of errors resulting from the timing of signal reception presents no severe difficulties, and it is easy to discriminate between direct sound and multipath sound due to reflections etc. This enables accurate positioning by the use of sound. If an additional data transmission function can be included in this system, so that transmission of both positioning and other information can be achieved with the same system components, the applicability and value of the system will be greatly enhanced.

At the first stage, we use ultrasonic waves and detect the time of reception by the voltage at the receiver. This can realize high accuracy, but requires an ultrasonic transmitter. This equipment is not in common use, which this could act as a barrier to the system coming into widespread use [3]. As a second stage, we used inaudible sound from a smartphone. In this system, a spread-spectrum sound source and a correlation calculation are used to detect the reception time at each microphone sensor. The position is then obtained from the time difference between sound reception at each receiver [4].

In the GPS/GNSS system, information such as satellite orbit data etc. are included in the radio wave used for positioning. It is thought that data can be similarly embedded in the sound used in this positioning system, as both radio and sound are wave phenomena. We are now investigating a realization of both of accurate indoor positioning and information data transmission, such as sensor data in one system.

This paper presents a method to embed additional information in the sound used for indoor positioning. A scheme using sound diffusion (Spread Spectrum (SS)) is proposed, and its validity is confirmed by numerical simulation in Section III. The experimental setup uses a speaker and a microphone and the transmission quality has been evaluated under a range of experimental conditions. It was shown in Section IV that data transmission can be achieved by the proposed method, which points to a system that has the dual functions of indoor positioning and data transmission.

II. TARGET SYSTEM AND ADDED FUNCTION

A. Summary of composed indoor positioning system

The principle of the indoor positioning system to which the authors will add a data transmission function is shown in Figure 1 [5]. The positioning principle of this system is based on the Time Difference Of Arrival (TDOA) principle.

The spread spectrum sound diffused by an M sequence code is transmitted from the positioning target to microphone sensors connected to each receiver. The timing in the received signal is detected by correlation calculation, and the time differences at each sensor (t_1, t_2, t_3) are obtained. The position of the sound source (positioning target) is calculated from the time difference data. The positioning principle is based on the difference between arrival times. The positioning principle is based on the differences between times of arrival. This method can yield quite high accuracy, with an error of less than a few centimeters. An example of the experimental results is shown in Figure 2. If the sound can include information, such as sensor data, the system can be used not

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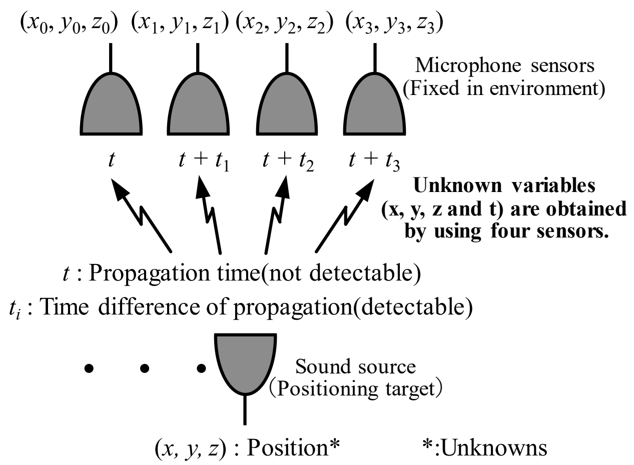

Figure 1. Positioning principle.

only for positioning but also for the transmission of other information, which would appear to add value to the system.

III. INFORMATION TRANSMISSION BY SOUND USED FOR POSITIONING

A. *Process of embedding information*

Figure 3 shows the process used to embed data in the sound used for positioning. The elements surrounded by dotted block were added to the system described in [5]. The time of reception can be obtained by correlating the received sound, which is spread by the same M sequence code, with its replica stored in the receiver. The authors propose modulating the sound source with data such as the output of a temperature sensor, etc., as the first modulation before diffusion processing.

The receivers get this spread spectrum signal via a microphone sensor. The correlation calculation is carried out to determine the time of reception, and the obtained values are used for positioning as described in Section II-A. Here, inverse diffusion processing is applied using the same M sequence code from this timing to retrieve the modulated signal. After this demodulation processing, the summation of the demodulated signal is carried out in order to obtain the desired data.

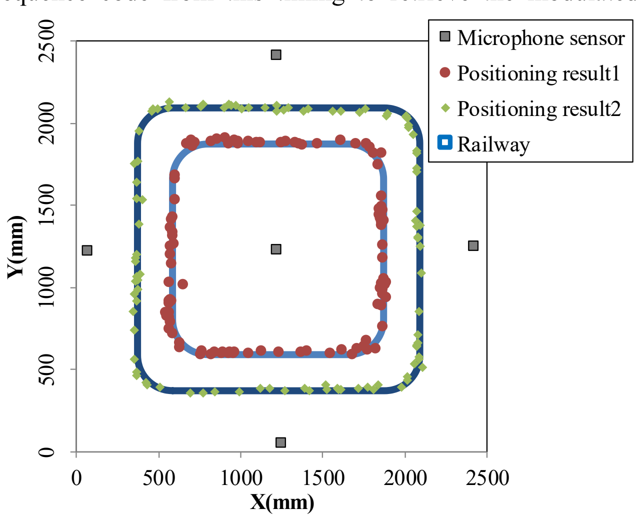

Figure 2. Example of experiment result.

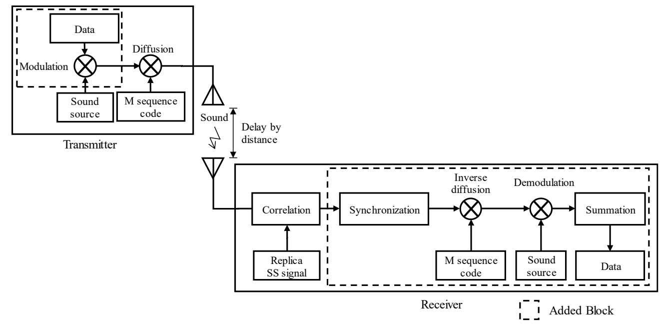

Figure 3. Block diagram for data embedding and transmission.

B. *Confirmation by numerical simulation*

Before constructing the experimental system, a simulation was carried out to verify the validity of the proposed procedure by using MATLAB programming. A 3rd order M sequence code was used in this simulation for ease in confirming the signal variation.

The simulation result is shown in Figure 4. The graph in the top line represents the data for transmission and that in the bottom is the data retrieved by the proposed procedure. The sound source is modulated based on the data (-1, 1, . . .), here -1, 1 which means 0, 1. This signal is modulated again by the M sequence code (1, 1, 1, -1, 1, -1, -1, . . . : -1 means 0 in the figure) as a second modulation. The received signal is the same as the transmitted signal, although its time of reception indicates the time required for sound propagation. The time of reception was determined by a correlation calculation using the replica, and the inverse diffusion was carried out from this timing. The signal to be demodulated was obtained after these procedures. The resulting data was obtained after summation of the demodulated signals.

It was confirmed that the modulated signal was regenerated on the receiving side after inverse diffusion. The required data is obtained after demodulation, and summation

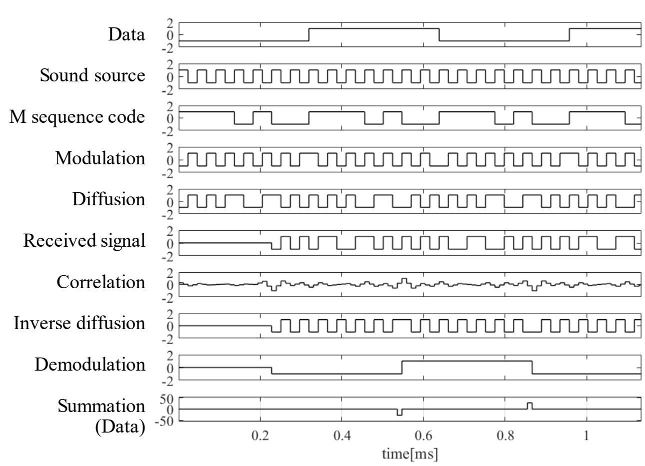

Figure 4. Simulation result for data transmission.

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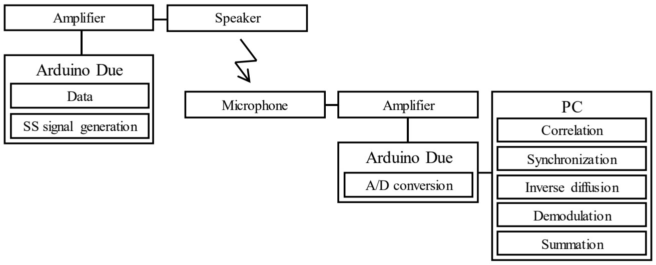

Figure 5. Experimental configuration for data transmission.

with this demodulated signal can enhance the reliability of data transmission, as shown in the graphs in Figure 4.

IV. EXPERIMENT AND EVALUATION

A. Experiment configuration

The authors have constructed an experimental system to confirm the validity of the proposed method. The experimental configuration is shown in Figure 5. A speaker (DB Products Limited: UM1515IA085008LFMP) connected to an amplifier was used as the sound source. The random 0 (-1), 1 code was used to provide the data, a rectangular wave was selected as the sound source, and this signal was modulated using the data and the M sequence code on an Arduino board (Arduino Due). Here, the data was created by a random data generator in advance and stored in the memory of the Arduino Due. The speaker was connected to this board via the amplifier to transmit the sound signal.

The signal received by a microphone sensor (Primo: EM-158) was amplified in the receiver, and A/D conversion was carried out in the Arduino Due before transmission to the PC. Then the correlation calculation, inverse diffusion, demodulation and summation were conducted to retrieve the original data that was embedded in the transmitted sound. The experimental setup is shown in Figure 6. The experiment was carried out in the laboratory room shown in this photograph and the campus cafeteria at lunch time, which provided a commonly encountered noisy environment.

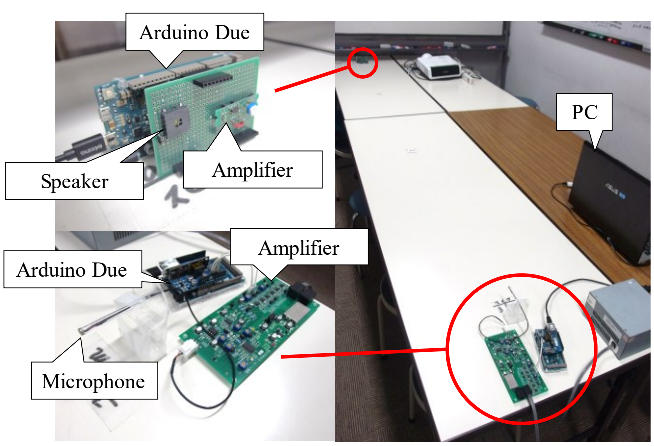

Figure 6. Experimental setup for data transmission.

TABLE I. EXPERIMENTAL PARAMETRES FOR DATA TRANSMISSION

Item	Unit	Parameters	
Sound source		Rectangular wave	
Source frequency : a	kHz	22.05	
Sampling frequency of transmission : b	kHz	44.1	b = 2a
Sampling frequency of reception : c	kHz	88.2	c = 2b
Data	bit	1000	
Order of M sequence : d		8	
Generator polynomial		$x^8+x^4+x^3+x^2+1$	
M sequence period : e	chip	255	$e = 2^d - 1$
1 sample length : f	μs	22.68	f = 1 / b
1 chip length : g	μs	45.36	g = 2f
1 bit time length of data : h	ms	11.6	h = ge
Transmission rate : i	bps	86.2	i = 1 / h

B. Experiment conditions

The basic experimental parameters are shown in Table I. The chips of M sequence code in one period for diffusion (255 chips) were embedded in 1 bit time length of the original data. The transmission quality was affected by many factors, such as the order of the M sequence code, the distance between the microphone sensor and the speaker, the sound frequency before diffusion, etc. Such factors were verified in this experiment, and also the bit error rate per 1000 information bits and the Signal to Noise Ratio (SNR) were evaluated. The SNR was estimated from the peak value of the correlation calculation and the noise level.

C. Experimental results and evaluation

The experiment was carried out five times under the same conditions to confirm repeatability. Then the average values were evaluated. An example of the correlation calculation result for an 8th order M sequence code is shown in Figure 7. It was verified that the peak appeared clearly, therefore positioning could be performed, and the inverse diffusion could be obtained from this result.

Figure 8 shows the summed results of demodulated signals for the results of 6th order and 8th order M sequence codes, respectively. Since the period of 8th order code is larger than that of 6th order code, the summation values in the 8th order result are larger. The positive values are considered to be 1, and the negative values are -1. It was confirmed that the summation values obtained with 8th order were more stable than those from 6th order. This means that more reliable data transmission can be achieved with an 8th order M sequence code, although its transmission speed (bits per second) is less than that of 6th order. Table II summarizes the experimental results. Nominal parameters for sound and diffusion are shown in Table I. One parameter and the experimental environment were changed to evaluate the effect on transmission performance. The distance between

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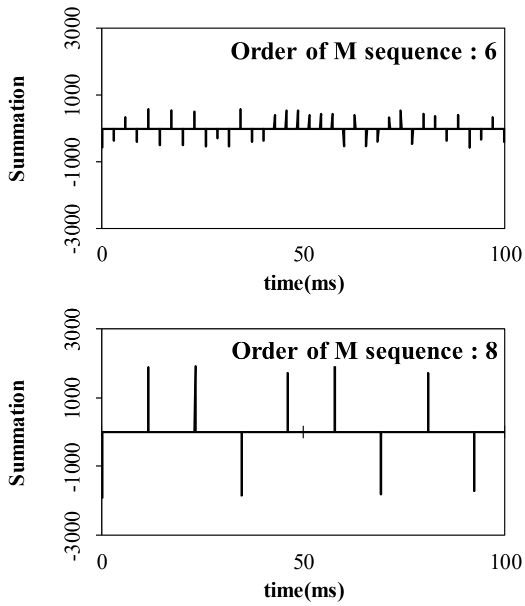

Figure 8. Results of summation after demodulation.

the speaker and the microphone sensor was set at 3 m, a length based on the floor to ceiling distance. The experiment was usually carried out in the laboratory room on campus (environmental noise: 40.4 – 56.2 dB) and the campus cafeteria at lunch time (66.7 – 93.4 dB). The following facts were confirmed in this experiment.

TABLE II. EXPERIMENTAL RESULTS FOR DATA TRANSMISSION

Parameters	Error bit		SNR	
	Usual [a]	*Noisy* [b]	*Usual* [a]	*Noisy* [b]
Order of M Sequence				
4	2 / 1000	Unmeasurable	2.2	2.2
6	0 / 1000	114 / 1000	3.7	2.8
8 (Nominal)	0 / 1000	0 / 1000	7.2	4.8
Sound frequency before diffusion [kHz]				
11.025	0 / 1000	0 / 1000	8.9	5.4
22.05 (Nominal)	0 / 1000	0 / 1000	7.2	4.8
33.075	0 / 1000	1 / 1000	7.1	4.9
Sound volume [dB]				
40.0	36 / 1000	N/A [d]	4.7	N/A [d]
55.0	0 / 1000	N/A [d]	6.1	N/A [d]
72.8 (Nominal)	0 / 1000	N/A [d]	7.2	N/A [d]
Order of low cut filter for transmitted sound data [c]				
0 (Nominal)	0 / 1000	3 / 1000	5.2	3.5
2	0 / 1000	11 / 1000	4.7	3.7

a. Environmental noise 40.4~56.2[dB]
b. Environmental noise 66.7~93.4[dB]
c. In this case, order of the M sequence was 7.
d. N/A means sound volume could not be measured due to a noisy environment.

(1) In the case of diffusion by a greater than 6th order M sequence code, quite a high quality of transmission performance can be obtained in a normal noise environment.

(2) Even in quite noisy conditions from 66.7 dB to 93.4 dB, the quality can be maintained if an 8th order M sequence code is used for diffusion. This result indicates that transmission using spread spectrum is robust in a noisy environment. The authors found it feasible to realize both of positioning and data transmission functions by one sound system.

V. CONCLUSION

The authors proposed a method to embed data in the sound used in a positioning system. The spread spectrum technique is applied to the sound signal, which is modulated based on the data. The feasibility and validity of the proposed method was confirmed by a numerical MATLAB simulation before an experiment was performed.

The experimental system was designed to verify the proposed method and to evaluate transmission quality. The order of the M sequence code for diffusion, the sound frequency and the environmental noise level were varied in the experiments. It was verified that the data can be transmitted along with the diffused sound data in which it is embedded, and high quality can be retained even in a noisy environment.

The final goal is to realize a system that can transmit both positioning information and sensor data. Integrating both functions remains as the next stage of the project.

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