Journal of Industrial Information Technology and Application *ISSN(Online): 2586-0852*
Vol.8 No.1

A method for improving latency between devices equipped with speakers

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Abstract. Delays in various wireless communication technologies in using IOT devices cause inconvenience to many people using the device. This imposes restrictions on properly enjoying various multimedia provided by modern society, and technologies to reduce delays are also steadily increasing to correct them. However, most studies are for connectivity at lower levels. Reducing time delay in the new standard will help develop future technologies, but ways to reduce delays in various existing wireless communications are expected to provide users with a better experience. In this paper, we introduce a method of correcting the time delay in a software way. In addition, we describe how to correct existing time delays by providing data on whether they can be corrected equally for various devices and various wireless transmission protocols. In this paper, a protocol to reduce delay time with Wi-Fi communication using Raspberry Pi was completed and an experiment was conducted. This is expected to provide a better communication environment to users in the future.

Keywords; IoT, Latency Improvement, Raspberry Pi

1. Introduction

What is IOT? It refers to any object that can be wirelessly connected to a network with the Internet of Things. These IOT devices provide a lot of convenience in human life, but there is bound to be delay in wireless communication technology. This causes inconvenience to users. Therefore, in this paper, we will introduce a technology for correcting delay in IOT devices.

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Received:; Accepted:; Published:

2. Existing Research

This chapter introduces existing studies and attempts to show the limitations of those studies.
A study on method of avoiding interferences between heterogeneous wireless transmission protocols in an IoT environment[1] This paper analyzes the delay in wireless communication. However, the delay cannot be corrected only by analysis.

A Study on Frequency Offset Compensation using 2-Phase Characteristic of Beacon Signal modulated by Satellite[2] Although proposed a circuit diagram for frequency correction, the disadvantage is that this study cannot be used unless this circuit diagram is used because it is a hardware correction rather than a software correction.

A Reliable Protocol for Real-time Monitoring in Industrial Wireless Sensor Networks[3] In the sensor network, all nodes are given a chance to relay in a visual way to increase the transmission probability. Since this is about reliability, not about correcting delays, delays are bound to exist

The above papers are papers that check the delay rather than the method of correcting the delay. Of course, confirmation is important, but I think the technology to correct the delay is more important. Therefore, in the next chapter, we will introduce the delay correction technology used in this paper.

3. Traditional communication methods

This chapter describes audio buffering, a delay calibration method for traditional Bluetooth earphones.

Audio buffering is a technical concept used to ensure consistency in delay and sound that can occur when processing and playing audio data. This allows for temporary storage of audio data for a period of time, allowing for smooth audio playback without delay or loss.

Here's how audio buffering works. When audio data is delivered, the playback device stores it in the audio buffer. The playback device takes out the data stored in the audio buffer and actually plays it. In this process, the playback device continuously processes the data that is piled up in the buffer to maintain smooth audio playback Fig1 is a picture of how audio buffering works.

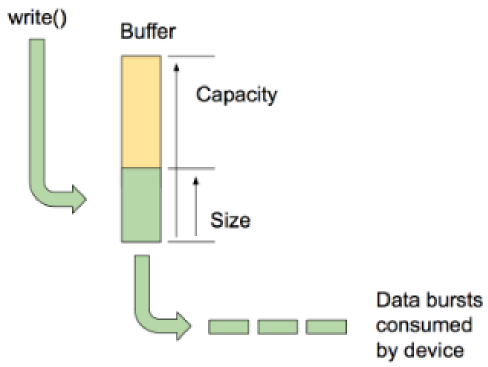

Figure 1. Example of a figure caption. (figure caption)

However, this method is stored in a buffer and exported, so if communication is not good, communication may be cut off even if communication is performed using the buffer, which is likely to provide a bad experience to users.

4. Delay Correction

In this chapter, the delay correction technology used in this study is described. It is a technology that corrects the delay by making it variable with Imin~Imax rather than keeping the delay fixed. This method starts with Imin rather than sending packets regularly, gradually increases to the Imax value if the communication status is good, and decreases the communication delay from the Imax value to the Imin value one step at a time if the communication status is bad.

In the past, when data packets are transmitted fixedly, data loss is inevitably high because they continue to be sent in the same cycle even if the communication is not good, but the method introduced in this study allows communication to be restored by transmitting faster when the communication is bad.

5. Experimental Method

In this experiment, P2P communication was performed using two Raspberry Pi 4B boards. The two devices belong to different networks, and the experiment was conducted by transferring files through UDP communication and then calculating the loss rate. However, since expensive equipment is required to communicate and test sound quality, in this experiment, the MP3 file was binarized, converted into a text file, and transmitted, and the device that received the text file was forced to convert the text file back into an MP3 file. For comparison, the exchanged binary files were compared.

When communication begins, text files are read and transmitted line by line, and at this time, each line is read by Delay to implement a variable delay time. Delay first started transmitting to Imin and communicated by gradually increasing the delay value to the maximum Imax if the communication status is good, that is, the loss rate is low. If the loss rate increased during communication, the experiment was conducted by reducing the delay value again. As a result of calculating the loss rate by repeating this process several times, it was confirmed that in addition to the method of fixing the delay time and then correcting it, it can be corrected through a variable delay time.

Each experiment was conducted by fixing 10 different sound source files at 5 ms, and the other was made to communicate between Imin and Imax values. fig 1. below shows the communication of sound source files. TABLE 1. represents the average of 10 experiments. A total of 40345555 bytes of packets were sent, and Experiment 1 was transmitted at 5 ms fixedly, and Experiment 2 was transmitted at Imax value from Imin. The results show that Experiment 2, which is the method proposed in this study, is 0.09% more efficient than Experiment 1.

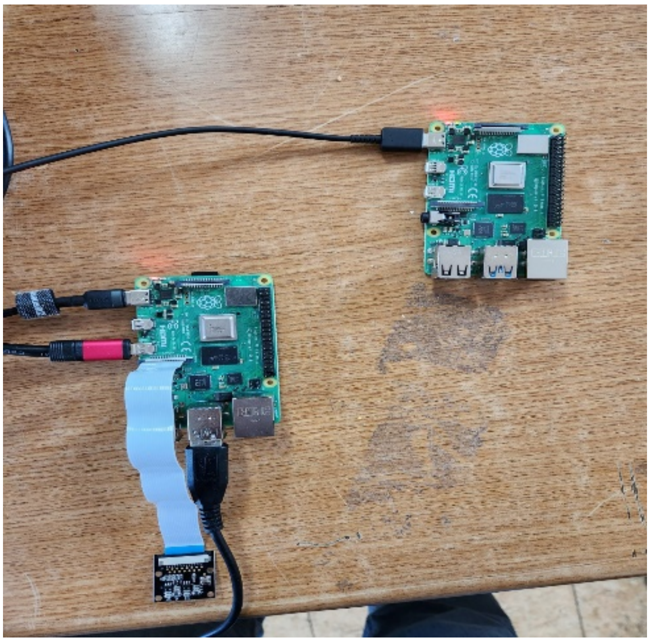

Figure 2. Example of a figure caption. (figure caption

Table I. EXPERIMENTAL RESULTS

	Experimental results			
	Experimental method	***Source bytes***	***Bytes of transferred files***	***Average percentage***
1	Fixed delay time	4,034,555	4,023,754	99.73%
2	Variable delay time		4,027,994	99.84%

6. Conclusion

In this paper, we explore innovative ways to provide users with a better experience in wireless communication environments. Conventional stationary communication methods can often face delays and performance problems of wireless communication. Therefore, this study aims to suggest efficient and optimized communication methods by controlling the delay of wireless communication.

The new methodology focuses on providing users with high-quality services by dynamically changing delays in wireless communication. By flexibly adjusting delays rather than relying on fixed delays, it is possible to effectively manage performance degradation caused by network congestion or other external factors.

In addition, this method has versatility applicable to various wireless communication technologies. This means that it can work effectively in environments where new technologies or devices are applied in response to the various current wireless communication standards and protocols. This allows users to expect consistent quality and service while utilizing various wireless communication devices and technologies

Taken together, this paper introduces innovative research that enhances the user experience through dynamic regulation of delays in wireless environments, and seeks both efficiency and stability by presenting universal solutions for various wireless technologies.

Acknowledgment

This work was supported by the Ministry of Education of the Republic of Korea and the National Research Foundation of Korea (NRF-2022S1A5C2A07091326)

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