FOREWORD to Vol. 11

Special Cluster on Advanced Communication Technologies in Conjunction with Main Topics of ICETC 2021

The 2021 International Conference on Emerging Technologies for Communications (ICETC 2021), organized by the Communications Society of the Institute of Electronics, Information and Communication Engineers (IEICE), was held online from December 1 to 3 and was a great success.

This conference was the second flagship international conference on all the technical fields covered by the IEICE Communications Society. This conference’s objectives are synergistic effects beyond the specialized fields, expanding research areas, and developing young researchers. This international conference is a place for discussions; therefore, preliminary results at the initial research stage are eagerly welcomed. Also, the younger researchers and students are well appreciated. As a result, we had 258 participants from various countries in the ICETC 2021.

This special cluster consists of but is not limited to submissions specifically from the authors of ICETC 2021. This cluster provided two submission deadlines to make a chance for revised submission. We received 13 and 15 submitted letters by the respective deadlines. After careful review, 12 letters were accepted in total.

I hope this special cluster will inspire researchers and promote further research activities in the communication research fields. Finally, I would like to express my sincere gratitude to all the authors for their outstanding contributions and to the reviewers and editorial board members for their tremendous efforts in making this special cluster a success.

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Roadside LED array acquisition for road-to-vehicle visible light communication using spatial-temporal gradient values

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Abstract: Most of the previous research on the acquisition of the LED Array transmitter assumes that the onboard receiver approaches the LED transmitter from the car’s front. In this study, we considered a case where the vehicle’s receiver crosses the LED transmitter. This letter modifies the algorithm using spatial-temporal gradient values to detect LED arrays. As a result, we achieved error-free acquisition and successfully communicated up to 9728 [bits] at a vehicle speed of 25 [km/h].

Keywords: intelligent transport systems, light-emitting diode transmitter, visible light communication, high-speed camera, LED array acquisition

Classification: Wireless Communication Technologies

References

[1] T. Yamazato, I. Takai, H. Okada, T. Fujii, T. Yendo, S. Arai, M. Andoh, T. Harada, K. Yasutomi, K. Kagawa, and S. Kawahito, “Image-sensor-based visible light communication for automotive applications,” IEEE Commun. Mag., vol. 52, no. 7, pp. 88–97, July 2014. DOI: 10.1109/MCOM.2014.6852088

[2] H.S. Liu and G. Pang, “Positioning beacon system using digital camera and LEDs,” IEEE Trans. Veh. Technol., vol. 52, no. 2, pp. 406–419, 2003. DOI: 10.1109/TVT.2003.808800

[3] I. Takai, T. Harada, M. Andoh, K. Yasutomi, K. Kagawa, and S. Kawahito, “Optical vehicle-to-vehicle communication system using LED transmitter and camera receiver,” IEEE Photon. J., vol. 6, no. 5, pp. 1–14, Oct. 2014. DOI: 10.1109/JPHOT.2014.2352620
1 Introduction

In the Visible Light Communication (VLC) research that transmits data from an LED Array transmitter to the moving vehicle, many works consider the LED Array is in front of the car. In contrast, this research assumes the vehicle (receiver) crosses the LED Array transmitter [1, 2, 3, 4, 5]. Figure 1(a) shows our model. There are two issues in this case. The first is that the LED Array in the captured image of the receiver moves significantly in the horizontal direction. The second is that the shooting time is extremely short. In this study, we tried to solve these two problems.

In the previous study [4], Usui et al. developed the LED acquisition based on the spatial-temporal gradient values. They succeeded in visualizing the effectiveness of using the spatial-temporal gradient values by the spatial-temporal cross-sectional image of the LED Array. However, the previous study has not considered the large movement of the transmitter in the captured image.

In this study, we considered a case where the vehicle’s receiver crosses the LED transmitter and conducted a field trial experiment.

We modified the algorithm of [4] to acquire the LED Array by correcting the horizontal position error of the transmitter in the captured images.

In addition, we conducted the LED acquisition with forward and reverse shooting orders. By combining these two shooting orders, we could demodulate the data without error in all the captured images except the first frame and the last frame.

The experiment result shows that LED acquisition achieves a 100% success rate and the communication performance achieves error-free.

2 System model

Figure 1(b) shows a block diagram of the system used in this study. The transmitter consists of 256 LEDs arranged in a $16 \times 16$ square matrix and a modulator. The input data is modulated by on-off keying (OOK) and the LEDs blink in response to the OOK signal.

The receiver consists of a high-speed camera, an image processing section, and a demodulator. First, the high-speed camera takes an image of the LED Array and outputs the image. Then, for the image processing section, the LED Array is acquired in the captured image in the LED array acquisition and tracking part, and the position and the luminance value of every LED in the LED Array are measured in the LED position estimation part. Finally, the measured luminance is demodulated by the OOK demodulator, and the transmitted data is obtained. In our proposed
acquisition method, we perform LED position correction in the image processing section. Figure 1(c) shows the experiment view. The LED Array transmitter is placed on the side of the road. The vehicle is moving across the LED Array from the right side to the left side, and the receiver is mounted on the back seat of the vehicle.

We show the experimental specifications below. The LED blinking frequency is 1000 [Hz]. The capturing frame rate is 2000 [fps]. The image resolution is \(512 \times 512\) [pixels]. The distance from transmitter is 5[m]. The vehicle speed is 25[km/h]. The time is daytime. The weather is sunny and cloudy.

This study designed two lighting patterns for the LED Array, called the 64-bit Array and the 256-bit Array. One bit of data is represented by the block of \(2 \times 2\) LEDs and \(1 \times 1\) LED for 64-bit Array and 256-bit Array, respectively. The block size of 256-bit Array is smaller than 64-bit Array; therefore, 256-bit Array can transmit

![Diagram of VLC system](image)

(a) Assumed environment (vehicle crossing the transmitting LED array).

![Block diagram of VLC](image)

(b) Block diagram of VLC.

![Experiment view](image)

(c) Experiment view.

**Fig. 1.** System model of road-to-vehicle visible light communication.
more bits per frame but is more challenging to acquire than 64-bit Array.

3 Experiment results

This section evaluates the proposed acquisition method and the communication performance from the captured images obtained by the experiment.

3.1 Evaluation of the proposed acquisition method

Figure 2(a) shows the captured image of a 64-bit Array where the LED Array is located at the center of the image, surrounded by background noises.

We created a spatial-temporal image and a spatial-temporal cross-sectional image of the 64-bit Array using the method in [4], as shown in Fig. 2(b) and (c). From Fig. 2(c), it can be found that for each LED in the LED transmitter, the luminance value changes considerably in the \( t \)-axis (time) and the position moves significantly in the \( x \)-axis (space) so that the LED Array is difficult to acquire using this cross-sectional image generated by the previous method. We can determine the amount of movement of the LED position from the slope of the space-time axes. Figure 2(d) is the image corrected the LED position based on the calculated amount of movement. From Fig. 2(d), we can see that the LED Array’s movement is corrected and the LEDs become parallel to the \( t \)-axis. As a result, we can use the corrected image with previous LED acquisition algorithm in [4] that uses scatter plot of spatial-temporal gradient and a threshold line to detect the LED Array.

In addition, we calculated the average values of the spatial and temporal gradients of each pixel over multiple frames to mitigate the effects of background noise. Figure 3(a) and (b) show the scatter plots of averaged spatial and temporal gradient values for a 64-bit Array over eight frames. Figure 3(a) used the previous method in [4] while Fig. 3(b) used the proposed method that corrected LED positions. Red solid circles are the LEDs and blue crosses are the background non-LED. We

![Fig. 2. Various view of spatial-temporal images](image)
classified LED and non-LED by drawing a threshold line. It can be noted that the non-LEDs in Fig. 3(b) are concentrated in a specific region and the LEDs have a specific spatial-temporal value compared to Fig. 3(a). Therefore, the proposed method is more effective in distinguishing between the LED transmitter and the non-LED background noises.

### 3.2 Evaluation of communication performance

We demodulated the data from the acquired LED transmitters and measured the BER of each captured image. In the previous study [4], the parameters of the LED transmitter were averaged in the order of shooting, so the transmitter in the first average number of frames could not be acquired. In this study, in addition to the order of shooting, we averaged the parameters in the reverse order of shooting. Figure 3(c) and (d) show BER in each frame. As a result, we could demodulate the data without error in all the captured images except the first and last frames by combining forward and reverse shooting orders. The first and last frames cannot be acquired because the temporal gradient cannot be calculated. The numbers of frames transmitted in this experiment were 42 frames and 40 frames for 64-bit Array and 256-bit Array, respectively. We achieved the data transmission of 2560 bits for 64-bit Array and 9728 bits for 256-bit Array.
4 Conclusion

In this study, we proposed a new LED acquisition algorithm that improves upon the previous study’s algorithm. The proposed method achieved LED acquisition by correcting the positions of the LED array transmitter under the case of a vehicle crossing the transmitter whose position changes significantly on the horizontal direction.

We performed the LED acquisition in the forward order and the reverse order and we successfully acquired the transmitter in all the captured images except the first and last frames by combining these two methods. The combination increased the number of acquirable LED Array images compared to the previous study.

We achieved error-free acquisition and demodulated the data for the captured LED transmitters, where the data rate achieved 64 [bit/frame] for 64-bit Array and 256 [bit/frame] for 256-bit Array. We succeeded in communicating up to 9728 [bits] at a vehicle speed of 25 [km/h].

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A method for generating graphs to derive maximum flow and its evaluation

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Abstract: The study presents a “virtual grid system” currently being developed by the authors. The system consists of a power supply, load, and virtual grid hub (VG-Hub) network connecting them. The VG-Hub network must have a large power distribution capability. Therefore, the challenge is to obtain a graph to evaluate the maximum feasible flow when multiple VG-Hubs are connected to a power source or load. In this paper, we propose a method for generating a graph to derive the maximum feasible flow. A quantitative evaluation of the proposed method is presented.

Keywords: maximum flow, hub, battery, virtual grid, VG-Hub, IoT

Classification: Network System

References

[1] N. Morimoto, “Energy-on-demand system based on combinatorial optimization of appliance power consumptions,” Journal of Information Processing, vol. 25, pp. 268–276, Feb. 2017. DOI: 10.2197/ipsjjip.25.268

[2] H. Ichikawa, S. Yokogawa, Y. Kawakita, K. Sawada, T. Sogabe, A. Minegishi, and H. Uehara, “An approach to renewable-energy dominant grids via distributed electrical energy platform for IoT systems,” 2019 IEEE International Conference on Communications, Control, and Computing Technologies for Smart Grids (SmartGridComm), pp. 1–6, 2019. DOI: 10.1109/SmartGridComm.2019.8909762

[3] USB, I.F., USB Power Delivery standard, https://www.usb.org/usb-charger-pd

[4] H. Ichikawa, A. Ahmed, H. Hanafusa, S. Yokogawa, Y. Kawakita, K. Sawada, H. Mikami, and N. Yoshikawa, “Virtual grid for renewable energy society 2015,” IEEE Innovative Smart Grid Technologies – Asia (ISGT ASIA), 2015. DOI: 10.1109/ISGT-Asia.2015.7387117

[5] K. Tamura, Y. Kawakita, Y. Tobe, S. Yokogawa, and H. Ichikawa, “Practical issues in aggregation and distribution of electrical power among USB-PD-connected devices,” Intelligent Environments 2020, vol. 28, pp. 54–61, 2020. DOI: 10.3233/AISE200024
1 Introduction

Recently, there has been a growing interest in renewable energy sources. Among renewable energy sources, solar and wind power have a weakness in that the amount of power generated depends on the weather and is difficult to control. Therefore, batteries are required to alleviate variations in power generation. Batteries are also required to increase the portability of devices, which means that they are required for different purposes, thus creating inefficiency. Combining the batteries through a system with various outputs and capacities is desirable. Morimoto proposed a method to minimize the impact on the user’s quality of life (QOL) by determining the power allocation of home appliances based on various factors and controlling the power supply of home appliances [1]. However, with respect to power control, this method only performed power distribution, not synthesis. We are trying to use the VG-Hub network to distribute power to the load on an arbitrary scale using a group of batteries, which do not necessarily have the same output and capacity [2]. The main challenge is to obtain a graph to evaluate the maximum feasible flow when multiple VG-Hubs are connected to a power source or load. To address the above challenge, we propose a method for generating a graph to derive the maximum feasible flow. We further compare and evaluate the proposed method with one that sequentially adds edges to nodes representing loads and power supply.

2 Virtual grid system

Figure 1(a) shows a virtual grid system. The virtual grid system is an application system in which the power supply and loads are connected to a VG-Hub network. The VG-Hub network is responsible for power distribution and is monitored and controlled by a virtual grid controller (VG controller). The VG-Hub is a device that is controlled by the VG-Hub controller inside the VG-Hub. Furthermore, the VG-Hub communicates with the VG controller through the cloud and distributes power between the power supply and load devices connected to the VG-Hub port, as shown in Fig. 1(a). The VG-Hub ports can be connected to any VG device that complies with the USB-C PD standard. USB-C PD is increasingly being used because of its versatile features to control the flow of power. The USB Type-C connector is bidirectional and nonpolar. Based on the results of the USB PD power rule negotiation, up to 100 W of power flow can be allowed [3]. The VG controller controls the flow of electricity in the VG-Hub by specifying the role and maximum power of each VG-Hub port [4]. The power flow from the port is synthesized with the role of each load device in VG-Hub, and the synthesized power is distributed to VG devices connected to the port with the role of a power supply [5]. The prototype of VG-Hub is shown in Fig. 1(b). VG-Hub is a hardware unit, and the number of
By networking VG-Hub and using some ports of VG-Hub to connect VG-Hub units to each other, we can build a large hub with many ports. An increase in the number of ports used to connect VG-Hub units means that the VG-Hub network can be designed to carry more power flows. A method for generating graphs for maximum flow derivation is presented in the next section.

3 Proposed method for maximum flow derivation

Previously published literature presented a method for creating a VG-Hub network with power load connections from a VG-Hub network with no power and load connections [6]. In addition, the matrix representing the connection status of the power supply and load could be determined by the size of the adjacency matrix representing the VG-Hub network. It was shown that by generating the adjacency
matrix representing the connection status of the power supply and load in advance, it was possible to create a graph of a VG-Hub network with power supply and load connections. In this section, as a comparison to the proposed method, we also explain the sequential method of adding nodes representing power supply and load to VG-Hub one by one.

Assume that there is a VG-Hub network as follows.

There were four VG-Hub units, and they were all interconnected.

The number of ports on each VG-Hub unit was 6. Three ports were used to connect a VG-Hub unit to the rest. The other three ports were connected to the load and power supply.

Two of the four VG-Hub were connected to the load. The remaining two VG-Hub were connected to the power supply. First, we presented a sequential method.

1. Import the adjacency matrix representing the connection status of each VG-Hub unit.

Figure 2(a) shows the graph of the VG-Hub network not connected to the power supply and load based on the VG-Hub connections assumed earlier.

2. For one VG-Hub unit, sequentially add edges to each node for power and load connections. Figure 2(b) shows the graph of the power supply and load connections done using the sequential method.

3. A VG-Hub network with no power supply and load connected, (b) a VG-Hub network created using the sequential method with modified power load connections done using the sequential method, (c) a VG-Hub network created using the sequential method with modified power load connections done using the proposed method.
connected to the VG-Hub using the sequential method based on the connection status of the power supply and load defined in the previous section.

3. Apply the Ford-Fulkerson algorithm to calculate the maximum flow.

Next, we present the proposed method.

1. Import the adjacency matrix representing the connection status of each VG-Hub.

Figure 2(a) shows the graph of the VG-Hub network not connected to the power supply and load based on the VG-Hub connections assumed earlier.

2. Import the right-hand concatenation matrix representing the connection status of the load.

3. Import the lower concatenation matrix representing the connection status of the power supply.

4. Concatenate the right and lower matrices to the adjacency matrix.

5. Add an edge to each node to combine the nodes into one node, load for load, and power for power. Figure 2(c) shows the graph of the power supply and load connected to each VG-Hub unit using the proposed method based on the connection status of the power supply and load defined in the previous section.

6. Apply the Ford-Fulkerson algorithm to calculate the maximum flow.

In the next section, we compare and evaluate the two methods presented in this section.

4 Evaluation

We compared the sequential method with the proposed method and evaluated them using the ordered method and measured values. The number of VG-Hub units is used as the parameter. Let the number of VG-Hub units be $N$ and the number of ports connected to the load and power supply be $A$. Using the order method, the proposed method depends on the number of ports $A$ connected to the load and power supply, which can be expressed as $O(2A)$ since there are power supply and load nodes. The sequential method also depends on the number of VG-Hub units because it sequentially adds edges to the nodes representing the number of ports $A$ and each VG-Hub unit connected to the load and power supply. Thus, the sequential method can be expressed as $O(NA)$. The results are shown in Fig. 3(a). The horizontal axis is the number of VG-Hub units, and the vertical axis represents the computational complexity. It can be seen that the computation time of the proposed algorithm is smaller than that of the sequential algorithm. The blue line in Fig. 3 represents the proposed method, and the orange line represents the sequential method. The measured values were measured in MATLAB using tic and toc functions. The results are shown in Fig. 3(b). The proposed algorithm was faster by approximately 0.002 s during the actual measurement.

5 Conclusion

In this paper, we propose a graph-generation method for maximum flow derivation. The proposed method and sequential method were quantitatively evaluated and compared. The maximum flow rate can be easily derived using the proposed method. However, the proposed method is not necessarily faster than the sequential method.
Therefore, in future work, we plan to increase the number of VG-Hub units to 10 to demonstrate the superiority of the proposed method.

**Fig. 3.** (a) A comparison graph between the proposed method and the sequential method using the order method, (b) a comparison graph between the proposed method and the sequential method by actual measurements connections

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Fairness-aware resource allocation technique for UAV-aided information collection system in a disaster

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Abstract: Unmanned aerial vehicles (UAVs) have garnered attention for disaster response because of their flexibility and mobility. In particular, a UAV-aided disaster communication network allows rescue teams to tackle various missions efficiently and flexibly. In this study, we investigate a UAV-aided information collection system in a disaster-affected area. Then, we propose a bandwidth allocation algorithm that improves the fairness among the ground user equipments (UEs) by utilizing their communication histories and location information. Finally, the simulation results demonstrate the considerable validity of the proposed method compared with the straightforward scheme in terms of fairness.

Keywords: disaster response, OFDMA, resource allocation, UAV

Classification: Wireless Communication Technologies

References

[1] M. Erdelj, E. Natalizio, K.R. Chowdhury, and I.F. Akyildiz, “Help from the sky: leveraging UAVs for disaster management,” IEEE Pervasive Comput., vol. 16, no. 1, pp. 24–32, Jan. 2017. DOI: 10.1109/mperv.2017.11
[2] N. Zhao, W. Lu, M. Sheng, Y. Chen, J. Tang, F.R. Yu, and K.-K. Wong, “UAV-assisted emergency networks in disasters,” IEEE Wireless Commun., vol. 26, no. 1, pp. 45–51, Feb. 2019. DOI: 10.1109/mwc.2018.1800160
[3] American Red Cross, “Drones for disaster response and relief operations,” 2015.
[4] X. Lin, V. Yajnanarayana, S.D. Muruganathan, S. Gao, H. Asplund, H.-L. Maattanen, M. Bergstrom, S. Euler, and Y.-P.E. Wang, “The sky is not the limit: LTE for unmanned aerial vehicles,” IEEE Commun. Mag., vol. 56, no. 4, pp. 204–210, May 2016. DOI: 10.1109/mcom.2018.1700643
[5] R.K. Jain, D.-M.W. Chiu, and W.R. Hawe, “A quantitative measure of fairness and discrimination for resource allocation in shared systems,” Digital Equipment Corporation, Tech. Rep. DEC-TR-301, Sept. 1984.
1 Introduction

An unmanned aerial vehicle (UAV)-aided communication system is one of the most promising solutions for providing communication services in areas where base stations have been damaged by disasters. In addition, UAVs can be used for various purposes depending on the type and stage of a disaster [1, 2, 3]. Particularly in a disaster environment, it is important to collect information from more devices for efficient rescue operations. Thus, UAV-aided disaster communication is required to control communication resources, which prioritize fairness more than usual communication. In this study, we focus on the information collection system of ground user equipments (UEs) by a single UAV using orthogonal frequency division multiple access (OFDMA) and propose a bandwidth resource allocation method for fairness-aware communication. The contributions of this study are as follows: First, we formulate an optimization problem for maximizing the minimum amount of transmitted data of the UEs and explain the difficulty of this problem due to the presence of unknown variables. To address this issue, we introduce the “data quantity function,” defined by utilizing the communication history and location of the UE to reformulate the optimization problem. Furthermore, we propose an algorithm for solving this optimization problem and demonstrate the effectiveness of our method through numerical simulations.

2 System model

As shown in Fig. 1, we consider a UAV-aided information collection system in which a single UAV is employed as a flying base station to collect information from ground UEs. The UE set is denoted by $k \in \mathcal{K} = \{1, \cdots, K\}$ with $|\mathcal{K}| = K$, and they are assumed to exist at a fixed location. We consider a three-dimensional Cartesian coordinate, and the locations of the UAV and $k$-th UE are denoted by $(x_{UAV}, y_{UAV}, H_{UAV})$, and $(x_k, y_k, 0)$, respectively. The UAV is assumed to fly in the $+x$ direction at a fixed altitude $H_{UAV}$ and constant speed $V$. The other assumptions are as follows: First, the UAV can only communicate with the UEs in its coverage and the locations of the UAV and UEs in the coverage are known. Second, the UAV and UEs are connected by adopting OFDMA. Finally, the UAV calculates the bandwidth allocation in each discretized time, which is denoted by $n \in \mathcal{N} = \{1, \cdots, N\}$ with $|\mathcal{N}| = N$, and the length of the time frame is defined by $\delta_T$. In this system, the communication opportunities of UEs depend on their locations with respect to the flight trajectory of the UAV, resulting in different amounts of transmittable data.

![Fig. 1. System model.](image-url)
in the absence of an appropriate resource allocation that ensures fairness. From the perspective of the provided information, we aim to equalize and maximize the amount of transmitted data of UEs to enable high-fairness information collection by employing a fairness-aware bandwidth allocation. To design the bandwidth allocation algorithm, we formulate an optimization problem to maximize the minimum amount of transmitted data among ground UEs.

In general, the line-of-sight (LoS) probability increases as the altitude of the UAV increases. For instance, the LoS probability is more than 95% when the UAV is 120 m [4]. Thus, in this study, we use the free-space pass loss model as the propagation model for simplicity, that is, the channel gain of the $k$-th UE at the $n$-th time frame, $g_{kn}$, is assumed to mainly depend on the distance and can be expressed as follows:

$$g_{kn} = \left(\frac{\lambda}{4\pi l_{kn}}\right)^2; \tag{1}$$

where $l_{kn} = \sqrt{(x_{UAV} - x_k)^2 + (y_{UAV} - y_k)^2 + H_{UAV}^2}$ represents the distance between the UAV and the $k$-th UE in the $n$-th time frame, and $\lambda$ denotes the wavelength of the carrier wave. In the OFDMA system, the UAV can divide the system bandwidth, denoted by $B$, into subcarriers allocated to the UEs in the coverage. Therefore, by letting the bandwidth allocated to the $k$-th UE in the $n$-th time frame as $w_{kn}$, the constraint on the bandwidth resources can be expressed as follows:

$$\sum_{k=1}^{K} w_{kn} \leq B, \quad \forall n \in \mathcal{N}, \tag{2}$$

$$0 \leq w_{kn}, \quad \forall n \in \mathcal{N}, k \in \mathcal{K}. \tag{3}$$

Accordingly, the transmission rate derived from the Shannon–Hartley theorem on the $k$-th UE in the $n$-th time frame, $r_{kn}$, can be expressed as follows:

$$r_{kn} = w_{kn} \log_2 \left(1 + \frac{g_{kn}p_{UE}}{w_{kn}N_0}\right), \tag{4}$$

where $p_{UE}$ and $N_0$ denote the transmit power of the UEs and the spectral density of the additive white Gaussian noise at the UAV receiver, respectively. The amount of transmitted data of the $k$-th UE, $d_k$, is formulated as follows:

$$d_k = \sum_{n=1}^{N} r_{kn}\delta_T. \tag{5}$$

To balance the total amount of transmittable data and the fairness among UEs, we formulate the bandwidth allocation problem as follows:

$$\max_{w_{kn}} \min_k \in \mathbb{R}, \quad \text{subject to} \quad \sum_{k=1}^{K} w_{kn} \leq B, \quad \forall n \in \mathcal{N}, \tag{6.a}$$

$$0 \leq w_{kn}, \quad \forall n \in \mathcal{N}, k \in \mathcal{K}. \tag{6.b}$$

Notably, the optimization problem cannot be directly solved because (6.a) contains variables over all time frames, the values of which are unknown at a specific time because the UAV can only recognize the UEs within its coverage.
3 Proposed method

In this section, we propose a fairness-aware bandwidth allocation algorithm that uses a data quantity function as an alternative to the original objective function (6.a). The data quantity function is required to have the same units as the original objective function, that is, bit, and not to include information about the UEs outside of the UAV coverage. Hence, we define the data quantity function at the \( k \)-th UE in each time frame as follows:

\[
f_k(w_k) = D_k^{\text{past}} + D_k^{\text{now}}(w_k) + D_k^{\text{future}}, \tag{7}
\]

where \( D_k^{\text{past}} \) denotes the communication history of the \( k \)-th UE, that is, the amount of data transmitted to the UAV in the past, and the \( k \)-th UE reports it to the UAV at each time frame. The term \( D_k^{\text{now}}(w_k) \) is the amount of data that can be transmitted in the current frame and is calculated using (4), \( D_k^{\text{future}} \) represents the amount of data that can be transmitted when the \( k \)-th UE is in the communication range, which is estimated as follows:

\[
D_k^{\text{future}} = CT_k, \tag{8}
\]

where \( T_k \) represents the remaining time for the \( k \)-th UE to communicate with the UAV and it can be calculated based on the location of the \( k \)-th UE as follows:

\[
T_k = \frac{x_k - x_{UAV} + \sqrt{R^2 - y_k^2}}{V}, \tag{9}
\]

where \( R \) denotes the radius of the coverage of the UAV. The term \( C \) with a unit of bps is a constant independent of UEs and is defined as follows:

\[
C = \frac{r_{\text{max}} + r_{\text{min}}}{2}, \tag{10}
\]

where \( r_{\text{max}} \) and \( r_{\text{min}} \) represent the maximum and minimum achievable rates in the coverage, respectively. By using (7), the original optimization problem shown in (6.a)–(6.c) is redefined as follows:

\[
\begin{align*}
\text{maximize} & \quad \min_k f_k(w_k), \quad \text{(11.a)} \\
\text{subject to} & \quad \sum_{k=1}^{K} w_k \leq B, \quad \text{(11.b)} \\
& \quad 0 \leq w_k, \quad \forall k \in \mathcal{K}. \quad \text{(11.c)}
\end{align*}
\]

In our scheme, this optimization problem is solved in each time frame by using Algorithm 1, and the flow of the process is as follows: First, the system bandwidth \( B \) is divided into \( M \) narrow bandwidths with a bandwidth equal to \( dw \). Next, the array \( D_0 \) is initialized by the sum of the first and last terms on the right side of (7) as the initial state, and the array \( A \) used for storing the number of narrow bandwidths allocated to each UE is initialized by zero. Then the array \( D \) is used to store the data quantity function value of UEs in each iteration \( i \). Second, one of the narrow bandwidths is allocated to the UE corresponding to the smallest element in \( D \). This process is executed until the entire system bandwidth is fully allocated. This algorithm allocates bandwidth to the UE with the lowest amount of data that can be transmitted in the past and in the future in each iteration, thus increasing fairness among all UEs.
Algorithm 1  Bandwidth allocation algorithm

\[ dw = B / M \]
\[ D_0[k] \leftarrow D_{k}^{past} + D_{k}^{future}, A = [0, \ldots, 0], (|D_0|, |A| = |K|) \]
\[ D[k] \leftarrow D_0[k] \]

for \( i = 1, \ldots, M \) do

\[ k_{\text{min}} \leftarrow k | \arg \min D[k] \]
\[ A[k_{\text{min}}] \leftarrow A[k_{\text{min}}] + 1 \]
\[ D[k_{\text{min}}] \leftarrow D_0[k_{\text{min}}] + A[k_{\text{min}}]dwT \log_2 \left( 1 + \frac{g_{k_{\text{min}}p_{\text{UE}}}}{A[k_{\text{min}}]d_wN_0} \right) \]

end for

4 Simulation results

In this section, we present the results of the simulation using MATLAB and evaluate the performance of the proposed bandwidth allocation method. The UE is generated by following a random uniform distribution on the street; for instance, an example of a UE distribution is shown in Fig. 2(a), and for a given UE distribution, the UAV collects information while flying from \( \left[ 0, 0, H_{\text{UAV}} \right] \) to \( \left[ 970, 0, H_{\text{UAV}} \right] \). Altitude \( H_{\text{UAV}} \), and the velocity, \( V \), of the UAV are set as 100 m and 10 m/s, respectively. The transmit power of the UEs, \( p_{\text{UE}} \), is set as 10 mW. The system bandwidth, \( B \), is 20 MHz, and the frame length, \( \delta T \), is set as 10 ms. The other parameters are \( \lambda = 0.15 \text{ m}, N_0 = -174 \text{ dBm/Hz}, M = 1000, \) and \( R = 65.2 \text{ m} \). We consider the following three bandwidth allocation algorithms for performance comparison.

- **Uniform**- An algorithm allocating the bandwidth equally to UEs within the coverage of the UAV.

- **Prop w/o future**- Algorithm 1 using the data quantity function defined in (7) with the modification of \( D_{k}^{future} = 0 \).

![Fig. 2. Simulation results.](image)
• **Prop w/ future** - Algorithm 1 using the data quantity function defined in (7) without change.

The minimum transmitted data and Jain’s fairness index (FI) are used as performance metrics. FI takes a value from 0 to 1, and a larger value indicates better fairness [5].

Figure 2(b) shows the minimum amount of transmitted data in each method with different number of UEs. The performance of the UE with the worst quality of service is improved by using the proposed method. Furthermore, by comparing “Prop w/o future” and “Prop w/ future,” we observe that the performance can be improved by considering the estimated amount of data to be transmitted in the future. Figure 2(c) shows the values of FI for different numbers of UEs. In this figure, we observe that our bandwidth allocation using the data quantity function enables the highest fairness regardless of the number of UEs.

5 Conclusion

In this study, we focused on a UAV-aided information collection system using OFDMA in a disaster-affected area and proposed a resource allocation method for fair communication. We also introduced the data quantity function defined by the communication history and the location of the UE to formulate the optimization problem at each time frame. Then, we proposed an algorithm to solve this problem. Finally, through numerical simulations, we demonstrated that the proposed method can dramatically improve the fairness among UEs while maintaining high values in the amount of minimum transmitted data.

Acknowledgments

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Crest factor reduction of spectrogram art signals based on clipping and filtering processes

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Abstract: Spectrogram art communications exchange visual messages drawn on sound signals. This study attempts to reduce the crest factor of spectrogram art signals to improve their visibility. This paper describes some experimental results examining our proposed technique that employs clipping and filtering processes.

Keywords: inaudible sound communication, spectrogram art communication, crest factor reduction

Classification: Fundamental Theories for Communications

References

[1] N. Aoki, S. Anazawa, K. Ozeki, Y. Dobashi, K. Ikeda, and H. Yasuda, “Possibilities of visual communications using additive and subtractive spectrogram arts,” International Workshop on Smart Info-Media Systems in Asia 2020 (SISA2020), 2020.
[2] Y. Shibasaki, K. Asami, A. Kuwana, Y. Du, A. Hatta, K. Kubo, and H. Kobayashi, “Study on multi-tone signals for design and testing of linear circuits and systems,” International Conference on Technology and Social Science 2018 (ICTSS2018), 2018.
[3] https://rfmw.em.keysight.com/wireless/helpfiles/n7614/Content/Main/Crest_Factor_Reduction_Concept.htm
[4] S. Abouty, L. Renfa, Z. Fanzi, and F. Mangone, “A novel iterative clipping and filtering technique for PAPR reduction of OFDM signals: System using DCT/IDCT transform,” International Journal of Future Generation Communication and Networking, vol. 6, no. 1, pp. 1–8, 2013.

1 Introduction

Spectrogram art communications are expected to exchange visual messages drawn on sound signals by using microphones and speakers equipped in ordinary laptop com-
puters. It enables near field communications without any advanced wireless radio devices. Transmitting such visual messages, we can realize robust communications that are directly understandable by human users.

Figure 1 shows our pilot implementation of a visual communication system that exchanges spectrogram art signals consisting of text messages. Text messages are easily understandable by human users even if some parts are inevitably degraded due to undesirable environmental noises. It is because text messages are very familiar symbols to human users.

![Fig. 1. Our pilot implementation of a visual communication system. Sound signals are emitted from a personal computer and received by a smartphone. This example employs a commercial software, SpectrumViewPlus for visualizing the text message.](image)

2 Proposed technique

This study employs a DFT (discrete Fourier transform) filtering method to generate spectrogram art signals [1]. This method performs filtering source signals with mask patterns to generate spectrogram art signals. A white noise is one of the best candidates for the source signals. A mask pattern is a spectrogram art itself. Filtering a source signal with a mask pattern is equivalent to be the multiplication of these information in the frequency domain. The inverse transform of the multiplication is a resultant sound signal consisting of a spectrogram art.

The visibility of spectrogram art communications depends on the power of sound signals. If the power of sound signals decreases, the visibility of the messages must be degraded. It happens especially in the case of using low power loud speakers equipped in ordinary laptop computers. To improve the visibility of the messages as much as possible, the crest factor of spectrogram art signals should be reduced appropriately. The crest factor is defined as the peak amplitude of a waveform divided by the RMS (root mean square) value [2]. If the crest factor is minimized, the power of each frequency component included in spectrogram art signals is maximized.

To reduce the crest factor of spectrogram art signals, this study employs a clipping process, one of the approaches proposed for the crest factor reduction so far [3]. This technique is actually the same as a sound effect named compressor employed in the audio signal processing of music production. It is used for maximizing sound signals.
The crest factor of sound signals is reduced by amplifying the sound signals with a compressor.

Even though it is a quite simple approach, such a clipping process surely reduces crest factor. However, this nonlinear process inevitably generates unnecessary frequency components decreasing the visibility of the spectrogram art signals as its side effect. Therefore, this study applies a filtering process to the resultant signals by using the same mask pattern employed in the generation of the spectrogram art signals to decrease such defects.

The proposed technique iterates these processes. Figure 2 shows that the iteration of these processes reduces further the crest factor. It indicates that just a few number of iteration may potentially be enough for obtaining the crest factor that is close to its convergence value.

The proposed technique is rather similar to the conventional iterative clipping and filtering algorithm used to reduce PAPR (peak-to-average power ratio) in OFDM (orthogonal frequency division multiplexing) radio signals [4]. One main difference is that the conventional technique uses a time-invariant rectangular window for the filtering process, while the proposed technique uses time varying mask patterns.

To examine the effectiveness of the proposed technique, a computer simulation was conducted. Figure 3 (a) shows the spectrogram art of a sound signal without applying any crest factor reduction techniques. Its crest factor became 7.6.

Figure 3 (b) shows the spectrogram art of a sound signal after the clipping process. Since our simulation employed a signum function for the clipping process, the crest factor of the resultant signal became 1.0, the theoretically smallest value of the crest factor. However, unnecessary frequency components appear instead. Such annoying noises are not only audible but also decrease the visibility of the spectrogram art signal.

Figure 3 (c) shows the spectrogram art of a sound signal after the clipping and filtering processes of the proposed technique. The number of the iteration was 10. Its crest factor became 3.1. Even though the crest factor increases, the visibility of the spectrogram art signal is much clarified compared with the original one.

Figure 3 (d) shows the spectrogram art of a sound signal after the clipping and
Fig. 3. Spectrogram arts obtained in a computer simulation: (a) the original, (b) with clipping process, (c) with clipping and filtering processes of the proposed technique, and (d) with clipping and filtering processes of the conventional technique. The crest factors are 7.6, 1.0, 3.1, and 2.0 respectively.

filtering processes of the conventional technique using a time-invariant rectangular window for the filtering process. To reduce audible noises under 18 kHz, the high pass filter of 18 kHz cut-off frequency was employed. The number of the iteration
was 10. Its crest factor became 2.0. As shown in the figure, the spectrogram art is contaminated with noises, even though its crest factor is reduced much more than the proposed technique. It indicates that the conventional technique cannot reduce unnecessary frequency components very well, so that it may not be suitable for improving the visibility of the spectrogram art signals.

3 Conclusions

The simulation indicates that the proposed technique may potentially be an appropriate method that improves the visibility of spectrogram art signals. Further quantitative experiments will be performed to confirm how the proposed technique appropriately works in real situations.
Faster channel allocation by relaxing the threshold of the terminal judgment of DCRO control and reducing the number of initial calibrations in backscatter synchronous streaming protocols

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Abstract: Recently, multi-point synchronous sensing, which is used for structural health monitoring, is desirable to allocate channels to many sensor tags in a short time. In this study, we have proposed an approach for accelerating channel allocation in the synchronous streaming protocol. A sensor tag uses subcarrier frequency for backscatter communication. The sensor tag requires recursive calibration of the subcarrier frequency due to the instability of the subcarrier frequency oscillator inside the tag’s circuit. We have demonstrated that recursive calibration of subcarrier frequency is a dominant factor in channel allocation time. Furthermore, we found that by combining the reduction of initial calibrations and threshold’s relaxation for terminal judgment, as well as its quantitative evaluation, Experimental evaluation shows that the proposed approach can accelerate channel allocation by up to 43% compared with existing approach.

Keywords: RFID, channel allocation, accelerating, over-the-air (OTA)
Classification: Wireless Communication Technologies

References

[1] J. Mitsugi, Y. Kawakita, K. Egawa, and H. Ichikawa, “Perfectly synchronized streaming from multiple digitally modulated backscatter sensor tags,” *IEEE J.*
1 Introduction

The authors have been researching and developing Multiple Subcarrier Multiple Access (MSMA), which is an extension of ISO/IEC18000-63, an RFID international standard protocol, for multi-point synchronous streaming over multiple subcarrier channels [1, 2, 3]. The MSMA system consists of a sensor tag that serves as an information medium, a reader/writer antenna that reads and writes sensor tag data, and a host signal processing server that handles several reader/writer antennas. The MSMA system allocates different subcarrier frequencies to the sensor tag, and the signal processing server removes the mutual interference between the subcarrier frequencies. In this approach, the MSMA system achieves pseudo-FDMA and performs multi-point synchronous streaming with multiple sensor tags. At the allocated subcarrier frequency, the sensor tag communicates in backscatter. The sensor tag does not require a battery because it is wirelessly powered by backscatter communication. The sensor tag generates a subcarrier frequency by dividing the frequency oscillated by the digital controlled ring oscillator (DCRO) embedded in the sensor tag’s internal circuit. However, the DCRO frequency is unstable due to the process, power supply voltage, and temperature (PVT characteristic) of the tag [4].

Thus, the MSMA system performs a recursive DCRO control word calibration to produce the acceptable DCRO frequency. The MSMA system will stop the recursive DCRO calibration if the sensor tag’s DCRO frequency falls below the threshold for terminal judgment of DCRO calibration (hereafter referred to as the...
threshold). So far, we have shown over-the-air (OTA) allocation in the MSMA system [5]. Figure 1 (a) depicts the DCRO frequency’s convergence transition caused by recursive DCRO calibrations. The vertical axis shows the error from the set DCRO frequency, and the horizontal axis shows the number of DCRO calibrations. The red box shows the initial calibration stage of the DCRO frequency, where the actual relationship between a given DCRO control word and the produced subcarrier frequency is estimated for the subsequent convergence stage. The blue box shows the convergence stage of the DCRO frequency in which the DCRO control word is asymptotically adjusted. Figure 1 (a) shows at subcarrier frequencies of 240 kHz (DCRO frequency 1.92 MHz), 320 kHz (1.28 MHz), and 400 kHz (1.60 MHz) is converged within the threshold that is set to 1 kHz, respectively. According to the results, the sensor tag requires DCRO calibrations from 18 to 24 times. The recursive DCRO calibration is the dominant factor of the allocation time in a channel. Figure 1 (b) shows the packet error rate (PER) for synchronous streaming of 100,000 packets and the average time for 30 times channel allocations. As shown in Fig. 1 (b), each subcarrier frequency maintains a PER of $10^{-5}$ and takes from 240 to 314 ms for channel allocation. The MSMA system is expected to be used for structural health monitoring such as civil structures and machines, and the ability to perform an accelerated channel allocation is desirable.

In order to reduce the channel allocation time, the authors have worked on accelerating channel allocation by relaxing the threshold for the terminal judgment of DCRO control [6] and by reducing the initial calibrations in DCRO control [7]. This letter addresses the issue of PER degradation caused by accelerating channel allocation.

![Graph showing frequency transitions with recursive DCRO calibration.](image1)

(a) Frequency transitions with recursive DCRO calibration.

| Subcarrier (DCRO Frequency) | Channel allocation time | PER (Packet Error Rate) |
|-----------------------------|-------------------------|-------------------------|
| 240 kHz (1.92 MHz)          | 240 ms                  | $3.00 \times 10^{-3}$   |
| 320 kHz (1.28 MHz)          | 314 ms                  | $3.99 \times 10^{-5}$   |
| 400 kHz (1.60 MHz)          | 268 ms                  | $2.00 \times 10^{-3}$   |

(b) Validity of the subcarrier parameter in PER.

**Fig. 1.** Demonstration of synchronous streaming with MSMA.
allocation and the combination of two acceleration approaches and their quantitative evaluation. The remaining sections are organized as follows. Section 2 describes the accelerating approach for channel allocation. Section 3 presents a quantitative evaluation of the accelerating approach using a single sensor tag. Section 4 concludes this study.

2 Combined approach for Accelerating Channel Allocation in Synchronous Streaming Protocols

This chapter describes two approaches for accelerating channel allocation. The relaxation of a DCRO calibration threshold is the first approach. Relaxing the threshold of recursive DCRO calibration helps accelerate channel allocation. However, this acceleration approach may degrade the PER in the demodulator at the later stage of the reader/writer’s internal circuit if the subcarrier frequency used by the sensor tag has a large error. Literature [6] experimentally delivered the threshold in DCRO control as a parameter that a threshold of 64 kHz is the most effective parameter based on the comparison of the deterioration of PER against the reduction of channel allocation time. Therefore, we configured setting the threshold of DCRO calibration to 64 kHz.

Reducing the number of initial calibrations in DCRO calibration is the second approach. The DCRO frequency of the sensor tag is unstable due to variations in PVT characteristics. In initial stage, to asymptotically adjust the target value from the error in the DCRO frequency of the sensor tag, the reader/writer intentionally overshoots the DCRO control with a large swing. The possibility and effect of reducing the initial calibrations were investigated because the quality assurance during the manufacturing of the tag chip results in less instability of the DCRO frequency [7]. By combining the above two approaches, this study analyzes the effectiveness of accelerating.

3 Evaluation

In order to verify the effectiveness of the combined acceleration of the two accelerating approaches, we used a single sensor tag to evaluate the channel allocation time. The sensor tag uses a subcarrier frequency of 240 kHz (DCRO frequency 1.92 MHz) and 320 kHz (1.28 MHz) and 400 kHz (1.60 MHz). The threshold for recursive DCRO calibration was relaxed from 1 kHz to 64 kHz. We reduced the number of iterations at the initial calibration stage from eight to three. We used 40 kbps as bit rate parameter. We conducted 30 times OTA allocations at each subcarrier frequency and measured the average channel allocation time. As a quantitative evaluation of synchronous streaming, we also measured the PER at 100,000 packets.

Figure 2 shows the experimental environment of a combination of accelerating approaches. A prototype reader/writer and a prototype sensor tag comprise the experimental system. The prototype reader/writer was implemented using the NI PXI system, USRP-2940R, LabVIEW NXG 5.0, and the reader/writer antenna (hereafter referred to as the antenna). A single transmitter antenna was used and two receiver antennas were set for receive diversity. The transmitter antenna output a modulated signal at 916.8 MHz, and the receive diversity antenna received the subcarrier
The sensor tag was attached to the tripod and fixed in place. The height of the receiver antenna from the ground is 170 cm. The height of the transmitter antenna from the ground is 110 cm. The distance between the antennas is 60 cm. The distance between the tag and the antenna is 180 cm. The height of the tag from the ground is 110 cm.

Figure 3 shows the results of DCRO calibration, PER and channel allocation.
time measurements using a combination of accelerating approaches. Figure 3 (a) shows the DCRO frequency transition with the reduction in the number of initial DCRO calibrations and threshold relaxation. As shown in Fig. 3 (a), each subcarrier parameter is settled to a DCRO frequency error within the threshold of 64 kHz from four to seven times the DCRO calibration. Figure 3 (b) shows that the PER is the best parameter when the bit rate is 40 kbps and is kept at $10^{-5}$. At each bit rate, the average channel allocation time was 280 ms. According to the results, the combination of relaxing the threshold of the DCRO calibration and reducing the number of initial calibrations resulted in 200 ms reduction of the channel allocation time. Figure 3 (b) shows the PER and channel allocation time measurement results at each DCRO frequency. Figure 3 (b) shows that the PER is kept at $10^{-5}$ even after applying the two accelerating approaches. At each subcarrier, the average channel allocation time was from 86 to 134 ms. According to the results, the combination of relaxing the threshold of the DCRO calibration and reducing the number of initial calibrations resulted from 240 ms to 86 ms at 240 kHz, from 314 ms to 134 ms at 320 kHz, and from 268 ms to 102 ms at 400 kHz at each subcarrier frequency. The results showed that at each subcarrier frequency, the channel allocation time at 320 kHz had the largest reduction of 43%.

4 Conclusion

We proposed the accelerating channel allocation approach by relaxing the DCRO calibration threshold and reducing the number of initial calibrations in the backscatter synchronous streaming protocol. The relaxation of the threshold for DCRO calibration can result in the degradation of PER. In the combination of the accelerating approaches, we examined the bit rate as a parameter and found that the channel allocation time can be reduced from 480 ms to 280 ms while retaining the PER at $10^{-5}$. This result indicates that the number of channels to be allocated per unit time can be increased. Furthermore, a bit rate of 40 kbps is the most effective parameter for maintaining PER at a subcarrier frequency of 240 kHz. Experimental evaluation shows that the proposed approach can accelerating channel allocation by up to 43% compared to existing approach.

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Design and evaluation of antenna pointing control system onboard fixed-wing UAV to realize video transmission relay station

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Abstract: Unmanned Aerial Vehicles (UAV) are used for many services in various fields. We propose a video transmission relay system in which a fixed-wing UAV is used as a relay station. In order to establish the video transmission link, it is necessary to accurately point the antenna onboard a fixed-wing UAV to both the target ground station and the UAV for photography while making the UAV turn over the designated area. In this study, we propose a new antenna pointing control system, which uses a 2-axis gimbal to greatly reduce the attitude motion of the UAV, and describe evaluation results of the performance of the antenna pointing control system.

Keywords: fixed-wing UAV, antenna pointing control system, 2-axis gimbal, video transmission relay station

Classification: Navigation, Guidance and Control Systems

References

[1] Y. Xing, F. Hsieh, A. Ghosh, and T.S. Rappaport, “High altitude platform stations (HAPS): architecture and system performance,” 2021 IEEE 93rd Vehicular Technology Conference (VTC2021-Spring), pp. 1–6, April 2021. DOI: 10.1109/vtc2021-spring51267.2021.9448899
[2] P. Di Vito, D. Fischer, R. Rinaldo, and L. Duquerroy, “HAPs operations and service provision in critical scenarios,” SpaceOps Conference, 2018. DOI: 10.2514/6.2018-2504
[3] M. Ueba and T. Uemura, “Study on autonomous flight control for tracking meander trajectory for observation of farm land by fixed-wing UAV and flight verification,” Proceedings of the 58th Aircraft Symposium, JSASS-2020-5022, Nov. 2020.
[4] D. Kingston, R. Beard, T. McLain, M. Larsen, and W. Ren, “Autonomous vehicle technologies for small fixed wing UAVs,” 2nd AIAA “Unmanned Unlimited” System, Technologies, and Operations, AIAA-2003-6559, Sept. 2003. DOI: 10.2514/6.2003-6559
[5] Y. Takaku, Y. Abe, M. Ueba, S. Kitazawa, and K. Higuchi, “Performance evaluation experiment of highly accurate and responsive tracking antenna control
system for simultaneous observation by unmanned air vehicles,” IEICE Technical Report, SAT2016-66, Feb. 2017.

[6] K. Hamajima, K. Yasukawa, and M. Ueba, “Evaluation of antenna pointing control error onboard fixed-wing unmanned aerial vehicle for a realization of video transmission relay station,” 59th Aircraft Symposium, 3C03, Nov.-Dec. 2021.

[7] H. Hojo, “Effect of antenna drives on attitude control systems,” Systems, Control and Information, vol. 1, no. 4, pp. 137–143, 1988. DOI: 10.5687/iscie.1.137

1 Introduction

Unmanned Aerial Vehicle (UAV) have recently been used to provide many kinds of services, such as monitoring crops and forest; inspecting tunnels, bridges, and buildings; measuring locations; transporting commercial goods; and relaying video and data [1, 2]. Some are actually being provided, some of them are experimentally tried and under research and development. In response to the need for high image quality and long-distance transmission by UAV, frequencies in the 169 MHz, 2.4 GHz, and 5.7 GHz bands have been legally established and are expected to be applied to various services.

The above services are often carried out by a multi-copter type UAV by using direct radio waves. The number of mountain accidents has increased in recent years, and debris flows and landslides caused by record heavy rains are happening frequently. In cases of providing search services for victims in mountainous areas or observing a disaster area, radio waves are often blocked by complex terrain and it is difficult for the UAV to receive the transmission image or the maneuvering command. In addition, victims and disaster areas should be observed from a long-distance point, several kilometers away, with a wide area well over 1 km square. Generally, a multi-copter type UAV can fly for only 20 minutes. Therefore, it is not appropriate for flying distant and vast areas.

Taking those situations into consideration, we propose a video transmission system by a fixed-wing UAV [3, 4] appropriate for long flights (Fig. 1). The proposed system makes use of the 5.7 GHz band for a relay device. The fixed-wing UAV stays in a predetermined area by turning instead of hovering. This turning flight causes the direction of the antenna onboard the fixed-wing UAV to change in both elevation and azimuth angle. Therefore, to make these antennas point to the predetermined ground station and the UAV for photography, the fixed-wing UAV should be equipped with a 2-axis mechanical gimbal [5] so as to reduce the attitude motion of the UAV. The antenna is driven by the gimbal while it is influenced by the UAV motion. Besides as the UAV continues to turn, the command to the antenna direction continues to change [6].

In this paper we propose an antenna pointing control system, which consists of a fixed-wing UAV control system and an antenna driving control system driven by a 2-axis gimbal system. We describe complex equation of motions including the UAV and the gimbal and evaluation results of antenna pointing control error by computer simulations.
2 Antenna pointing control system

2.1 Configuration

An antenna pointing control system consists of two independent control systems: an antenna driving control system and a flight control system (Fig. 2). Since this antenna is attached to the bottom of the fixed-wing UAV, the attitude angle error of the fixed-wing UAV is directly added to the pointing direction error of the mounted antenna as shown in Fig. 2.

From the acceptable link margin, we determine the target error of the antenna pointing control in this system to be ±3 degrees on the transmitter side, and ±1.3 degrees on the receiver side.

2.2 Target command of the antenna pointing direction

In the proposed system, two pieces of 2-axis gimbals with directional antennas are attached to the underside of the fixed-wing UAV. These point to the ground station
and the target fixed-wing UAV independently by driving the azimuth and elevation of each 2-axis gimbal. In addition, the target command input to the control system is calculated based on the position, attitude of the fixed-wing UAV, and the position of the ground station and the UAV for photography. Moreover, this command is not a constant value and changes from moment to moment.

This time, assuming one 2-axis gimbal for transmission between the relay fixed-wing UAV and the ground station, the target commands for the elevation and azimuth of the 2-axis gimbal were derived using the positional relation in Fig. 1. The elevation angle target command $\theta_{GC}$ is shown in Eq. (1), and the azimuth target command $\psi_{GC}$ is shown in Eq. (2).

$$\theta_{GC} = \sin^{-1} \frac{H \cos \phi + (R + D \sin \psi_P) \sin \phi}{\sqrt{R^2 + D^2 + H^2 + 2RD \sin \psi_P}}$$

$$\psi_{GC} = \cos^{-1} \frac{-D \cos \psi_P}{\sqrt{(D \cos \psi_P)^2 + (H \sin \phi - (R + D \sin \psi_P) \cos \phi)^2}}$$

2.3 Equations of motion and design

The fixed-wing UAV used in the proposed system point the antenna using the 2-axis gimbals. To design the system, it is very important to derive equations of motions for the UAV and the gimbal.

Rotational motions around each axis of the fixed-wing UAV and 2-axis gimbal are derived as shown in Eq. (3). First three lines indicate the attitude motions of the UAV and the next two lines indicate the angle motions of the 2-axis gimbal.

$$\begin{bmatrix}
I_{xx} & 0 & 0 & -I_{GY} \sin \psi_{AZ} & 0 \\
0 & I_{yy} & 0 & I_{GY} \cos \psi_{AZ} & 0 \\
0 & 0 & I_{zz} & 0 & I_{GZ} \\
-I_{GY} \sin \psi_{AZ} & I_{GY} \cos \psi_{AZ} & 0 & I_{GY} & 0 \\
0 & 0 & I_{GZ} & 0 & I_{GZ}
\end{bmatrix}\begin{bmatrix}
\dot{\psi}_G \\
0 \\
0
\end{bmatrix} = \begin{bmatrix}
L \\
M \\
N \\
T_{GY} \\
T_{GZ}
\end{bmatrix}$$

From equations of motion, the attitude motion of the fixed-wing UAV and the angle motion of the 2-axis gimbals interacts with each other through angular accelerations. However, the amount of influence between two kinds of equations of motion is judged to be very small by using interference factor [7]. Therefore, it is found that the flight control system and the antenna driving control system can be designed independently.
3 Simulations

3.1 Conditions

Based on Eq. (3) and the control design method as the two independent control systems, the simulation of the antenna pointing control system was carried out using Matlab & Simulink. We assumed a fixed-wing UAV with a weight of 3.03 kg, a wingspan of 1.6 m, and a moment of inertia of $I_{xx} = 0.23$, $I_{yy} = 0.15$, $I_{zz} = 0.34 \text{ kg} \cdot \text{m}^2$ as the simulation conditions. The condition for the 2-axis gimbal was $I_{GY}, I_{GZ} = 0.01 \text{ kg} \cdot \text{m}^2$.

The turning radius and the altitude, turning flight speed and the horizontal distance between the UAV and the ground station were set to be 100 m, 100 m, 22 m/s, 3 km respectively. In addition, the wind disturbance was set to 3 m/s. Furthermore, noises for sensors were considered as random variables with a zero mean and a specified variance $\sigma^2$. The variances for attitude angles, altitude, beam direction, aircraft speed and GPS position were set to be $(0.5 \text{ degrees})^2$, $(0.2 \text{ m})^2$, $(0.01 \text{ degrees})^2$, $(0.17 \text{ m/s})^2$ and $(1 \text{ m})^2$, respectively.

3.2 Results

Figure 3 shows results of the simulation under the above conditions. Figure 3 (a) and (b) show the antenna pointing control error, and (c), (d), and (e) show the attitude angle profile of the fixed-wing UAV for relay. These antennas pointing control error are defined as in Eq. (4) and Eq. (5) using target command in Eqs. (1) and (2) and

![Antenna pointing control error](image)

(a) Pointing control error (Elevation)

![Antenna pointing control error](image)

(b) Pointing control error (Azimuth)

![Fixed-wing UAV Roll angle](image)

(c) Fixed-wing UAV Roll angle

![Fixed-wing UAV Pitch angle](image)

(d) Fixed-wing UAV Pitch angle

![Fixed-wing UAV Yaw angle](image)

(e) Fixed-wing UAV Yaw angle

Fig. 3. Antenna pointing control error and UAV attitude
antenna pointing direction $\theta_A$ for elevation and $\psi_A$ for azimuth.

\[
\Delta \theta_A = \theta_A - \theta_{GC} \quad (4)
\]

\[
\Delta \psi_A = \psi_A - \psi_{GC} \quad (5)
\]

$\sigma_{\theta_A}$ and $\sigma_{\psi_A}$ are standard deviation of antenna pointing control error.

### 3.3 Evaluation of pointing control error

From results in Fig. 3, the standard deviation of antenna pointing control error was ±0.54 degrees for the elevation angle and ±0.67 degrees for the azimuth angle, even with the attitude fluctuation of the fixed-wing UAV.

Therefore, the target antenna pointing control error of the transmitter side and receiver side are satisfied for both elevation and azimuth angles.

### 4 Conclusion

We proposed a video transmission system using a fixed-wing UAV, and configured an antenna pointing control system consisting of the fixed-wing UAV and 2-axis gimbal. We also clarified the motion of the UAV and the 2-axis gimbal onboard antenna, and confirmed by simulation that the antenna pointing control error satisfied well with the required pointing control accuracy regardless of the attitude motion of the fixed-wing UAV. Based on these results, we will carry out experiments using an actual model airplane with a 2-axis gimbal onboard.

### Acknowledgments

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Potential of cooperative SIC for uplink NOMA in multi-cell network

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Abstract: Uplink non-orthogonal multiple access (NOMA) has attracted considerable attention as a key technology for beyond 5G/6G telecommunication. However, when adopting the uplink NOMA in multi-cell networks where an uplink signal may reach multiple base stations (BSs), the successive interference cancellation (SIC) process must be completed at every BS, resulting in smaller throughput owing to bottlenecks. To cope with this issue, we propose a novel approach called cooperative-SIC (C-SIC), where SIC processes at different BSs are mutually combined by exploiting signal transfer through backhaul links, which can prevent some signals from experiencing bottlenecks. Through performance comparison with the conventional NOMA, we reveal the significant potential of C-SIC for the uplink NOMA in a multi-cell network.

Keywords: cooperative interference cancellation, C-SIC, NOMA, power allocation, SIC, signal transfer, uplink

Classification: Wireless Communication Technologies

References

[1] S. Sen, N. Santhapuri, R.R. Choudhury, and S. Nelakuditi, “Successive interference cancellation: a back-of-the-envelope perspective,” Hotnets-IX: Proceedings of the 9th ACM SIGCOMM Workshop on Hot Topics in Networks, Article No. 17, pp. 1–6, Oct. 2010. DOI: 10.1145/1868447.1868464
[2] W. Mei and R. Zhang, “Uplink cooperative NOMA for cellular-connected UAV,” IEEE J. Sel. Topics Signal Process., vol. 13, no. 3, pp. 644–656, June 2019. DOI: 10.1109/jstsp.2019.2899208
[3] NTT DOCOMO, INC., https://www.soumu.go.jp/main_content/000527890.pdf, accessed Feb. 17, 2022.
[4] ITU-R P.1411-8: “Propagation data and prediction methods for the planning of short-range outdoor radiocommunication systems and radio local area networks in the frequency range 300 MHz to 100 GHz,” July 2015.
1 Introduction

Resource allocation in uplink non-orthogonal multiple access (NOMA) is performed by controlling the transmit power of user equipments (UEs). The transmit power of each UE is calculated from the propagation attenuation of the UEs’ signals and the signal-to-interference-plus-noise ratio (SINR) required to meet the target throughput [1]. In the case of uplink NOMA, where the UEs’ signals can reach multiple base stations (BSs), the throughput is dominated by the BS with the lowest SINR. This is because successive interference cancellation (SIC) at each BS must decode the signals of the UEs. Throughput bottlenecks can be prevented by transferring the signal from the BS with the best SINR to other BSs with lower SINRs. Signal transfer allows SIC to be performed without decoding at the other BSs [2].

In this study, we aim to reveal the potential performance of the uplink NOMA employing cooperative-SIC (C-SIC), which exploits the signal transferred between BSs. First, we clarify the mechanism of the throughput bottleneck occurring in uplink NOMA. Then, the advantage of mutually combined SIC processes in C-SIC is demonstrated, and the derivation method of transmit power allocation is provided. Finally, we demonstrate the performance of C-SIC through numerical analysis and compare it to the conventional approaches.

2 Cooperative SIC for uplink NOMA

2.1 Communication capacities under inter-cell interference

In this study, we use the parameters achieved through standardization of the 3.7 GHz band [3], and the channel model is assumed to be a propagation model that integrates theoretical and experimental models for standardization [4]. This model is based on the two-wave model for line-of-sight communications over $R_s$ [m] to 1000 m. In an urban environment with low antenna height and heavy traffic, the propagation attenuation breakpoint disappears due to the unstable reception of the reflected wave. In real-world communications, attenuation varies with time due to fading, but in this study, we use the median value as the propagation attenuation, ignoring the effects of fading and thermal noise in the transmitter and receiver. We assume that the interference power from outside the cell is constant. We considered the uplink communication between two BSs and four UEs, i.e., UE1 and UE2, and UE3 and UE4, corresponding to BS1 and BS2, respectively, and the cells are defined as in Fig. 1(a). A backhaul link with sufficient bandwidth connects BS1 and BS2, and both BSs can receive the channel response of all the UEs. Assume that UE$i$ transmits signals in the $F$ [MHz] band with the $B$ [MHz] bandwidth and the transmit power $p_i$ [dBm] less than or equal to the maximum transmit power $p_{\text{max}}$ [dBm]. The average interference power arriving at a BS from outside both the cells is $\sigma^2$ [dBm/MHz]. The propagation attenuation from UE$i$ to BS$j$, $L_{i,j}$, is given by Eq. (1), where $d_{i,j}$ [m], $\lambda$ [m], and $R_s$ [m] denote the distance from UE$i$ to BS$j$, carrier wavelength, and a parameter of the reference distance, respectively [4].

$$L_{i,j} \text{(dB)} = -20 \log_{10} \frac{\lambda}{2\pi R_s} - 30 \log_{10} \frac{d_{i,j}}{R_s} - 6.$$ (1)
The communication capacity from UE \(i\) to BS \(j\), \(C_{i,j}\) [Mbps], is defined based on the SINR observed at BS \(j\) from UE \(i\), \(\text{SINR}_{i,j}\), as formulated in Eq. (2), where \(K_{i,j}\) represents a set of indices of UEs with lower received power levels than that of the UE \(i\) at BS \(j\).

\[
C_{i,j} = B \log_2(1 + \text{SINR}_{i,j}). \quad (2a)
\]

\[
\text{SINR}_{i,j} = \frac{L_{i,j}p_i}{B\sigma^2 + \sum_{k \in K_{i,j}} L_{k,j}p_k}. \quad (2b)
\]

### 2.2 Cooperative SIC to tackle the bottleneck issue

Using the example of power allocation and coordinates presented in Fig. 1(b), the SIC process and the order of the received signal strength (the received order) in each BS can be summarized as shown in Fig. 1(c). For all the signals to be decoded without error, the rates of UEs must be less than the capacity for decoding. Therefore, in the processes shown in Fig. 1(c), UE\(i\)’s achievable throughput, \(R_i\), is calculated as Eq. (3). Even if the SINR at the corresponding BS is sufficiently large, throughput bottlenecks still occur because of the low SINR at another BS.

In contrast, Fig. 2 shows the combined SIC process when C-SIC is applied to the scenario depicted in Fig. 1(c). In this case, a throughput bottleneck occurs only at UE4. Because the communication capacity in each decoding step is obtained from Eq. (2), the achievable throughput is calculated as Eq. (4).
Fig. 2. A combined SIC process with transfer.

\[
\begin{align*}
R_1 &= \min (C_{1,1}, C_{1,2}), \\
R_2 &= C_{2,1}, \\
R_3 &= \min (C_{3,1}, C_{3,2}), \\
R_4 &= \min (C_{4,1}, C_{4,2}). \\
\end{align*}
\]

(3)

\[
\begin{align*}
R_1 &= C_{1,1}, \\
R_2 &= C_{2,1}, \\
R_3 &= C_{3,2}, \\
R_4 &= \min (C_{4,1}, C_{4,2}). \\
\end{align*}
\]

(4)

### 2.3 Optimized transmit power allocation

Equation (4) shows the achievable UE throughputs, and the smallest among $R_1$ to $R_4$ could be varied by adjusting the transmit power allocation. In other words, the maximized UE’s minimum rate and the transmit power allocation vector achieving it can be obtained by solving the maximin optimization problem as shown in Eq. (5). It is evident from Eqs. (3)–(5) that the presence of bottlenecks affects the optimal solution.

\[
\begin{align*}
\text{maximize} & \quad \min_i R_i, \\
\text{subject to} & \quad p \in \mathbb{P} = \{ p | p_1, p_2, p_3, p_4 \leq p_{\text{max}} \}.
\end{align*}
\]

(5)

### 3 Numerical analysis of cooperative SIC

#### 3.1 The assumed combined SIC process and optimization method

The combined SIC process depicted in Fig. 2 is an example of the possible methods of exploiting signal transfer in C-SIC. Although an alternative process can be adopted, we adopt the following strategy to design the combined SIC process for simplicity in this study. When both the BSs need to cancel a signal that is not of the corresponding UE, the BS1 preferentially decodes the UE, and the process proceeds to the next stage. For example, as shown in the second stage of the combined process in Fig. 2, the BS1 decodes the UE4, and the process proceeds to the next stage; accordingly, the BS2 generates a replica of the UE1’s signal based on the transferred signal.

For deriving the optimized transmit power allocation by solving the optimization problem in Eq. (5), we employ the following parallelization technique to determine...
the global solution using the fminimax in MATLAB quickly. The objective function in Eq. (5) is discontinuous at the points where the received order changes; there are 576 possible combinations of the received order in the scenario involving two BSs and four UEs. Therefore, we divide the optimization problem according to the received order, and add Eq. (6) as a constraint, which allows us to determine a solution in each received order relatively easily because of the reduced number of local solutions in each search space. Note in Eq. (6) that \( s_{(n,j)} \) shows an index of the UE with the \( n \)-th largest received signal strength at BS\( j \). The global optimal solution can be quickly determined by choosing the solution with the largest objective function from all obtained solutions.

\[
L_{s_{(n+1,j)}} \times p_{s_{(n+1,j)}} \leq L_{s_{(n,j)}} \times p_{s_{(n,j)}}, \quad \forall n = 1, 2, 3, \forall j = 1, 2. \tag{6}
\]

### 3.2 Minimum rate comparison

We compared the minimum UE rate achieved by NOMA with C-SIC, namely the proposed NOMA, and conventional methods in the same environment. In the conventional NOMA, the UEs in the same cell share a spectrum, and each cell uses a different frequency of the same bandwidth. In OMA, all the UEs transmit signals with the same transmit power, \( p_{\text{max}} \), using the frequency bandwidth allocated to each UE. The parameters are shown in Fig. 3(a), where \( h_1 \) and \( h_2 \) represent the heights of the BS and UE, respectively. The analysis is based on 1000 samples obtained through simulations by randomly changing the distribution of the UEs. The performance

| Parameters          | Value |
|---------------------|-------|
| \( F \) [MHz]       | 3700  |
| \( R_j \) [m]       | 20    |
| \( h_1 \) [m]       | 4     |
| \( h_2 \) [m]       | 1.6   |
| Traffic             | Heavy |
| Propagation         | Outdoor, LoS |
| \( B \) [MHz]       | 100   |
| \( p_{\text{max}} \) [dBm] | 23 |
| \( \sigma^2 \) [dBm/MHz] | -110 |
| \( r \) [m]         | 500   |

(a) Parameters for simulation.

(b) Minimum rate comparison. (c) Out-of-cell interference comparison.

Fig. 3. Performance evaluations.
ratio is calculated by normalizing the minimum rate of the proposed and conventional NOMA based on the minimum rate of the OMA in each sample. The cumulative distribution function (CDF) is plotted as shown in Fig. 3(b), where a ratio smaller than one indicates a lower performance than that of the OMA, and a ratio greater than one implies that the method outperforms the OMA. In the CDF, we removed four samples where UEs are distributed within 20 m of each BS, because the domain of definition in Eq. (1) is \( d_{i,j} \geq R_s = 20 \). The average, minimum, and maximum performance ratios are shown in Fig. 3(b). These results show that the proposed NOMA outperforms the OMA in 81.53% of the samples.

### 3.3 Interference power comparison

Under the same environment and parameters as in the previous subsection, we compare out-of-cell interference. We calculated the average interference power density at a point \( x \) [m] away from the center point between the two BSs and plotted the average over 996 samples, as shown in Fig. 3(c). In the proposed NOMA, all the UEs transmit signals using the full bandwidth; therefore, the interference power density is constant throughout the band. In conventional NOMA and OMA, each UE transmits using different frequencies; therefore, the interference power densities differ depending on the frequency. In Fig. 3(c), the conventional NOMA and OMA are plotted as the mean of the interference power density averaged over the frequencies. The interference power density from outside both the cells assumed in this environment is represented by a dotted line. This result shows that the proposed NOMA exhibits lower out-of-cell interference than that of the other methods.

### 4 Conclusion

In this study, we revealed the potential performance of C-SIC for uplink NOMA in adjacent cells. We described a method for preventing throughput bottlenecks via signal transfer using a backhaul link between BSs, and a method for solving the optimization problem considering the combined SIC process. Through numerical analyses, we demonstrated that the proposed method achieved higher minimum rates than conventional methods in more samples, while displaying less out-of-cell interference. Thus, the throughput can be improved by reusing the spectrum more efficiently. The uplink NOMA with C-SIC has significant room for further improvement in performance and feasibility by revisiting the scheduling strategy in C-SIC, the optimization technique for transmit power control, the highly efficient backhaul utilization method. This work was supported in part by Japan Society for the Promotion of Science KAKENHI Grant Number JP20K11785.
Proposal of flexible service deployment environment for the edge processing system

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Abstract: The key to realizing a smart city is the introduction of flexible services using a large number of IoT devices. Still, most research has focused on services using dedicated devices with limited applications. This research proposes an edge framework and cloud services using general-purpose devices. Specifically, we suggest implementing a new service deployment environment to support independent service installation for the edge processing system. As a prototype, we have implemented a surveillance camera service and measured the execution performance required to provide the service.

Keywords: Internet of Things, smart city, edge processing system, flexible service deployments

Classification: Internet

References

[1] A. Giri, S. Dutta, S. Neogy, K. Dahal, and Z. Pervez, “Internet of things (IoT): a survey on architecture, enabling technologies, applications and challenges,” Proceedings of the 1st International Conference on Internet of Things and Machine Learning, Article No. 7, 2017. DOI: 10.1145/3109761.3109768
[2] A. Nakamura, K. Naito, and T. Yamazato, “Proposal of an edge processing system supporting flexible service deployments,” IEEE GCCE 2021, pp. 926–929, Oct. 2021. DOI: 10.1109/GCCE53005.2021.9622046
[3] P. Caballero-Gil, C. Caballero-Gil, and J. Molina-Gil, “Ubiquitous collision avoidance system for red light running,” PE-WASUN ’18, Proceedings of the 15th ACM International Symposium on Performance Evaluation of Wireless Ad Hoc, Sensor, & Ubiquitous Networks, pp. 76–83, Oct. 2018. DOI: 10.1145/3243046.3243054
[4] J. Bauer and N. Aschenbruck, “Towards a low-cost RSSI-based crop monitoring,” ACM Trans. Internet Things, vol. 1, no. 4, Article No. 21, June 2020. DOI: 10.1145/3393667
[5] J. Redmon, “Darknet: open source neural networks in C,” https://pjreddie.com/darknet/, 2013–2016.
1 Introduction

As the concepts of smart cities are practical applications with the Internet of Things (IoT) devices and regional smart city services should process local information from neighbor IoT devices. In traditional systems, the cloud server processes local information via the Internet, which puts an unnecessary processing load on the cloud server. In order to process local information efficiently, we need a distributed information processing system [1]. Additionally, many developers will provide many different types of services shortly. We encourage the importance of an edge framework and cloud services using general-purpose devices [2]. Specifically, we suggest implementing a new service deployment environment to support independent service installation for the edge processing system.

The authors have proposed a new flexible service deployment environment for the edge processing system [2], and this letter extends the proposed system. We try to give a solution to the conventional systems that focused on special services and implemented special devices [3, 4].

Our environment provides sharing mechanism of IoT devices with container technology. The proposed system consists of an edge framework and a cloud service. The edge framework supports container architecture to realize flexible service deployments, and the cloud service manages user account and service containers for edge devices. The prototype implementation provides multiple independent service processing on a single edge device. As a prototype, we have implemented a surveillance camera service and measured the execution performance required to provide the service.

2 Proposed system

Figure 1 shows the proposed system. The proposed system consists of the edge framework and the cloud service. The edge framework is the processing environment for several services provided by developers. The cloud service has some management functions for account management, service certification, service container distribution to edge frameworks. The edge framework also implements some sensors to collect neighbor information, the special sensing container changes real sensor values to abstracted sensor information and provides sensor information to service containers. As a result, each service container can access sensor information.
based on abstracted sensor categories such as air temperature, humidity, illuminance, etc.

The proposed system is an open architecture for providing multiple services on edge devices. We assume that multiple contributors, such as service developers, device owners, and cloud administrators, will build the distributed processing system.

The proposed system assumes that owners of the edge devices select some edge services on the cloud service. Each edge framework downloads some selected service containers from the cloud service to provide the service to users in its neighborhood. The proposed system realizes the functions as one main board with multiple sensors instead of installing multiple edge devices with different sensors.

Service developers are the first contributor to develop an application service for edge devices. They create the application services as a service container for easy distribution to the edge devices. Since the edge devices have the sensing container to provide sensor information, the service container can access it to process its service. The cloud service manages the list of the required sensor information for each service container. The available service container can be filtered by the required sensor information and deployed sensor list on the edge devices.

The owners of edge devices are 2nd contributors to this system because they install the real edge devices in service areas. They buy edge devices and deploy them to provide service in the service areas. Since they can select available services on the cloud for the edge devices, they can easily install the service into their own edge devices. The selected service works for user terminals around the edge devices.

The proposed system assumes that users use their own devices, called user devices, such as smartphones and tablets. Since the edge devices provide Wi-Fi access point service, they can access the services on the edge devices via the Wi-Fi connection.

The cloud administrator is the administrator of the proposed system who maintains the authentication of users, certification of services, distribution of services. Therefore, the proposed system provides a user account to developers and owners to access the cloud service. Since it also has a certification process of services before the distribution, it provides safety of the distributed services.

2.1 Container configuration

Edge devices prepare the virtual environment to ensure the services’ independence and realize the easy replacement of the service. As the virtual environment, the proposed system employs container-type implementation that realizes easy creation, distribution, and removal of the service environment. Since each edge device has some sensors, the sensing container handles these sensors and converts the sensor values to abstracted sensor information. They deploy essential functions as three containers: user terminal communication, management, and sensing. The service containers are also installed according to the selected service list. Details are shown below.

- User terminal communication container

  The user terminal communication container handles users’ requests. The edge device prepares a Wi-Fi access point for user terminals to provide services.
Therefore, it provides the service lists to the user terminal connected to the device. When a user selects a service, it requests to start the service to the management container to start the service container. When the service container is available, it provides access information to the user.

• Management container

The management container manages the edge device’s service container and computational resources. When it receives the service requests from the user terminal container to start the designated service, it evaluates the required resource for the requested service based on the remaining resource and starts the service for the user. It also notifies the access information to the user terminal communication container after the service’s startup. Another function of the container is monitoring the status of each service container. When users do not access a service container or a service container consumes much resource, the management container stop the service container not to reduce the computational resources.

• Sensing container

The sensing container manages all sensors on the edge device. Since several sensor devices are used as the same object of sensors, the sensing container requires special procedures for each sensor device. For example, SENSIRION’s SHT31 and BOSCH’s BME280 have the temperature sensor function. However, the access method to the temperature sensor value is totally different based on the sensor devices. Since the edge device may use different sensor devices for the same sensor information, the sensing container exchanges the sensor values to the abstracted sensor information such as temperature. As a result, the service containers can easily access the sensor information by the abstracted sensor name.

• Service container

The service containers are additional containers on the edge device because the service containers are downloaded from the cloud according to the selected service list. Each service container has a designated application service. The application service can access sensor information by accessing the sensing container with abstracted sensor names. After the service’s startup, the service container provides its own service to users.

3 Implementation

As the proposed service framework demonstration, we have implemented a prototype system. The prototype system supports a service container for surveillance camera service. The prototype uses multiple containers to display the images from the onboard camera on the user terminal. The prototype consists of user equipment, a cloud service, and an edge device equipped with sensors. Figure 2 shows the evaluation model of the prototype system.
We used the Raspberry Pi 4 as the hardware equivalent of the cloud and the edge device, and the iPod touch as the user terminal. The edge device was equipped with a Logicool Webcam C270 camera as a sensor. We used Docker as a container virtual technology and MJPG-Streamer to distribute the streaming video from the camera via a web page.

As the preparation, the edge device deploys the terminal communication container in advance and obtains the service container from the cloud. The terminal communication container receives the service request from the user terminal and requests the service container to the cloud. The service container uses the values of the sensors on the edge device to perform service processing.

The prototype system assumes that users access the service through their smartphones. Therefore, the prototype edge device provides a Wi-Fi connection service for user equipment. We also developed an iOS application for the prototype service.

When users start the application, the currently available service name is displayed as a button on the screen. Then users can select the service name, and the application sends a service request to the edge device. As a result of the service, the camera’s image installed in the edge device is displayed on the application.

### 4 Performance evaluation

Figure 3 shows the sample image displayed on the screen of the prototype smartphone application. The image was taken by the edge device and delivered to the smartphone application. As the verification process, we have measured the period to prepare

![Fig. 2. Prototype evaluation model.](image-url)
an available service state. The period starts when the user terminal sends a service request and ends when the user terminal receives the service information. We have taken ten measurements for the following two states and calculated the average. Figure 2 shows the measurement result of the processing time. The first is the service execution state that means the service container is running. The second is the executable service state that means the service container does not exist, but the Docker image is retained. In addition, we have measured the internal processing time in the executable service state ten times. The internal processing time was measured by dividing the phase into two parts: the service preparation time and service execution processing time. The service preparation time is extracting the Docker image and starting the service container. The service execution processing time is making service requests to the running service container and receiving the results. We found that the executable service state took 2,352 ms longer than the service execution state, and the most time-consuming part was the deployment of the Docker image. Therefore, our system strategy should not delete service containers frequently.

We also verified the resource status of edge devices when operating multiple service containers. The edge device equips bme280 as a temperature sensor. We have added a sensing container that acquires the value of the temperature sensor and implemented two types of service containers. When two service containers that receive and display sensor values from the sensing container are operated simultaneously, the average CPU usage rate was 1.45%, and the memory usage rate was 0.75%. When two service containers that detect image objects [5] were operated simultaneously, the average CPU usage rate was 98.8%, and the memory usage rate was 39.45%. As a result, it worked for services with different resource loads.

5 Conclusion

This paper has proposed and implemented a new service deployment environment to support independent service installation for the edge processing system. As a prototype, we have implemented a surveillance camera service and measured the execution performance required to provide the service. We also showed that it is possible to provide multiple services simultaneously.
Evaluation for cloud-based load distribution model in ICSN

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Abstract: We have proposed Information-Centric Networking (ICN) based wireless Sensor Networks (ICSN). ICSN Platform (ICSNP) is an environment to provide multiple Internet of Things (IoT) services using a single sensor network. Excessive requests will cause load congestion at the sink node as the number of concurrently running services increases. To avoid sink node congestion, we propose a cloud-based load distribution model in this paper. As a result, the experimental results show that the load is distributed at the sink node with a reduction of about 45% compared to ICSNP.

Keywords: IoT, ICN, cloud, load distribution
Classification: Network System

References

[1] Ministry of Internal Affairs and Communications, Information and communications white paper 2021 (in Japanese), https://www.soumu.go.jp/johotsusintokei/whitepaper/ja/r03/pdf/n0100000_hc.pdf, July 2021 (accessed Dec. 26th, 2021).
[2] B. Ahlgren, C. Dannewitz, C. Imbrenda, D. Kutscher, and B. Ohlman, “A survey of information-centric networking,” IEEE Commun. Mag., vol. 50, no. 7, pp. 26–36, July 2012. DOI: 10.1109/mcom.2012.6231276
[3] K. Kimura and O. Mizuno, “Performance requirements evaluation of network functions in information-centric networking based wireless sensor network,” IEEJ Trans. Information and Systems (in Japanese), vol. 140, no. 6, pp. 583–584, June 2020. DOI: 10.1541/ieeiejss.140.583
[4] E. Nagaoka, M. Yoshii, R. Banno, and O. Mizuno, “Cloud-based load distribution model in information-centric networking wireless sensor networks,” Proceedings of the IEICE International Conference on Emerging Technologies for Communications (ICETC) 2021, C2-1, 2021. DOI: 10.34385/proc.68.C2-1
[5] Cefore, https://cefore.net/ (accessed Dec. 11th, 2021).
1 Introduction

The IoT is spreading worldwide; in 2023, it is expected that about 34.1 billion IoT devices will be connected to the Internet [1]. Many IoT services collect data from sensor devices automatically and send it to the cloud via gateways. A sensor network is often built by an IoT service. As a result, when new services are introduced, construction and operating costs increase. We, on the other hand, use Information-Centric Networking (ICN) technology [2] to build sensor networks. It is called ICN-based wireless Sensor Networks (ICSN). We also build ICSN Platform (ICSNP) [3] to realize an environment where multiple IoT services exist in one sensor network.

However, as the number of simultaneously running services increases, load congestion at the sink node increases. As a result, a load distribution mechanism is required. We propose a cloud-based load distribution model to solve the problem. We supplemented the previously published paper [4] with new cloud and gateway evaluations in this paper. Based on these findings, the benefits and potential drawbacks of the proposed model are discussed.

2 ICSN and ICSNP

ICSN is a network that applies ICN technology to sensor networks. Sensor devices are strategically placed throughout the facility. As the cluster head (CH), one sensor device from each area is chosen. The sensor data is forwarded by the selected CH. If the sensor data is the same as the data from the own area, the data is cached.

ICSNP is a platform that realizes user request processing and service execution environment in a sink node. The sink node has an application and a database. For example, suppose a user requests the temperature data of Area B at 18:00 on December 25. The sink node that receives a user request converts it into “B/1225/1800/temperature” as Content ID in Interest. The sensor data is registered in the database after it is received, and the result is displayed to the user in the application. This allows for operation in an environment with multiple IoT services in a single sensor network. However, as the number of concurrent services increases, the number of requests increases rapidly, and the load on the sink node may become congested. As a result, there is a risk of decreased availability, such as service disruptions. It is necessary to develop a mechanism to address these issues.

3 Cloud-based distribution model

To address the issue of load congestion at the sink node, we propose a cloud-based load distribution model. The conventional model of ICSNP is shown in Fig. 1 (a), and the proposed model is Fig. 1 (b). The sink node in the traditional model handles all application execution and database functions. The proposal model, on the other hand, introduces a new gateway and a cloud to distribute the functions possessed by the sink node. In terms of each node’s role, the cloud runs the application, the gateway manages the data, and the sink node collects it. As a result, the load on the sink node can be distributed and the service’s availability can be guaranteed.

Also, the conventional model executes applications on each sink node, making it difficult to manage the operation of the services collection. On the other hand,
the cloud can manage the plural services in the proposed model. Furthermore, the database can be configured in the gateway so that it can be shared and used by other services. As a result, there is no need to create a database for services, which reduces the increase in load.

Fig. 1. Conventional model and proposal model

4 Evaluation

To evaluate the effectiveness of the proposed model, we evaluate node load against sensor data acquisition interval. Table I shows the experimental parameters.

Table I. Experimental parameters

| Parameters               | Values                                              |
|--------------------------|-----------------------------------------------------|
| Number of Nodes          | Sensor : 6, Sink : 1, Gateway : 1, Cloud : 1        |
| Memory Capacity          | Sensor : 1[GB], Sink : 4[GB], Gateway : 8[GB], Cloud : 10[GB] |
| Number of Areas          | 2                                                   |
| Measurement Time         | 300 [sec]                                          |
| Sensor Data Acquisition Interval | 60, 30, 10 [sec/req]                      |
| Number of Applications   | 1, 2, 4, 8                                          |
| Number of Trials         | 10                                                  |

This experiment uses two host terminals and implements them as virtual machines on Virtual Box. The first host terminal started a virtual terminal assuming a cloud, a gateway, a sink node, and sensor nodes. The second host terminal launched a virtual terminal that presupposed multiple users. To create the ICN environment for the proposed model, we use the ICN soft router Cefore-0.8.1 [5]. It runs on the host terminal in a virtual terminal. Also, because we assume the cloud application
is a web application, we used XAMPP to implement it. Furthermore, the gateway makes use of MariaDB for database construction. The comparison method is the conventional method, ICSNP alone.

Figures 2 (a)–(c) shows results of CPU usage rate on each node; the sink node,
the gateway, and the cloud, respectively.

Figure 2 (a) shows that the CPU usage rate on the sink node increases with the number of applications in the conventional method. Also, as the interval between sensor data acquisition shortens, the CPU usage rate increases. On the other hand, the proposed method suppresses the CPU usage rate, regardless of the number of applications or the shortening of the sensor data acquisition interval. Focusing on the number of applications \( (N = 8) \) and the acquisition interval \( (T = 10 \text{ [sec/req]}) \), the CPU usage rate of the proposed method is about 14%, whereas that of conventional method is about 59%. That is, the proposed method achieves about 45% reduction compared to the conventional method. The web application and database are used in the traditional method to receive request information and sensor data. As a result, the load is incurred. In the proposed method, however, those two functions are distributed to the cloud and the gateway, allowing the load to be reduced.

Figure 2 (b) shows that the CPU usage rate on the gateway increases slightly with the number of applications and the shortening of the acquisition interval. Focusing on the number of applications \( (N = 8) \) and the acquisition interval \( (T = 10 \text{ [sec/req]}) \), CPU usage rate is about 42%. Figure 2 (c) shows the CPU usage rate on the cloud increases slightly as well, reaching about 30% when the number of applications \( (N = 8) \) and the acquisition interval \( (T = 10 \text{ [sec/req]}) \). The increases load on the gateway and the cloud can be less than that of the conventional sink node.

These results show that the proposed method can distribute the load at the sink node more effectively than the conventional method and ensure the availability of the IoT service. In this experiment, the cloud does not have both the application execution and the database to manage the sensor data. The gateway has the database to reduce load caused by adding and referencing process sensor data. When applications are run in the cloud and sensor data acquisition processes, such as referencing sensor data acquisition history, occur in a short period of time, the cloud’s load increases. As a result, the proposed model, in which the cloud executes the applications and the gateway accesses the database, reduces the load on the cloud. The proposed method is assumed to apply to Edge Computing. In Edge Computing, the gateway is set up between the cloud and the devices. A gateway is used to distribute the processes in the cloud. This allows for cloud load distribution, timely sensor data acquisition, and real-time process execution.

Meanwhile, the proposal requires a new gateway and cloud, which requires extra implementation costs. Therefore, it is a trade-off that increases deployment cost but eliminates the load congestion at a particular node.

5 Conclusion

This paper proposes the cloud-based load distribution model to solve the load congestion at the sink node. To evaluate the effectiveness of the proposed method, we implemented it on a virtual terminal on a host terminal and conducted experiments. As a result, the sink node reduced CPU usage rate by approximately 45% when compared to the traditional method, and the load increase in the gateway and cloud was also reduced when compared to the conventional sink node. Furthermore, we demonstrate the relationship between the number of applications and CPU load. We
also show the relationship between the sensor data acquisition interval and CPU load. These findings suggest that the proposed method is more applicable in environments with multiple IoT services and can ensure the availability of IoT services.

In the future, we will implement and evaluate the proposed method on actual machines to show its effectiveness in a real environment.

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Buffering time optimization for path tracking accuracy in remote vehicle control with digital twin computing

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Abstract: On path tracking control of vehicles via the Internet, transmission delay and jitter prevent them from tracking a target path accurately. To improve the control accuracy, it is effective to apply digital twin computing and jitter buffer to the control system. The control system with jitter buffer has an issue of optimizing buffering time. Through simulations by using some realistic transmission delay models, we quantitatively evaluated the control accuracy depending on the buffering time. As a result, we showed that buffered packet rate ($BPR$) can be a key index to optimize the buffering time according to transmission delay.

Keywords: unmanned vehicle, remote control, transmission delay, jitter buffer, state-predictive control

Classification: Navigation, Guidance and Control Systems

References

[1] M. Tsuru, M. Takai, S. Kaneda, Agussalim, and R.A. Tsiory, “Towards practical store-carry-forward networking: Examples and issues,” IEICE Trans. Commun., vol. E100-B, no. 1, pp. 2–10, Jan. 2017. DOI: 10.1587/transcom.2016cq0001
[2] The Japan Times, “Japan to start testing unmanned vehicles on public roads,” https://www.japantimes.co.jp/news/2019/06/24/business/corporate-business/japan-start-testing-unmanned-vehicles-public-roads, June 2019 (accessed June 2019).
[3] C. Lozoya, P. Martí, M. Velasco, J.M. Fuertes, and E.X. Martín, “Simulation study of a remote wireless path tracking control with delay estimation for an autonomous guided vehicle,” The International Journal of Advanced Manufacturing Technology, vol. 52, no. 5, pp. 751–761, Feb. 2011. DOI: 10.1007/s00170-010-2736-x
[4] P. Isto, T. Heikkilä, A. Männelä, M. Uitto, T. Seppälä, and J.M. Ahola, “5G based machine remote operation development utilizing digital twin,” Open Engineering, vol. 10, no. 1, pp. 265–272, May 2020. DOI: 10.1515/eng-2020-0039
[5] L. Repelle, R. Muradore, D. Quaglia, and P. Fiorini, “Improving performance of networked control systems by using adaptive buffering,” IEEE Trans. Ind. Elec-
1 Introduction

With the spread of the Internet of Things, the demand for applications utilizing communication networks has increased. Autonomous driving of unmanned vehicles (UVs) is one of the technologies that have been attracting attention in recent years. The technology is expected to be applied not only to automobiles but also to various small vehicles, such as patrol vehicles [1] and goods delivery vehicles [2].

Remote control by cloud computing is effective to realize an application that autonomously controls a large number of UVs over a wide area. Centralizing control functions into a cloud server (CS) has advantages of system cost reduction, easy update of the functions and utilization of information collected through the network. We investigate the feasibility of a CS-based UV remote control system for path tracking control [3]. On the control via the Internet and wireless access, transmission delay and jitter prevent the UVs from tracking a target path accurately. Applying digital twin computing (DTC) [4] to the CS improves UV remote control accuracy even with transmission delay. By applying jitter buffer [5] to the UV, the control accuracy improves further.

The control system with DTC and jitter buffer has an issue of optimizing buffering time. The jitter buffer absorbs transmission delay fluctuation of control signal from the CS. It increases the average control delay between the CS and the UV. When a digital twin model (DTM) can not perfectly simulate a controlled UV, a prediction error by the DTM increases as the control delay increases. Therefore, too long buffering time deteriorates the UV control accuracy. On the other hand, too short buffering time also deteriorates it because the jitter buffer can not absorb the delay fluctuation sufficiently.

To achieve accurate UV control, it is necessary to optimize the buffering time according to transmission delay characteristics through communication network. In this letter, we proposed to use buffered packet rate (BPR) as a key index to optimize the buffering time according to transmission delay through simulations of UV remote control. This letter provides more detailed descriptions of the system and explanation of simulation results than our conference publication [6].

2 Simulator of remote path tracking control by cloud server

For path tracking control via a network, we assumed that a CS and a UV formed a feedback control loop. The UV travels following control signal from the CS, and cyclically sends its current traveling state as a status information packet. The CS has
target path information, and sends control signal referring to the status information to make the UV travel along the path. Transmission delays of control signal and status information are unavoidable when communicating via the Internet. As these delays increases, the accuracy of the path tracking control deteriorates and the UV deviates largely from the target path.

By applying DTC and jitter buffer, the control system can reduce the deterioration of UV control accuracy. To quantitatively evaluate the control accuracy, we created a UV remote control simulator. Figure 1 is an overview of our simulator. In the simulator, the UV is a program that simulates a small four-wheeled vehicle. To focus on evaluating the effect of transmission delay on UV control, we assumed that the UV can track the target path accurately when there is no transmission delay between the CS and the UV.

During the simulation, the UV receives control signal of UDP packet including values of traveling speed \((v)\), front wheel steering angle \((\theta)\) and sending time of the CS \((t_{cs})\). Received signals are temporarily kept in a jitter buffer. The jitter buffer has parameter \(D\) that represents the maximum buffering time. When a control signal is transmitted with a delay \(d\), the jitter buffer outputs the signal to the UV at time \(t_{cs} + \max(d, D)\). The UV simply calculates it own position \((x, y)\) on a virtual plane every 1 ms according to values of \(v\) and \(\theta\). When \(\theta \neq 0\), the position is calculated as follows.

\[
\begin{align*}
    r &= \frac{WB}{\sin \theta} \\
    \delta &= \frac{v}{r \times 1000} \\
    x_t &= x_{t-1} + r(1 - \cos \delta) \cos \phi_{t-1} + r \sin \delta \sin \phi_{t-1} \\
    y_t &= y_{t-1} + r \sin \delta \cos \phi_{t-1} - r(1 - \cos \delta) \sin \phi_{t-1} \\
    \phi_t &= \phi_{t-1} + \delta
\end{align*}
\]

Here, \(r\) and \(\delta\) are a turning radius and a rotation angle of a virtual vehicle, respectively. \(WB\) is a wheelbase of the virtual vehicle. \(v/1000\) indicates the distance traveled in 1 ms. The UV sends status information of UDP packet including its position and direction to the CS every 100 ms.

Before the remote control starts, the CS gets a target path data. To apply DTC to the control system, we implemented a DTM in the CS based on the UV program.
After sending a control signal, the CS uses the DTM to predict the UV traveling state that received the signal. Instead of status information from the UV, the CS cyclically refers to the prediction by the DTM to calculate a new control signal. Receiving the status information from the UV, the CS corrects the prediction based on the actual UV traveling state. The above procedures allows the CS to perform the state-predictive control of the UV.

3 Transmission delay models and target path in simulations

In each simulation, we used one of three transmission delay models described in Fig. 2(a) for packet transmission delay between the CS and the UV. These delay models reproduce the realistic transmission delays through the Internet and Wi-Fi. Ranges of delay fluctuations in delay model A, B and C were from 3 to 254 ms, 54 to 296 ms, and 120 to 363 ms, respectively. Each delay model consists of Internet and Wi-Fi delays. The three models have different characteristics of the Internet delay and the same characteristics of Wi-Fi delay. The Internet delays was created by fitting the Pareto distribution [7] to delay measurements with three cloud servers in Tokyo, San Francisco and Frankfurt in July 2021. The Wi-Fi delay was measurements in

![Graph showing transmission delay characteristics](image1)

![Graph showing target path](image2)

Fig. 2. Simulation conditions
IEEE 802.11ac with coexisting 300 Mbps background traffic.

The target path was as shown in Fig. 2(b). To evaluate UV control accuracy, we defined parameter \(MLD\) as the maximum lateral deviation between the target path and a UV trajectory in each simulation. A small \(MLD\) means that the UV could travel along the target path accurately.

4 Simulation result and discussion

Through simulations, we quantitatively evaluated the UV control accuracy depending on the \(D\) value. We carried out ten simulations for each condition. \(D\) was set between the minimum and maximum values of the reproduced transmission delay model. The traveling speed \(v\) was set to 2.0 m/s.

Figure 3(a) illustrates \(MLD\) characteristics for each delay model. Every point indicates the average of \(MLD\) for each condition. The vertical axis indicates \(MLD\) value. The horizontal axis indicates \(D\) value. For reference, we also carried out the UV remote control simulation without DTM and jitter buffer. In these simulations, the sending cycle of status information by the UV was 10 ms. The \(MLD\) values were much larger than those in Fig. 3(a). Specifically, at \(v = 2.0\) m/s, \(MLD = 0.16, 0.65\) and \(1.57\) m for delay model A, B and C, respectively. From those results, we confirmed effectiveness of DTM and jitter buffer to improve the accuracy of path tracking control.

Figure 3(a) shows that \(MLD\) decreases as \(D\) increases. As described in Section
1. $D$ should be small as long as $MLD$ does not increase greatly. To realize accurate UV control, it is effective to adaptively determine $D$ according to transmission delay characteristics. We considered that buffered packet rate ($BPR$) is an appropriate parameter to determine the optimal value of $D$. $BPR$ indicates a rate of control signal packets with the transmission delay of $D$ or less. Figure 3(b) is a graph in which the horizontal axis of Fig. 3(a) is replaced with $BPR$. This graph shows that the UV control accuracy did not deteriorate much for any transmission delay model even when $BPR$ decreased from 100% to 96%. Between 100% and 96% $BPR$, the variations of $MLD$ were 1.9, 1.7 and 1.0 cm for delay model A, B and C, respectively.

In this study, the DTM accurately simulates the UV and does not include any prediction error. For accurate UV control, $D$ value depends not only on transmission delay characteristics but also on modeling accuracy of the DTM. Our simulation results suggest that $BPR$ can be a key index to optimize the buffering time of control signal.

5 Conclusion

On CS-based path tracking control of UVs, DTC is effective to improve the UV control accuracy. By applying jitter buffer to absorb transmission delay fluctuation of control signal, the control accuracy further improves. To achieve the accurate control, buffering time of the jitter buffer should be adaptively optimized according to transmission delay.

In this letter, we quantitatively investigated how to optimize the buffering time by using some realistic transmission delay models based on actual measurements. The result showed that buffered packet rate ($BPR$) can be a key index to optimize the buffering time. Our simulations showed that maximum lateral deviation from a target path varied by only 1.9 cm at most between 100% and 96% $BPR$.

On UV control system with DTC, a prediction error caused by modeling accuracy of the DTM affects the control accuracy. In our future work, we will investigate the effectiveness of optimizing the buffering time using $BPR$ under various conditions of modeling accuracy.
A cache control mechanism in CCN for cyclic communication base IoT services

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Abstract: The content-centric network (CCN) is a reasonable solution for the application of the Internet-of-Things (IoT) as it enables the efficient transfer of data in IoT services through simplified sequences and reduced traffic via a networked cache. Generally, low latency IoT services are provided by cyclic communication. In this case, however, the latency of the data stored in the cache may result in serious situations. To solve this problem, this study proposes a new cache control mechanism with timed control. This paper also describes the utility of the proposed mechanism on high-reliability services with hot standby redundant systems in addition to low latency services.

Keywords: IoT, ICN, CCN, cache control, real-time service, redundant system

Classification: Network

References

[1] M. Amadeo, C. Campolo, J. Quevedo, D. Corujo, A. Molinaro, A. Iera, R.L. Aguiar, and A.V. Vasilakos, “Information-centric networking for the internet of things: challenges and opportunities,” IEEE Network, vol. 30, no. 2, pp. 92–100, 2016. DOI: 10.1109/mnet.2016.7437030

[2] B. Ahlgren, C. Dannewitz, C. Imbrenda, D. Kutscher, and B. Ohlman, “A survey of information-centric networking,” IEEE Commun. Mag., vol. 50, no. 7, pp. 26–36, 2012. DOI: 10.1109/mcom.2012.6231276

[3] S. Podlipnig and L. Bőszörményi, “A survey of web cache replacement strategies,” ACM Computing Surveys, vol. 35, no. 4, pp. 374–398, 2003. DOI: 10.1145/954339.954341

[4] P. Schulz, M. Matteo, H. Klessig, M. Simsek, G. Fettweis, J. Ansari, S. Ali Ashraf, B. Almeroth, J. Voigt, I. Riedel, A. Puschmann, A. Mitschele-Thiel, M. Muller, T. Elste, and M. Windisch, “Latency critical IoT applications in 5G: Perspective on the design of radio interface and network architecture,” IEEE Commun. Mag., vol. 55, no. 2, pp. 70–78, Feb. 2017. DOI: 10.1109/mcom.2017.1600435cm
1 Introduction

Despite the increased implementation of IoT services on the Internet in the recent years, there are key issues that remain unaddressed. Such concerns include the large protocol overheads in IP and upper layer protocols, and the subsequent increase in latency and traffic volume by these overheads and their processing. These problems should be mitigated especially in cases of transferring tiny small data blocks with high frequency as low-latency services.

For this purpose, information-centric network (ICN) technologies have been applied in several IoT services [1]. These technologies involve various mechanisms, and the content-centric network (CCN) is one of them [2]. The CCN provides simple communication sequences by utilizing paired messages of INTEREST and DATA. It also mitigates traffic volume through duplicated data transfer through the use of a networked cache.

However, on the networked cache in the CCN, conventional control mechanisms, such as the least recently used (LRU) and least frequency used (LFU), lead to an increase in latency in the transferred data [3]. This becomes a significant problem as some IoT services require low-latency data transfer [4]. Our previous studies have highlighted this dilemma, and have since provided a preliminary solution [5].

Building on the progress of our prior study, the present work proposes the timed cache control mechanism to solve the issue of increased latency in data transfers for low-latency IoT services. The current study, in a similar manner, recommends the application of the proposed mechanism to a redundant system for IoT high-reliability services as one of its use cases.

2 Traffic characteristics of IoT services

In IoT services, a number of tiny data blocks are transferred from end devices, e.g., sensors to servers across networks. To simplify communication sequences, these blocks are transferred cyclically with or without independent updating of data. Even
if data blocks are lost, retransmission of these blocks is not invoked. Instead, data are updated in the succeeding cycles. Such an architecture has been deployed ever since for industry plants [6] and factories [7].

The performance requirements for these low-latency services are summarized in [4]. It must be noted that the requirements for communication quality in these services are more stringent than those specified in ITU-T Y. 1541 [8] for legacy IP base services.

3 Data transfer by the CCN and its problem

The CCN provides simple communication sequences without a domain name system (DNS) for translation between IP addresses and data names on the Internet. It also provides a networked cache to reduce the transfer traffic volume by mitigating duplicate transfers. Therefore, the implementation of CCN serves to improve the performance of IoT services [1]. We have previously proposed the architecture and detailed operations of low-latency IoT services based on the CCN [9]. This architecture was referred to as the CCN with network initiative and traffic control (C-NAT). It provides triggers of data transferred by network facilities, i.e., interworking points (IWPs). However, in this architecture, it was assumed that the mechanism of the networked cache control is a legacy one (e.g., LRU).

It is a concern for low-latency IoT services that legacy cache control mechanisms maintain old data, as shown in Fig. 1. In CCN, end-devices transfer data by paired INTEREST. Several IWPs are then deployed as routers on the Internet. The IWPs accommodating end-devices send INTEREST according to the provisioning table including the timing of cyclic transfers. These operations have been described in a previous study [9]. As evidenced in Fig. 1, if Data #b and Data #c are lost, Server #1 recognizes that the current data is Data #a because the stored data in the cache of IWP #2 are transferred to Server #1. In addition, when Server #2 attempts to

![Fig. 1. Transfer operations in low-latency IoT services and their problems](image)
obtain the latest data, Data #a can be obtained as the latest data. In low-latency IoT services, this situation may be a factor in serious accidents.

4 Proposed mechanism on the cache control in IWPs

To solve the problem described in the previous section, the timed cache control mechanism is proposed to forcibly eliminate stored data. This section describes the operations of the proposed mechanism, and discusses the effectiveness of these protocols in various applications.

4.1 Operations of the proposed mechanism and use cases

The proposed mechanism for cache control is shown in Fig. 2(a). In this mechanism, an IWP keeps the received data in the cache for a fixed time interval. The stored data in the cache are then removed when this interval expires, or when the next set of data is received.

This mechanism can be applied to two use cases. One is in the sequential communication across multiple IWPs, as demonstrated in Fig. 2(b). In this mode of usage, the proposed mechanism contributes to the control and variation of latency. Another application is in the parallel communication across IPWs, as depicted in Fig. 2(c). This case is a typical configuration of the redundant system with hot standby [10], wherein the proposed mechanism promotes synchronization between both systems.

![Proposed mechanism and its applied use cases](image)

**Fig. 2.** Proposed mechanism and its applied use cases

4.2 Estimation of the fixed interval for the timed cache control

The fixed interval was tuned according to the interarrival period between consecutive DATA messages. The estimation utilized in this work was derived as follows.
Given the average interval, $a$, between consecutive data at a sensor, i.e., the transfer cycle, the arrival process of the interval at an ingress point of the IWP conforms to the exponential distribution of $a$. The probability density function (PDF) describing the arrival process, $f(x)$, is then described using Eq. (1):

$$f(x) = \frac{1}{a} e^{-\frac{1}{a}x}$$

(1)

If data loss is considered and its probability is denoted by $p$, then the arrival interval between data at an ingress point of an IWP conforms to the convolution of the geometric distribution. Meanwhile, an Erlang distribution is observed if the data loss conforms to the Bernoulli process. Therefore, its PDF, $h(x)$, is described using Eq. (2):

$$h(x) = \sum_{i=1}^{\infty} g_i * f_i(x)$$

(2)

$$g_i = p^{i-1}(1 - p)$$

(3)

$$f_i(x) = \frac{1}{a^i(i - 1)!} x^{i-1} e^{-\frac{x}{a}}$$

(4)

On the other hand, the cumulative distribution function (CDF) is obtained from the integration of Eq. (2) with respect to time, as shown in Eq. (5):

$$H(x) = \int_{0}^{x} h(t)dt$$

(5)

Results of numerical analyses according to Eq. (5) is shown in Fig. 3 below, where the characteristics of the fixed interval upon normalization with the transfer cycle $a$ is indicated. In Fig. 3, the data loss probability, $p$, is $10^{-6}$, which is the typical value in [4].

![Fig. 3. An example of the fixed interval by numerical analysis ($p = 10^{-6}$)](image)

Moreover, Fig. 3 depicts the relationship between the fixed interval normalized by $a$ and the corresponding probability. The curve obtained suggests that when the fixed interval was five times of $a$ or more, the next DATA was promptly received.
5 Conclusions

This study focused on the cache control in the CCN for low-latency IoT services. In line with this, a C-NAT architecture for IoT services was proposed. Moreover, a cache-control mechanism for C-NAT with its target applications was presented. The proposed mechanism eliminated data in the cache contained in an IWP, after the duration of a fixed interval. The incorporation of this time-controlled process into the C-NAT architecture is anticipated to provide low-latency IoT services on ICN-based infrastructures.

With the effectiveness of the protocol outlined in this study, future work will center on the deployment of the C-NAT architecture on various IoT systems, as well as the subsequent establishment of system implementation guidelines through international standardization bodies such as ISO/IEC JTC1/SC41 focusing on IoT and digital twin.
Turning path tracking control technology for fixed-wing UAV to realize video transmission relay station

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Abstract: Unmanned Aerial Vehicles (UAV) are currently used to provide many kinds of services. A video transmission relay station system using fixed-wing UAV that can fly for a long time are now investigated. In the system, the fixed-wing UAV needs to keep turning in the area accurately as a relay station. To realize that we propose a new turning path control system by both switching the roll angle commands to satisfy the balance of both forces and azimuth control to correct the turning radius deviation. The effectiveness was confirmed by computer simulations assuming an actual flight aircraft.

Keywords: fixed-wing UAV, turning, control, relay station

Classification: Navigation, Guidance and Control Systems

References

[1] K. Hasegawa, “Future of agriculture using unmanned aerial vehicle,” IEEEJ, vol. 45, no. 4, pp. 504–507, 2016.
[2] A. Brezouescu, T. Espinoza, P. Castillo, and R. Lozano, “Adaptive trajectory following for a fixed-wing UAV in presence of crosswind,” Journal of Intelligent and Robotic Systems, vol. 69, pp. 257–271, 2013. DOI: 10.1007/s10846-012-9756-8
[3] J.M. Levin, M. Nahon, and A.A. Paranjape, “Aggressive turn-around manoeuvres with an agile fixed-wing UAV,” IFAC-PapersOnLine, vol. 49, no. 17, pp. 242–247, 2016. DOI: 10.1016/j.ifacol.2016.09.042
[4] B.N. Pamadi, Performance, Stability, Dynamics and Control of Airplanes, 3rd edition, ed. J.L. Azevedo, AIAA, Virginia, pp. 131–133, 2015. DOI: 10.2514/4.102745

1 Introduction

Unmanned Aerial Vehicles (UAV) are currently used to provide many kinds of services, such as monitoring crops and forests, inspecting tunnels, bridges, and buildings, measuring locations, transporting commercial goods, and relaying video
and data [1]. Some services are actually being provided, while others are experimentally tried and under research and development. In response to the needs of high image quality and long-distance transmission by UAV, frequencies in the 169 MHz, 2.4 GHz, and 5.7 GHz bands have been legally established and are expected to be applied to various services.

The above services are often carried out by a multi-copter type of UAV and by using direct radio waves. The number of mountain accidents has increased in recent years, and debris flows and landslides caused by record heavy rains happen frequently. In cases of providing search services for victims in mountainous areas or observing a disaster area, radio waves are often blocked by complex terrain and it is difficult for the UAV to receive transmission images from the UAV for photography or maneuvering commands from the ground station. In addition, victims and disaster areas should be observed from a distance several kilometers away, with a wide area well over 1 km square. Generally, a multi-copter type UAV can fly for only 20 minutes. Therefore, it is not appropriate for the UAV to observe distant and vast areas.

Taking these factors into consideration, we propose a video transmission system by using a fixed-wing UAV, which flies at high altitude for a long time, as a relay station (Fig. 1). In the system, the UAV should continue to accurately turn along a predetermined circle path, with its radius specified because the UAV cannot hover above the designated point. There are many papers concerning turning technology [2, 3], but few concerning accurately turning along a specified circle path as well as flying for a long time, especially using a gasoline model airplane. Therefore, this paper proposes a new method of accurately tracking a turning path and describes its design methodology and validity via computer simulation.

Fig. 1. Video transmission relay system image
2 Issues and design policy

In order to realize this accurate path turning flight, we focus on 1) the balance of vertical and horizontal forces and 2) an azimuth control system that provides feedback to correct the turning radius as needed. Generally, the lift of the airplane is generated based on airspeed, which includes wind disturbances; its vertical components balance with vertical acceleration generated based on ground speed. Therefore, keeping vertical balance leads to generating horizontally unbalanced force, which causes radius deviation.

Therefore, we propose a new method to both change the roll angle command corresponding to a deviation of altitude and control the azimuth direction corresponding to a deviation of turning radius.

3 Design method

3.1 Derivation of force balance condition equation

From the upper right of Fig. 1, the balance of force can be expressed by Eqs. (1) and (2) [4]. Equation (1) is derived from the vertical balance equation, and Eq. (2) is from the horizontal balance equation. Here, \( L \) is lift, \( \phi \) is roll angle, \( m \) is aircraft mass, \( g \) is gravitational acceleration, \( V_c \) is ground speed, and \( R \) is turning radius.

\[
\begin{align*}
L \cos \phi_1 &= mg \\
L \sin \phi_2 &= \frac{mV_c^2}{R}
\end{align*}
\]

The lift is given by

\[
L = \frac{1}{2} \rho V_L^2 S C_L
\]

where \( \rho \) is air density, \( V_L \) is air speed, \( S \) is main wing area and \( C_L \) is lift coefficient. These equations can be expressed as Eqs. (4) and (5) by focusing on the roll angle.

\[
\begin{align*}
\phi_1 &= \cos^{-1}\left(\frac{2mg}{\rho S C_L V_L^2}\right) \\
\phi_2 &= \sin^{-1}\left(\frac{2mV_c^2}{\rho S C_L R V_L^2}\right)
\end{align*}
\]

The roll angle command from Eqs. (4) and (5) are switching by the flight conditions. This study focuses on turning flight, so it mainly follows roll angle commands that satisfy horizontal balance.

3.2 Relationship between turning radius and azimuth

It is not possible to return from a deviated path only by balance of force. Therefore, we focused on the turning radius and azimuth. Assuming the turning radius deviation is \( \Delta R \), expressing the UAV’s lateral speed \( \Delta R \) is obtained by using the azimuth deviation \( \Delta \psi \) from the directed azimuth angle in Eq. (6).

\[
\Delta R = V_L \sin \Delta \psi
\]

Assuming that \( \Delta \psi \) is not large, Eq. (6) becomes
The target turning radius error is corrected in Eq. (7) by controlling the azimuth. In addition, since the azimuth obtained in Eq. (7) is the value on the aircraft axis, it is necessary to convert Eq. (8) to the azimuth by the tangential axis on the target path using the turning center \((x_0, y_0)\) and the current position \((x, y)\) as shown in Fig. 2. Here, \(\psi_R\) indicates the azimuth angle on the tangential axis, and \(\psi_{\text{Body}}\) indicates the azimuth angle on the airframe axis.

\[
\Delta R = V_L \Delta \psi 
\]

\[\psi_R = \psi_{\text{Body}} - \left(90^\circ - \tan^{-1}\left(\frac{x - x_0}{y - y_0}\right)\right)\]  

Fig. 2. Conversion from airframe axis to tangential axis

4 Simulation

4.1 Control system design

The control system to track turning a path with a radius \(R\) accurately consists of roll angle control, azimuth control, height control, and speed control. All of these controls use PID elements. When the altitude deviation exceeds 10% of the target value it switches to the roll angle command that satisfies the vertical balance.

4.2 Control target

The performance of the control system was confirmed by computer simulation. The simulation, a fuel driven model airplane with a total weight of about 5 kg and a wingspan of 2 m is assumed. The purpose of this simulation was to confirm a turning path tracking deviation of the control system. In addition, the target performance of turning radius deviation is set to be \(\pm 6\) m at maximum. This is determined from the performance of the inertial navigation system used in the actual flight test.

4.3 Simulation conditions

The conditions of the simulation were 22 m/s for the initial groundspeed \(V_c\) and airspeed \(V_L\), and 0 degrees for each attitude angle. The target turning radius is 100 m and the target altitude is 100 m. The flight conditions used for the simulation included no wind, a wind speed of 3 m/s, and a wind speed of 5 m/s. It is assumed the wind is parallel to the x-y plane and is blowing from the negative direction of the x axis to the positive direction. Furthermore, assuming an actual flight experiment, a simulation time is set for a three-lap turn flight. The roll angle command switches from a horizontal command to a command that satisfies the vertical direction when the target altitude exceeds 10% deviation from the designated altitude.
4.4 Simulation results

Figure 3(a) shows the simulation trajectory of this method. The x and y axis shows the positions, and the z axis shows the height. When the wind was not blowing, it was confirmed that the UAV could fly with a maximum turning radius deviation of 2 m, as shown by the blue color in Fig. 3(b). In contrast, it was confirmed that when the wind speed was up to 5 m/s, the UAV could fly with a maximum turning radius deviation of 3 m, as shown in yellow. Also, as shown in Fig. 3(a), the altitude deviation does not exceed 10 m, which is 10% of the target altitude deviation; thus, all three flights follow the roll angle command that satisfies the lateral balance.

5 Conclusion

In this study we proposed a turning path tracking control system and confirmed its control performance by computer simulation for the UAV. The simulation confirmed that the flight was possible with a turning radius deviation of 2 m with no wind. Even with wind blowing at 5 m/s it was possible to have a turning radius deviation of 3 m. In the future, we will proceed with flight verification experiments using the target aircraft.

Acknowledgments

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Coordinate rectification of indoor neural network localization using filters

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Abstract: Today, we can know our outdoor location through global positioning system (GPS). However, it is not easy to estimate our indoor location because satellite signals are difficult to reach. Therefore, we study indoor localization using wireless local area network (LAN). In this work, we attempted to rectify our position estimated using a convolutional neural network (CNN). CNN of indoor localization is based on fingerprinting, and it can only estimate pre-measured coordinates. Simultaneously, CNN is not considered relation by time series. Thus, we suggest using filters like the Kalman filter or particle filter. We can estimate the pre-measured coordinates and inter-coordinates among them using filters. Additionally, we can improve the position estimation accuracy based on the temporal dependency of the user assuming a pedestrian. Our experimental validation shows that the proposed method improves the accuracy. We evaluated and enhanced the position estimation accuracy using CNN with the particle filter. Consequently, we obtained that the mean, median, and max errors decreased 0.27, 0.24, and 4.72 m, respectively, compared with only CNN.

Keywords: indoor localization, fingerprinting, deep learning, particle filter, CNN

Classification: Navigation, Guidance and Control Systems

References

[1] S. Aikawa, S. Yamamoto, and M. Morimoto, “WLAN Finger Print Localization Using Deep Learning,” IEEE APCAP Aug. 2018. DOI: 10.1109/APCAP.2018.8538306

[2] S. Aikawa, S. Yamamoto, and T. Muramatsu, “CNN Localization Using AP Inverse Position Estimation,” 2019 IEEE Conference Antenna Measurements and Applications(CAMA2019), OCTOBER23-25, 2019. DOI: 10.1109/CAMA47423.2019.8959663

[3] L. Xiao, A. Behboodi, and R. Mathar, “A Deep Learning Approach to Fingerprinting Indoor Localization Solutions,” 2017 27th International Telecommunication Networks and Applications Conference (ITNAC), 2017. DOI: 10.1109/ATNAC2017.8215428

[4] A. Mittal, S. Tiku, and S. Pasricha, “Adapting Convolutional Neural Networks for Indoor Localization with Smart Mobile Devices,” GLSVLSI’18, Chicago,
1 Introduction

Recently, we have been able to know our location with high accuracy outdoors through GPS positioning. However, the precision is low indoors and underground due to the difficulty in receiving satellite signals. Therefore, various techniques for location estimation have been studied for indoor localization. One of them is the estimation approach using wireless LANs. This approach accesses the received signal strength indicator (RSSI) of several wireless LANs. There are various methods for wireless LAN positioning, and the fingerprinting method has the highest accuracy.

Mean squared error (MSE) is the simplest mechanism to infer location using the fingerprinting method. It is easy to implement; however, its precision is not so high. In contrast, deep learning can be used to estimate more precisely than MSE [1]. In particular, a convolutional neural network (CNN), a kind of deep learning, is significantly accurate in location estimation [2]. It is the most exact model for rooms with elementary structures.

2 Related work

2.1 Fingerprint

In the fingerprinting method, the operator obtains the access point (AP) information in advance. The AP information is mainly the media access control (MAC) address and RSSI for each AP. When the operator measures AP information, they prepare an area map and set measured coordinates on the map. They also set several times measured. They periodically measure RSSI at these preset coordinates the preset number times of measurements [3]. This AP information is collected by their Android devices and stored as a database (DB) on PC. They make the neural network (NN) model and this model learns the AP information based on DB. The users’ device calculates location at any of the pre-measured coordinates using this model with received RSSI. Users can know their location results from the estimated coordinate in the display.

2.2 CNN indoor Localization

NNs are efficient in machine learning. They can solve a wide range of problems, e.g., image recognition and language processing. Even in localization, NNs are efficient and have been widely studied by many people. Various NN models, e.g.,
standard deep NN (DNN), CNN, ResNet, and other NN, are researched for indoor localization using wireless LAN. CNN is especially useful for such localization [4]. The model is often used in image recognition. In CNN, only certain units between adjacent layers have special layers with coupling. In these layers, it performs basic image processing operations: convolution and pooling. CNN is a two-dimensional (2D) input, and these layers downsample to 1D output.

In image recognition, the values of each pixel are input to this input. Meanwhile, the RSSI values of each AP are input to the input in the localization. Thus, the CNN method requires a decision input label (a mapping of each AP to its input position). The label is determined by estimating each coordinate RSSI based on the AP information that the operator collected. The operator decides its resolution using the map as the input to CNN. In this way, the input label of CNN is determined. Such a method is called inverse position estimation of APs.

CNN outputs pseudo probabilities at each coordinate. In conventional methods, the estimated user position is the position with the highest probability in these probabilities.

2.3 Filters
A state space model is used for time series analysis and rectification. It consists of two variables: state and observed values. In this model, the state \( x_t \) and the observed \( z_t \) value at time \( t \) are assumed according to Eqs. (1) and (2), as follows:

\[
\begin{align*}
x_t &= f_t(x_{t-1}, u_t) \\
z_t &= g_t(x_t, v_t)
\end{align*}
\]

where \( f_t \) and \( g_t \) are any functions; \( u_t \) is the state noise; \( v_t \) is the system noise. \( x_t \) is generated from \( x_{t-1} \) by \( f_t \), but includes noise \( u_t \). Similarly, \( z_t \) is generated from \( x_t \) by \( g_t \), but includes noise \( v_t \). Eq. (1) is called the equation of state, and Eq. (2) is called the observation equation. Filters are what is correct from \( z_t \) to \( x_t \) under these assumptions. There are various filters caused by the difference between its variables’ features.

Kalman filter assumed that \( f_t \) and \( g_t \) are linear, and noises are Gaussian. Additionally, particle filter assumed that \( f_t \) and \( g_t \) are nonlinear, and noises are non-Gaussian. In this work, we used these typical filters [5].

3 Proposed method

3.1 CNN with Kalman Filter
In this study, we have assumed an equation of state based on an equation of motion about Kalman filter. In other words, the equation of state of Kalman filter is expressed in Eq. (3) using user’s position \( x_t \) and speed \( \dot{x}_t \) at time \( t \).

\[
\begin{bmatrix} x_t \\ \dot{x}_t \end{bmatrix} = \begin{bmatrix} 1 & \Delta t \\ 0 & 1 \end{bmatrix} \begin{bmatrix} x_{t-1} \\ \dot{x}_{t-1} \end{bmatrix} + \begin{bmatrix} \Delta t^2/2 \\ \Delta t \end{bmatrix} [a_t], \quad a_t \sim N(0, \sigma^2_t)
\]

where \( \Delta t \) is the one period time getting AP information, and \( a_t \) is the Guassian noise following variance \( \sigma_a^2 \). Meanwhile, we defined the observation equation by Eq. (4)
as follows:

\[ z_t = x_t + v_t, \quad v_t \sim N(0, \sigma_v^2) \]  

(4)

\( v_t \) is a Gaussian noise following variance \( \sigma_v \). We developed a system that corrects from \( z_t \) as position estimated using CNN to \( x_t \) as user position.

### 3.2 CNN with Particle Filter

It is known that MSE, which is a conventional method with particle filter, can improve the localization accuracy [6]. In this study, we constructed a particle filter adapted for CNN based on [7].

In the particle filter of indoor localization, particles are first initialized on a map. The number of particles is set previously, and these particles are let to spread uniformly. Then, particles have equal weight, which only executes when the application is opened. After initializing particles, the application begins to receive AP information constantly. The application estimates on CNN every time it receives.

The particle filter carries out three steps: predication distribution, calculating likelihood, resampling, and estimation of modified position after every CNN estimation, as shown in Fig. 1. 1\textsuperscript{st} step: prediction distribution is moving the position of particles randomly. 2\textsuperscript{nd} step: calculating likelihood is computing particles’ likelihood as particles’ weight. Particles’ likelihood means the probability of which user exists at particle position. This probability is based on the CNN estimator and calculated using Eq. (5).

\[
\omega_l = \sum_i \frac{p_i}{\sqrt{2\pi\sigma_v^2}} \exp \left\{ -\frac{(c_i - x_l)^2}{2\sigma_v^2} \right\} 
\] 

(5)

where \( \omega_l \) and \( x_l \) are the \( l \)-th particle’s weight (likelihood) and position; \( p_i \) and \( c_i \) are the \( i \)-th coordinate’ probability (CNN output) and position. In the conventional method [7], we assumed particles’ weight based on a normal distribution. In contrast, we assumed them based on the Gaussian mixture model in this study. All four parameters are at time \( t \). Eq. (5) assumes a mixed normal distribution based on the probability outputted from CNN. Finally, 3\textsuperscript{rd} step is resampling. Particles’ weights are assumed to be equal, which is 1/\( l \); however, they are split or extincted following their weights instead. Repeating these three steps, particles density can approximate the probability distribution of user’s existence.

![Fig. 1. Algorism of the particle filter on indoor localization.](image)
Particle filter aims to express probability distribution, but localization needs to decide user location to one point. Therefore, user location $x_{PF}$ is determined by weighted average as expressed in Eq. (6).

$$x_{PF} = \sum l \omega_l \times x_l$$ (6)

This $x_{PF}$ also outputs as corrected user position every CNN estimating.

Comparing Kalman filter, particle filter can calculate complex distribution. On the other hand, particle filter need more computational complexity than Kalman Filter.

4 Experiment validation

4.1 Simulation conditions

We have collected data for eight days under the conditions shown in Table I. Measurements were taken while stationary for three days as an operator and while moving for five days as a user. The operator data were used as teaching data, and the user data were used to validate precision. The CNN model and parameter of the particle filter were decided by operator data. The CNN models were made of two kinds, whose fingerprinting interval is 2 or 5 m. The user data were in chronological order.

| Table I. Measurement conditions |
|--------------------------------|
| Place | University of Hyogo of engineering, 6th floor, Building B, corridors |
| Date | 8/17/2021—8/19/2021 (an operator) |
| Measuring conditions | Stationary for 3 days (an operator) |
| Coordinates interval | 2 or 5 m (an operator) |
| Num. of measurement | 50 times/coordinates (8/17, an operator) |
| | 20 times/coordinates (8/18,19, an operator) |
| | 800 coordinates/day (a user) |

4.2 Parameter

The operator data are divided into days. The model learned from data of the first and second day. In particular, weights in the CNN model were mainly determined data of the first day (August 17). The data of the second day (August 18) were used to protect overfitting. The data of the third day (August 19) were used to evaluate model performance by the operator. Additionally, the $\alpha$ parameter of the particle filter was decided from the data of the third day.

4.3 Result

We confirmed that we could estimate points other than coordinates. Figure 2 shows the error distribution of each method and each interval.

We can confirm that CNN with particle filter with an interval of 2 m is the best method. Compared with only CNN (2 m), the mean, median, and max errors decreased 0.27, 0.24, 4.72 m, respectively. In addition, compared with particle filter and Kalman filter, each precision of particle filter is lower than Kalman filter.
Especially, the mean, median, and max errors of using particle filter (2m) are 0.13, 0.11, 0.65 m lower than these of using Kalman filter (2m).

5 Conclusion
In this paper, we used filters for validation in fingerprinting indoor localization. A new architecture using CNN with particle filter could interpolate coordinates. Moreover, this architecture delivered lower localization mean error values of about 29-32 cm than only CNN. Not only that, it did lower localization mean error values of about 12-13 cm than CNN with Kalman filter. Therefore, the particle filter could coordinate interpolation and high-precision estimate with the number of particles within the range that does not affect the UI of Android applications.

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Operation and management signal detection using quadrant photodiode and auxiliary management communication channel for simple and stable free-space optical communication systems

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Abstract: Free-space optics (FSO) is key technology for next-generation communication systems and is used for communications with mobile nodes. In an FSO system, both a client signal and a network operation and management (OAM) channel are required to maintain the system. Acquisition of a new radio wave does not occur during transference over an optical signal; however, misalignment tolerances for the laser beam and receiver aperture decrease and the loss of the OAM signal leads to the network control is lost for a while. Although a photodetector with a larger aperture than that of the client signal improves the tolerance, more devices are required. Therefore, we propose receiving the OAM signal with the quadrant photodiode (QPD), which is generally mounted onto FSO equipment for position detection of laser beams. We determined that the allowable displacement expanded to about three times the usual amount without requiring more optical components.

Keywords: free-space optics, FSO, OAM, AMCC, quadrant photodiode
Classification: Wireless Communication Technologies

References
[1] M.A. Khalighi and M. Uysal, “Survey on free space optical communication: A communication theory perspective,” IEEE Communication Surveys & Tutorials, Vol. 16, No. 4, pp. 2231–2258, Fourth Quarter 2014. DOI: 10.1109/comst.2014.2329501
[2] H. Kaushal and G. Kaddoum, “Optical communication in space: Challenges and mitigation techniques,” IEEE Communication Surveys & Tutorials, Vol. 19, No. 1, pp. 57–96, First Quarter 2017. DOI: 10.1109/comst.2016.2603518
1 Introduction

Free-space optical communication, or “free-space optics” (FSO), uses a laser beam to transmit signals and is key technology for next-generation mobile networks [1, 2]. The FSO provides a wide bandwidth with mobile networks by using a license-free spectrum. Both the client signal and the operation and management (OAM) signal maintain and manage an FSO system. However, OAM signal transmission through a radio wave can pick up interference from other communication links and require acquisition of a new radio license. Therefore, the optical channel is preferable for transferring the OAM signal.

One of the main technical issues in the FSO system is a tracking failure. A tracking failure is more likely to occur than radio waves because the laser beam is narrow. The FSO equipment does not receive OAM signals and loses the network control when a large deviation occurs between the axis of the laser beam and the aperture of the optical receiver due to the movement and vibration of the FSO equipment. By using photodetectors with a larger aperture, the tolerance to the higher deviation increases. Meanwhile, as the aperture size of the photodetector increases, the response frequency tends to be lower. Therefore, enlarging the photodetector’s aperture limits the data rate of the client signal. Transmitting the OAM signal on a different wavelength from that for the client signal and adding a receiver with a large diameter only for the reception of relatively low data-rate OAM signals could increase the tolerance to the movement and vibration of the FSO equipment without restricting the data rate of the client signal. However, this approach makes the receiver configuration complex, which is not desirable considering the size and weight requirements for FSO equipment.

Therefore, we propose and experimentally determine the feasibility of a scheme in which the OAM signal is transmitted on an auxiliary management communication channel (AMCC) [3] and received with a quadrant photodiode (QPD). AMCC is a low-speed communication channel superimposed over the lower frequency region on a wavelength of the client signal that does not require any additional optical components, unlike wavelength-division multiplexing. The QPD is essentially used in the FSO system to detect the misalignment of the laser beam from the center of its aperture [4]. In general, the response frequency of the QPD is slow due to its large aperture; however, its response frequency is high enough to receive relatively low data-rate OAM signals. Moreover, its large aperture enhances the tolerance to the movement and vibration of the FSO equipment, which enhances the stability of the FSO system.
2 System model and experimental setup

Figure 1 shows a schematic and experimental setup of our proposed system. The transmitter superimposes the OAM signal as the AMCC to the client signal. At the receiver, the output of a large diameter QPD comprised of four photodiodes is combined and demodulated into a low-speed OAM signal. Because the bandwidth of the QPD is much lower than that of the client signal, the client signal is cut off, and only the OAM signal is output.

On the transmitter side, the 1.55-μm laser diode (LD) is directly modulated with the client signal output from the pulse pattern generator (PPG). The client signal is a non-return to zero (NRZ) signal with a bit rate of 10.3125 Gbit/s and a bit pattern of PRBS $2^{31}-1$. The 128-kbit/s OAM signal with the condition of NRZ, on-off keying (OOK), and PRBS $2^{7}-1$ is superimposed onto the client signal as a 500-kHz AMCC by externally modulating the LD output with a small modulation index using a lithium-niobate (LN) modulator. After amplification to +10 dBm using erbium-doped fiber amplifiers, the signals are emitted into the FSO section of 3 m. The radius of the laser beam is 6 mm.

On the receiver side, the input light beam is divided into two with a half mirror. Half of the light beam is collimated into a single-mode fiber (SMF) through a lens and is detected with the avalanche photodiode (APD). The core size of the SMF is 10 μm. Then, the bit error rate (BER) of the client signal is evaluated with the error detector. To measure BER characteristics, the input to the APD is varied by the variable optical attenuator (VOA). Meanwhile, the other half of the beam enters the QPD (Thorlabs PDQ30C) to receive the OAM signal. The aperture diameter, the bandwidth, and the maximum output voltage of the QPD are 3 mm, 150 kHz, and 10 V, respectively. The output from the QPD is captured with a digital oscilloscope for offline processing. During the offline processing, the OAM signal is demodulated,
and the Q-value of its eye pattern is calculated for BER estimation.

We mounted the receiver onto a stage that moves in the direction orthogonal to the laser beam to examine how the misalignment tolerance was enhanced using the QPD and AMCC for the OAM signal. We evaluated the reception characteristics of the OAM signal while changing the position of the receiver. For comparison, we also evaluated the reception characteristics of the OAM signal, which was extracted from the output of the APD using the LPF.

3 Experimental results

First, we evaluated the reception characteristic of the client signal to examine the effect of the AMCC superimposition shown in Fig. 2 (a). As a baseline, the values plotted with closed circles are the BER when the output of the LN modulator was connected to the receiver without the FSO section, only through an SMF, and the AMCC was not superimposed. The values plotted with open circles and triangles are the BER in cases with the FSO section. The BER degradations from adding the FSO section and superimposing the AMCC were negligible. Figure 2 (b) shows the waveform when the AMCC was superimposed. When the symbol of the OAM signal is mark, the amplitude fluctuated a little during a 500-kHz cycle for 7.8 μs corresponding to one bit length of a 128-kHz OAM signal.

Figure 3 (a) shows the tolerance to the misalignment for the OAM signal. The BER values were plotted while the receiver’s position changed orthogonally to the laser beam. Rather than by our method, the number of optical components was maintained by obtaining the OAM signal from the output of the APD. In the method, the BER of the OAM signal (□) rapidly deteriorated from 2 mm, and the maximum displacement with the BER below $10^{-3}$ was 3.7 mm. However, with our method, the BER of the OAM signal demodulated from the QPD output (■) remained less
Fig. 3. Reception characteristics of OAM signal. (a) Relationship between displacement and BER and (b) QPD output.

than $10^{-12}$ until an 8-mm displacement, and the maximum displacement with a BER below $10^{-3}$ is 10.7 mm. This value is about three times larger than the value when using the APD.

Figure 3 (b) shows an example of the waveform of the AC component output from the QPD. The 128 kbit/s OOK NRZ signal appears clear. The bit train of the AMCC signal “1101101...” can be observed.

4 Conclusion

We proposed a FSO system that achieves stable detection of OAM signals with a simple configuration. The QPD installed in the FSO equipment receives low-rate AMCC signals and does not require an additional receiver. We achieved Superimposing AMCC on the client signal without degrading BER in FSO system and higher displacement tolerance with a QPD that had a larger aperture than the APD. The tolerance was tripled from 3.7 mm to 10.7 mm.
Performance evaluation of bidding price for Spot Instances on Amazon EC2

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Abstract: Amazon EC2 offers surplus instances, called Spot Instances, to consumers at a discounted price. Due to the nature of surplus resources, Spot prices change dynamically according to long-term demand trends. Consumers bid for Spot Instances and are allocated an instance if their bid exceeds the Spot price. On the other hand, if the bidding price falls below the Spot price, the instance is deallocated. Therefore, the setting of the bidding price significantly affects the availability of Spot Instances. In this study, by employing 10 types of bidding prices for Spot Instances and evaluating the usage time, we demonstrate that the availability of Spot Instances depends on the bidding price.

Keywords: Amazon EC2, Spot Instance, Spot price, bidding price

Classification: Network

References

[1] G. Isobe, D. Katayama, S. Aketani, and T. Koita, “Classification of price fluctuations for spot instances on Amazon EC2,” IEICE Commun. Express, vol. 10, no. 12, pp. 991–996, 2021. DOI: 10.1587/comex.2021xb0155
[2] S. Karunakaran and R.P. Sundarraj, “Bidding strategies for spot instances in cloud computing markets,” IEEE Internet Comput., vol. 19, no. 3, pp. 32–40, 2015. DOI: 10.1109/mic.2014.87
[3] V. Khandelwal, A.K. Chaturvedi, and C.P. Gupta, “Bidding strategies for Amazon EC2 spot instances—a comprehensive review,” Proceedings of the International Conference on Computing Communication Control and Automation, pp. 1–5, 2018. DOI: 10.1109/iccubea.2018.8697462
[4] M.B. Chhetri, M. Lumpe, Q.B. Vo, and R. Kowalczyk, “To bid or not to bid in streamlined EC2 spot markets,” Proceedings of the IEEE International Conference on Services Computing, pp. 129–136, 2018. DOI: 10.1109/scc.2018.00024

1 Introduction

Amazon Elastic Compute Cloud (Amazon EC2), a service of Amazon Web Services (AWS), provides surplus resources as Spot Instances. The price of Spot Instances (Spot price) fluctuates according to long-term trends in demand. Consumers can set the maximum price they are willing to pay (bidding price) for a Spot Instance.
Spot Instances are allocated when the bidding price is above the Spot price. If the bidding price falls below the Spot price, the instance is deallocated. In this study, we focused on the bidding price setting when using Spot Instances.

Isobe et al. previously showed that Spot prices can be classified based on the characteristics of their price fluctuations [1]. However, bidding prices have not been indicated for classified Spot Instances. Karunakaran and Sundarraj proposed four bidding strategies: bidding at a price close to the Reserved Instance price, bidding at a price higher than the average Spot price (estimated from historical Spot prices), bidding at a price close to the On-Demand price, and bidding at a price higher than the On-Demand price [2]. On the other hand, Khandelwal et al. proposed seven bidding strategies: minimum Spot price, average Spot price, maximum Spot price, higher price than the current Spot price, 70% of On-Demand price, On-Demand price, and above On-Demand price [3]. They showed that there is no bidding strategy that fits all consumers. This means that, instead of a single bidding strategy, a bidding strategy should be tailored to the characteristics of the instance’s price variation. Currently, the maximum bidding price has been changed and bids cannot exceed the On-Demand price. Therefore, a new investigation is inevitable. Additionally, in 2017, Amazon EC2 launched a new Spot pricing model. Chhetri et al.’s study, using econometrics to classify Instances, confirmed that Amazon EC2’s new Spot pricing model removed extreme spikes and seasonal factors [4].

In this study, we employ 10 different bidding strategies based on On-Demand prices for Spot Instances and show the change in availability that the bidding price setting gives to the Spot Instances.

2 Evaluation

We evaluated 922 current generation instances provided by Amazon EC2 in ap-northeast-1 (Tokyo), for which price history was available. We collected Spot price histories for 90 days from October 22, 2021 to January 19, 2022. Following the previous research [4], Spot price histories have no seasonal factors. For the Spot price histories, we can obtain Spot prices and timestamps when Amazon EC2 changes the Spot prices. Timestamps have irregular time intervals, so we converted them to hourly Spot prices to even the time units.

2.1 Spot price normalization

Instances have various On-Demand prices and Spot prices are set on the basis of the discount rates for On-Demand prices. Data using only Spot prices are not accurate because Spot prices depend on the On-Demand prices. We normalized the Spot prices by On-Demand prices to unify their scales. If the Spot price has the same value as the On-Demand price, the normalized Spot price is 1. We calculated the normalized Spot prices by the following equation:

\[
\text{normalized Spot price} = \frac{\text{Spot price} [\$]}{\text{On-Demand price} [\$]}
\]

Furthermore, the normalized Spot price values represent more than 0.1 since AWS states that Spot Instances are available up to a 90% discount compared to On-Demand prices.
2.2 Bidding price
We set 10 different bidding prices with each set based on the On-Demand price. We calculated the bidding prices by the following equation:

\[
\text{Bidding price} = \text{On-Demand price} \times x,
\]

where \( x = 0.1, 0.2, \ldots, 1.0 \).

Spot Instances are discounted by up to 90% from the On-Demand price. Therefore, \( x \) has a minimum value of 0.1. In addition, \( x \) is set to a maximum of 1.0 since the bidding price cannot be set above the On-Demand price. The bidding price is expressed by the value of \( x \), omitting the On-Demand price. For example, if \( x \) is 0.5, the bidding price is 0.5.

2.3 Usage time and Interruption
We executed a bidding simulation with 10 different bidding prices. When the bidding price falls below the Spot price, the use of the Spot Instance is interrupted. We define the time when the bidding price is above the Spot price as usage time and the time when the bidding price is below the Spot price as interruption time. The number of interruptions is defined as \# of interruptions. We used 90 days of price history, so the maximum usage time is 2,160 hours.

3 Evaluation results
Examples of the evaluation results are shown in Fig. 1. Figure 1(a) shows the results of a simulation on the c6gd.4xlarge instance with a bidding price of 0.5. The usage time was 1,692 hours and the interruption time was 468 hours. The \# of interruptions, the number of times the bidding price was below the Spot price, was 2. Table I shows the evaluation results for all bidding prices. Red-colored cells indicate more instances and blue-colored cells mean fewer instances. Comparing the bidding prices of 0.2 and 0.3, the number of instances with zero interruptions increased from...
### Table I. Evaluation results for all bidding prices

| Bidding price | # of interruptions |  |  |  |  | [6,73] |
|---------------|--------------------|---|---|---|---|-------|
| 0.1           | 0                  | 911 | 0 | 0 | 0 | 0 | 11 |
| 0.2           | 23                 | 870 | 9 | 2 | 0 | 0 | 18 |
| 0.3           | 439                | 381 | 47 | 16 | 7 | 7 | 25 |
| 0.4           | 740                | 125 | 25 | 3 | 1 | 1 | 27 |
| 0.5           | 799                | 73 | 20 | 3 | 0 | 1 | 26 |
| 0.6           | 815                | 70 | 14 | 0 | 0 | 0 | 23 |
| 0.7           | 824                | 67 | 7 | 0 | 1 | 0 | 23 |
| 0.8           | 832                | 64 | 4 | 0 | 0 | 0 | 22 |
| 0.9           | 835                | 61 | 4 | 0 | 0 | 0 | 22 |
| 1             | 901                | 1 | 0 | 0 | 0 | 0 | 20 |

### Table II. Detailed results by bidding price

#### (a) Result of bidding price = 0.2

| Bidding price = 0.2 | # of interruptions |  |  |  |  | [6,73] |
|---------------------|--------------------|---|---|---|---|-------|
| number              | 23                 | 870 | 9 | 2 | 0 | 0 | 18 |
| average usage time[h] | 2,160             | 32 | 1,304 | 797 | N/A | N/A | 484 |
| max. usage time[h]  | 2,160              | 2,057 | 2,013 | 1,080 | N/A | N/A | 1,778 |
| min. usage time[h]  | 2,160              | 0 | 338 | 514 | N/A | N/A | 101 |
| standard deviation  | 0                  | 229 | 532 | 283 | N/A | N/A | 362 |

#### (b) Result of bidding price = 0.3

| Bidding price = 0.3 | # of interruptions |  |  |  |  | [6,73] |
|---------------------|--------------------|---|---|---|---|-------|
| number              | 439                | 381 | 47 | 16 | 7 | 7 | 25 |
| average usage time[h] | 2,160             | 384 | 1,294 | 1,193 | 1,149 | 1,243 | 1,402 |
| max. usage time[h]  | 2,160              | 2,154 | 2,147 | 2,132 | 1,945 | 2,102 | 2,063 |
| min. usage time[h]  | 2,160              | 0 | 46 | 214 | 1,945 | 709 | 359 |
| standard deviation  | 0                  | 718 | 567 | 598 | 453 | 417 | 512 |

#### (c) Result of bidding price = 0.4

| Bidding price = 0.4 | # of interruptions |  |  |  |  | [6,73] |
|---------------------|--------------------|---|---|---|---|-------|
| number              | 740                | 125 | 25 | 3 | 1 | 1 | 27 |
| average usage time[h] | 2,160             | 1,098 | 1,286 | 878 | 753 | 2,070 | 1,719 |
| max. usage time[h]  | 2,160              | 2,154 | 2,110 | 1,775 | 753 | 2,070 | 2,063 |
| min. usage time[h]  | 2,160              | 0 | 33 | 381 | 753 | 2,070 | 1,056 |
| standard deviation  | 0                  | 866 | 493 | 636 | 0 | 0 | 223 |
Similarly, comparing the bidding prices of 0.3 and 0.4, the number of instances with zero interruptions increased from 439 to 740. However, there was no significant change in the number of instances when comparing the bidding prices of 0.4 and 0.5. We therefore opine that a bidding price of 0.4 makes a significant change in the availability of Spot Instances.

Table II shows the detailed results for each bidding price. Table II(a) shows the results for a bidding price of 0.2. Tables II(b) and II(c) show the results for bidding prices of 0.3 and 0.4, respectively. For Table II(a), when the # of interruptions is 1, the average usage time is 32 hours. Similarly, for Tables II(b) and II(c), the average usage times are 384 and 1098 hours, respectively, when the # of interruptions is 1. As the bidding price increases, the average time increases. However, each standard deviation is large. In other words, there is a large variation in the usage time.

4 Conclusion

In this study, we conducted bidding simulations with 10 different bidding prices for Spot Instances in Tokyo. The results show that a bidding price of 0.4 has a significant impact on the availability of Spot Instances. With a bidding price of 0.4, 80% of all instances were available without interruption. We also found that there is a large variation in the usage time even when only one interruption occurs. In the future, we will investigate the optimal bidding price for Spot Instances classified by the characteristics of price fluctuation.
An approach for suppressing waveform distortion of SOAs by controlling the temperature on in-line optical amplifier systems

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Abstract: Semiconductor Optical Amplifier (SOA) has attracted attention as a key component in supporting large capacity transmission systems. However, SOAs cause waveform distortion depending on the carrier lifetime, which ranges from picoseconds to nanoseconds, and this distortion degrades the signal quality. Thus, we are investigating suppressing the distortion by controlling gain characters and the carrier time by changing the temperature of SOAs on in-line optical amplifier systems. This paper reports that the distortion is suppressed by setting the temperature of the amplifier lower within the settable range, as a result of evaluating the temperature dependence of the distortion in optical amplification with SOAs theoretically.

Keywords: SOA, wave distortion, in-line optical amplifier system

Classification: Transmission Systems and Transmission Equipment for Communications

References

[1] J. Renaudier, “100nm ultra-wideband optical fiber transmission systems using semiconductor optical amplifiers,” Proc. 2018 European Conference on Optical Communication, Rome, Italy, pp. 1–2, 2018. DOI: 10.1109/ecoc.2018.8535354
[2] A. Napoli, N. Costa, J.K. Fischer, J. Pedro, S. Abrate, N. Calabretta, W. Forsyiaik, E. Pincemin, J.P.F.-P. Gimenez, C. Matrakidis, G. Roelkens, and V. Curri, “Towards multiband optical systems,” Proc. 2018 Photonic Networks and Devices, Zurich, Switzerland, NeTu3E, 2018. DOI: 10.1364/networks.2018.netu3e.1
[3] J. Renaudier, A.C. Meseguer, A. Ghazisaeidi, P. Tran, R.R. Muller, R. Brenot, A. Verdierr, F. Blache, K. Mekhazni, B. Duval, H. Debregeas, M. Achouche, A. Boutin, F. Morin, L. Letteron, N. Fontaine, Y. Frignac, and G. Charlet, “First 100-nm continuous-band WDM transmission system with 115Tb/s transport over 100km using novel ultra-wideband semiconductor optical amplifiers,” Proc. 2017 European Conference on Optical Communication, Gothenburg, Sweden, pp. 1–3, 2017. DOI: 10.1109/ecoc.2017.8346084
1 Introduction

Recently, transport and inter/intra datacenter networks require large capacity communication systems for broadband applications such as video streaming services. Semiconductor Optical Amplifiers (SOAs) have attracted attention as key components in supporting large capacity transmission [1, 2, 3]. SOAs are small-form semiconductor elements that can be integrated in other devices such as CFP/CFP2 and coherent receivers, so they are expected to be deployed as boosters and preamplifiers in inter/intra-datacenter networks. Additionally, SOAs are expected to be used
as in-line amplifiers on ultra-wideband transmission such as multiband transmission on behalf of Erbium-Doped Fiber Amplifiers (EDFAs). Because EDFAs are mainly used in only single-band transmission [2], and on the other hand, ultra-wideband SOAs for in-line amplification are feasible now [1, 3]. Ultra-wideband transmission with in-line optical amplifier systems will be required in future inter-datacenter networks and metro networks [4, 5].

However, SOAs can cause waveform distortion depending on the carrier lifetime, and this distortion often degrades the signal quality. Approaches for suppressing waveform distortion have been intensively investigated [6, 7]. Since the previous studies require light sources and high-speed control circuits, the configuration of an optical amplifier is complex. This can lead to an increased failure rate. To address this issue, we investigate approaches for suppressing waveform distortion by controlling the gain character and the carrier time by changing the temperature of SOAs. The proposed approach does not require any light sources or high-speed control circuit.

In this paper, we theoretically evaluate the temperature dependence of the waveform distortion in optical amplifications with SOAs at in-line optical amplifier systems. We demonstrate that the waveform distortion can be suppressed at the level of error-free transmission.

2 Suppressing waveform distortion by controlling temperature

The reason for SOAs causing distortion in optical amplification is that the carrier lifetime of SOAs is in the picosecond to nanosecond order. When the distortion occurs, we refer to the difference in the power in the amplified signal as “overshoot” and time to settle down from the rising part of amplified signal as “time constant” in Figure 1. The overshoot amount is determined by input-signal power, input-signal pulse width, amplification gain, and amplification time constant in shown [8]. The time characteristic of gain is shown like Eq. (1) by unsaturated gain $G_0$ and stationary gain $G_{CW}$ [7]. Time constant $\tau$ [s] is shown like Eq. (2) by the carrier lifetime $\tau_R$ [s] and transition probability coefficient $W'$.

$$G(t) = G_{CW} + (G_0 - G_{CW}) \exp\left(-\frac{t}{\tau}\right).$$  \hspace{1cm} (1)

$$\frac{1}{\tau} = \frac{1}{\tau_R} + W'. \hspace{1cm} (2)$$

If the overshoot amount exceeds the dynamic range of a receiver, not only is the signal disabled but also the receiver may be destroyed. Therefore, the overshoot must be suppressed within the dynamic range.

According to Eq. (1) and (2), the overshoot is likely to be suppressed by making the difference from $G_0$ and $G_{CW}$ smaller and making the slope of the amplified-signal waveform gentler by lengthening $\tau$. One approach is to control these parameters by optimizing the semiconductor materials and structure design inside an amplifier [9]. However, we focus on controlling the temperature of an amplifier because we can use standard amplifiers. Therefore, we suppress the overshoot by controlling $G_0$, $G_{CW}$, $\tau$ by changing the temperature.

The theory behind suppressing the overshoot by changing the temperature of SOAs is explained below. Note that the model uses material of InP-InGaAsP for
the 1.55 μm band [10]. The parameters used in the following equations are listed in Table I. According to Connelly [10], a gain coefficient $g$ and an absorption coefficient $\alpha$, which are amplification characteristics, are shown in Eq. (3) and Eq. (4). Here, $A$ is constant. According to Nilsson [11], $f_c$ and $f_v$ is the parameter depending on temperature. Therefore, they show that $g$ and $\alpha$ change depending on changing the temperature.

$$g = A(f_c - f_v).$$  \hspace{1cm} (3)

$$\alpha = A(1 - f_c)f_v.$$  \hspace{1cm} (4)

$$A = \frac{e^2}{4\sqrt{2}\pi^2 n_1 \tau R_0 v^2} \left( \frac{2m_e m_{hh}}{\hbar (m_e + m_{hh})} \right)^{1/2} \times \sqrt{\frac{v - E_g}{\hbar}}.$$  \hspace{1cm} (5)

According to Connelly [10], the traveling photon rate of the signal and Amplified Spontaneous Emission (ASE) depends on $g$ and $\alpha$. Also, gain is shown in Eq. (6).

$$G = (\Gamma g - \alpha)L.$$  \hspace{1cm} (6)

In Eq. (6), $G_0$ is gain when input-signal power is too small, and $G_{CW}$ is gain when continuous-signal power is input. In this paper, $G_0$ is gain when input-signal power is $-40$ dBm, and $G_{CW}$ is gain when input-signal power is $-16.5$ dBm.

Moreover, the rate equation of carrier density $n$ depends on the traveling photon rate of the signal and ASE [11]. Lastly, the carrier lifetime $\tau_R$ is shown in Eq. (7). This shows that carrier lifetime depends on the carrier density.

$$\frac{1}{\tau_R} = A_{rad} + B_{rad}n.$$  \hspace{1cm} (7)

Therefore, $G_0$, $G_{CW}$, and $\tau$ can be changed by changing the temperature. The overshoot amount by changing the temperature of SOAs is analyzed using Eq. (1)–(7). The time dependency of overshoot amount $D(t)$ is shown in Eq. (8).

$$D(t) = 10\log \frac{G(t)}{G(t_b)} = 10\log \frac{G_{cw} + (G_0 - G_{CW}) \exp(-t \left( \frac{1}{\tau_R} + w_0 G_0 P_{in}(t) \right))}{G_{cw} + (G_0 - G_{CW}) \exp(-t_b \left( \frac{1}{\tau_R} + w_0 G_0 P_{in}(t) \right))}.$$  \hspace{1cm} (8)

Figure 1 illustrates the evaluation configuration that resembles multiband transmission. We evaluate a single-pulse signal instead of WDM signals, as our objective is to clarify the effect of suppressing the overshoot. The pulse signal that is $-5$ dBm of the peak power and 100 ps of the pulse width $t_b$ is transmitted from the pulse laser (P-LD). Also, the rising time to $-5$ dBm of the pulse signal waveform is set to 10 ps supposing the modulated characteristic in a transmitter.

First, SOA recovers the node loss of 11.5 dB as the booster amplifier. Next, SOA recovers the span loss of 11.5 dB as an in-line amplifier or preamplifier. This is repeated up to six times. SOA uses the Automatic Current Control (ACC) mode.

It is supposed that the node loss of 11.5 dB consists of a C+L Arrayed Waveguide Grating (AWG) loss of 7.5 dB for multiplexing C-band and L-band signals and a $2 \times 1$ coupler loss of 4 dB for multiplexing S-band signals, and the span loss of 11.5 dB consists of fiber loss of 11.5 dB (0.2875 dB/km).

Table I lists the values of the parameters used for evaluation.
3 Evaluation results

Figure 2 shows evaluation results of the overshoot depending on the temperature in each the number of times for amplification. The number of times for amplification is shown as D1, D2, . . . , D6. The output power for the time dependency is shown for when the temperature T is set as 270 K and 300 K in Fig. 2. This shows signal waveforms after amplification.

The amount of overshoot becomes larger as the number of amplifications grows and is smallest at 270 K. The higher the temperature is above 270K, the larger the overshoot amount. This is because the higher the temperature, the saturated gain shrinks gently while the stationary gain shrinks linearly by Eq. (1)–(7). Also, the lower the temperature is below 270K, the larger the overshoot amount. This is because the carrier lifetime becomes long and the slope of amplification-signal waveform becomes gentle as the temperature lowers.

Masumoto et al. [12] showed that the overshoot amount is 7.82 dB in case of 12 times amplification on optical transmission while overshoot occurs, and error-free transmission can be achieved with more than 5 dB of Q margin. Therefore, when the overshoot amount is less than 7.82 dB, error-free transmission can be achieved.

Table I. Parameters of calculation

| Name of parameter | Parameter | Value | Unit |
|-------------------|-----------|-------|------|
| Speed of light    | c         | 3.00 x 10^8 | m/s  |
| Active region refractive InGaAsP index | n_s | 3.22 | - |
| Carrier lifetime of 300K | 1/τRO | 1.13 x 10^9 | s^{-1} |
| Optical frequency | ν         | 1.95 x 10^{-15} | s^{-1} |
| Conduction band electron heavy hole effective masses | m_n | 4.10 x 10^{-32} | kg |
| Valence band electron heavy hole effective masses, | m_p | 4.19 x 10^{-31} | kg |
| Bandgap energy    | E_g       | 1.23 x 10^{-19} | J |
| Planck constant   | h         | 6.626 x 10^{-34} | Js |
| Dirac constant    | h         | 1.055 x 10^{-34} | Js |
| Boltzmann’s factor | k | 1.381 x 10^{-23} | J/K |
| Injected current  | I         | 300 | mA |
| Active region length | L | 7.00 x 10^{-4} | m |
| Active region thickness | d | 4.00 x 10^{-7} | m |
| Central active region width | W | 4.00 x 10^{-7} | m |
| Optical confinement factor | λ | 0.45 | |
| Linear radiative recombination coefficient | A_{rad} | 1.00 x 10^{7} | s^{-1} |
| Bimolecular radiative recombination coefficient | B_{rad} | 5.60 x 10^{-16} | m^3 s^{-1} |
with several Q margin in this paper. This is referred to as the “Overshoot limit” in Fig. 2. At 300 K, which is considered as ordinary room temperature, the amount of overshoot is 6.1 dB at D2 and 8.76 dB at D3. These results show that error-free transmission can be achieved to the 2nd measure point in Fig. 1 and a single span because the overshoot limit is 7.82 dB. However, when we lower the temperature to 270 K, the amount of overshoot is suppressed. It is 7.19 dB at D4 and 9.41 dB at D5. This shows that error-free transmission can be achieved to the 4th measure point in Fig. 1 and three spans. Here, a single span is set to 40 km. Therefore, transmission distance can trebled from 40 to 120 km by lowering the temperature from 300 K to 270 K.

We also found out that lower temperature improved the noise figure in SOAs and improved the transmission quality. This leads to longer transmission distance. In addition, the active volume of the semiconductor, which is \( L \times d \times W \) in Table I, is a parameter that affects the overshoot, and this changes \( \tau_R \). When the active volume is increased with reference to [13, 14], \( \tau_R \) increases gradually and the overshoot is further suppressed at temperatures lower than 270 K. However, the temperature at which the overshoot is most suppressed is 270 K, the same as the result using Table I. Therefore, setting the temperature lower than room temperature can suppress the overshoot independent of the SOA type.

![Fig. 2. Evaluation results for overshoot while varying the temperature](image)

**4 Conclusion**

In this paper, we theoretically evaluated the temperature dependence of the waveform distortion in optical amplifications with SOAs. Results demonstrated that waveform...
distortion can be suppressed and the transmission distance can be made three times longer by adequately setting the operating temperature lower than room temperature regardless of the SOA type.