A study of LTE uplink resource allocation with long delay environment for satellite-airplane communication

Yoshimi Fujii¹ᵃ, Koichi Seki¹, Mariko Sekiguchi¹, and Hiroyuki Tsuji²

¹ Kozo Keikaku Engineering,
4–38–13 Honcho, Nakano-ku, Tokyo 164–0012, Japan
² National Institute of Information and Communications Technology,
4–2–1 Nukui-Kitamachi, Koganei, Tokyo 184–8795, Japan
a) yfujii@kke.co.jp

Abstract: Unlike terrestrial radio communication, airborne Internet access services are still based on satellite communication systems, which use single carrier FDMA/TDMA access interfaces. To maximize the resource utilization and acquire maximum benefit of the services, it would be natural to apply OFDMA/SCFDMA based terrestrial systems like LTE for the satellite-airborne communication systems. However, these systems were originally designed for short delay conditions in microseconds and have difficulties under the very long delay between satellite and airplanes. In this paper, we propose a simple and effective way of resource allocation for LTE uplink SCFDMA which would be applicable to airborne-satellite communication systems.

Keywords: LTE, long delay, resource scheduling, uplink, SCFDMA

Classification: Satellite Communications

References

1 Introduction

Recently, airborne Internet access services, provided by both domestic and international commercial carriers, are becoming more popular. Most of them are provided with satellite-based systems. Historically, satellite communication systems have selected relatively simple physical layer systems such as single carrier modulation and TDMA, apart from OFDMA/SCFDMA which is commonly used for modern terrestrial wireless communication systems like LTE. To support the increasing number of airplanes sharing radio resources at a time, TDMA has a potential disadvantage against LTE in terms of inflexibility and packet delay, caused by linear and sequential time slot allocations. Despite that, there are some reasons not to choose LTE. For instance, some of them originated from physical layer characteristics [1], such as from the bad shape of PAPR. Others are related to the very long delay compared to the terrestrial systems. There have been many studies regarding the former issue.

In this paper, we concentrate on the latter issue and try to introduce an alternative way of effective algorithms for uplink radio resource allocation for the satellite-airborne communication systems using the network simulator QualNet.

2 Issues and our goal

2.1 Assumptions

We assume that each airplane, which provides airborne Internet access to passengers, performs as a UE in the LTE based satellite network system. On the other hand, a GEO satellite has a capability of an eNB, and is not a bent-pipe type traditional communication satellite. This means that the roundtrip time between UE and eNB is approximately 250 msec, not 500 msec, although this is not the necessary condition for the proposed method.

2.2 Issues

A potential issue of the LTE’s traditional uplink resource scheduling method is the fact that they depend on BSR (Buffer Status Report) and has an assumption that the transmission buffer length information always propagates immediately. This is caused by the maximum delay between them, which is less than 20 microseconds. Instead, assume that the roundtrip time is about 250 msec. In this case, BSR is not the proper way of transferring necessary information for eNB to allocate uplink resource because the transmission buffer size, which is the amount of data to be transmitted, is valid only when it is being transferred and used for allocation within a few TTI period (here, 1 msec). Resource allocation using BSR under long delay should cause unstable bearer condition and may have some bad effects on the upper layer protocol behavior such as TCP, and affect unnecessary throughput declination.
of application layer and increase latency. To avoid this, it is evident that estimating the current amount of traffic and some control based on the estimation is required to stabilize the uplink traffic.

2.3 Related study

Uplink resource allocation methods which satisfy minimum throughput and maximum delay constraints were proposed in a paper [2]. Not many studies had been conducted in the same field regarding satellite communication based on terrestrial LTE technology. There are two radio interface technologies (RITs) proposed at the ITU-R: BM-SAT and SAT-OFDM. They were introduced in the ITU-R Recommendation M.2047 [3]. Both have little concern about resource scheduling under the constraint of long delay. Exceptionally, SAT-OFDM proposes a different approach regarding buffer status report from the original terrestrial LTE. SAT-OFDM mentions that BSR of the terrestrial LTE has no meaning for long roundtrip delay environment and proposes “successive reporting operation”, which reports increase (or decrease) of buffer size instead. There is no description about any particular method of resource allocation using this. Our study focuses on that point.

3 Proposed method in detail

3.1 Proposed resource allocation method

We introduce feedback control between UE’s transmission data amount and resource allocation at eNB. At the UE side, instead of reporting the current transmission buffer size with BSR, it reports the generated amount of data during the report interval to eNB. The eNB calculates the weighted average of that. As resource allocation method, we propose a PID control method as seen in Fig. 1. We define ‘Resource Margin’ as a control target. Here, the ‘Resource Margin’ means the difference between the allocated resource amounts and the generated data amount, which is y(t) and r(t) respectively in Fig. 1. At the early stage, we evaluated our method by basic level simulation using Microsoft Excel with (1) ‘Stable’ traffic (2) ‘Moderately Changing’ traffic and (3) ‘Rapidly Changing’ traffic

\[
\begin{align*}
\sum e(t) &\rightarrow K_p e(t) \\
\sum e(t) &\rightarrow K_i e(t) dt \\
\sum e(t) &\rightarrow K_d \frac{de(t)}{dt} \\
\end{align*}
\]

\[
r(t) = \text{Average of arrival traffic bytes} \\
e(t) = r(t) - y(t) \\
u(t) = \text{Control result for resource allocation (bytes)} \\
y(t) = \text{Resource bytes actually allocated.}
\]

\[
K_p, K_i, Kd \text{ are the constant factors which are determined appropriately.}
\]

Fig. 1. PID control (figure by Wikipedia)
under the assumption that the resources are always sufficient. The evaluation results showed that the ‘Resource Margin’ moves slightly in each case and we confirmed the effectiveness of the method. We named it as ‘SLA’ or ‘Sat-SLA’.

In addition to the PID-based control, we introduce the policy to allocate unused resources, if exists, to UEs. This turned out to absorb unintended steep increase of uplink traffic. This may relatively suppress dramatic increase of transmission buffer size, and therefore stabilize uplink resource allocation as the result. We proposed a simple resource allocation method with this idea and confirmed that this is effective for very long delay with simulation results [4].

3.2 Advantages of our study
It appears that resource allocation methods for terrestrial LTE is designed under the assumption that UEs and eNB always share the traffic information in real time manner. On the other hand, considering the inherent long delay characteristics of satellite communication, it is not realistic to control the resource allocation so as to trace traffic demand rapidly, as for the case of terrestrial communication with very short delay. We decided not to pursue real-time conformability to the traffic, and concentrated on stabilizing the amount of resource allocation to each uplink connection, which succeeded in drastically minimizing the packet delay.

4 Performance evaluation
We implemented the proposed resource allocation method into the LTE model library [5], which was modified to support very long delay, on the network simulator QualNet [6]. The network topology consists of a satellite as an eNodeB, an MME node, an application server host and five UEs. Each UE represents an airplane and accommodates all of the passengers’ uplink traffics, which goes to the application host through satellite. WiFi network model inside each airplane is omitted. The link between UE and eNB is modeled as LTE radio link, while the feederlink as an error free link, both with 120 msec delay. We compared the results with traditional LTE’s RR, PF and FIXED scheduling.

4.1 Application traffic model
We evaluate our resource allocation method with traffic models which combine the following application traffics [7].

a) Voice communication at 32 kbps
b) Video uploading at 128 kbps
c) Burst file transfer with FTP, the mean file size is 3 Mbytes
d) Web browsing with HTTP

In an aircraft, it is assumed that there are tens of passengers using their own applications and thus, the traffic between the airplane and the satellite would be the combination of them.

Regarding FTP and HTTP, this study intends to evaluate the performance of uplink of air-interface and excludes any uncertainty caused by the control of upper layer traffic, such as TCP. We choose a simple constant bit rate type UDP traffic model at this stage.
4.2 Simulation results

We conducted a series of network simulations. The graphs in Fig. 2 are the simulation results at the airplane which generate the highest traffic among them. Thus it is observed that some of the results show that the traffic is saturated at the peak load. In each graph, the blue line shows the packets’ delay. Orange bars show the number of received bytes at the server. Note that it is not the same as the amount of traffic because some packets have dropped at the peak traffic.

Fig. 2 shows that the application packet delay of our proposed scheduler drastically reduced against the others. With our LTE model, 5,000 ms is the maximum delay and packets which exceed this are dropped from the UE’s transmission buffer. In cases of ‘FIXED’, ‘RR’ and ‘PF’, other than ‘Sat-SLA’, packets’ delay exceed it or most of the time exceed 1,000 ms and approach to the limit. On the other hand, in case of ‘Sat-SLA’, the maximum delay is 1,600 ms and most of the time below 500 ms.

Table I shows the index values of min/max/average delay of each application traffic. They are average values for each UE. The min value for the index is from the FIXED scheduler. It appears that the UE with the smallest traffic results to good performance for FIXED resource allocation. For average delay and max delay among UEs, our proposed Sat-SLA scheduling achieved the best score.

We also observed another index, the transport block (TB) utilization [7]. This indicates the efficiency of the scheduler’s resource allocation from the point of fairness. If the deviation of the indicators among the UEs (airplanes) are small, it could be said that the scheduling is fair. With this indicator, our proposed scheduling shows the best results as follows.

<table>
<thead>
<tr>
<th></th>
<th>min</th>
<th>average</th>
<th>max</th>
</tr>
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<tbody>
<tr>
<td>FIXED</td>
<td>325.6/70.0</td>
<td>1209.5/79.5</td>
<td>4827.8/99.4</td>
</tr>
<tr>
<td>RR</td>
<td>358.2/78.7</td>
<td>815.4/84.3</td>
<td>2730.3/97.5</td>
</tr>
<tr>
<td>PF</td>
<td>396.3/78.2</td>
<td>1106.1/84.8</td>
<td>4217.0/99.6</td>
</tr>
<tr>
<td>Sat-SLA</td>
<td>464.3/83.7</td>
<td>570.5/85.3</td>
<td>823.7/88.3</td>
</tr>
</tbody>
</table>
5 Conclusion and further study

We evaluated the traditional LTE schedulers and our proposed scheduler under the condition of very long delay with the two indices, one is packet delay and the other is transport block (TB) utilization. We observed that the former index shows that our proposed scheduler got the best result for the UE which has the average and maximum uplink traffic. On the other hand, FIXED got the best result with minimum traffic. It would be inferred that under minimum traffic, which means that no resource control is required, FIXED, in other words “no control”, is better than SLA, elaborate resource control. With the result above, we can conclude that our proposed method is the best choice when resource control is truly required. The second index shows that from the aspect of TB utilization, our proposed scheduler shows very stable TB utilization value regardless of traffic conditions. With those results, we conclude that our proposed scheduler, Sat-SLA, brings us the best packet delay performance when the traffic deviation among UEs (airplanes) is relatively large. It can also maximize the resource utilization at the same time.

Most of the real world applications are based on TCP, rather than UDP, which we used in this paper. In the next stage, we would like to evaluate the performance of real world TCP applications with our satellite radio resource allocation method.

Acknowledgments

This study is conducted under a commissioned research of the “Research and development on narrowband technology using active electronic scanning array antenna that can be mounted on small aircrafts” by the Ministry of Internal Affairs and Communications.
Efficient acquisition of map information for service robots using local data sharing over multi-layered wireless network*

Rui Teng, Shirayuki Araki, Satoru Shimizu, Kazuto Yano, and Yoshinori Suzuki
Wave Engineering Laboratories, Advanced Telecommunications Research Institute International, Kyoto 619–0288, Japan

Abstract: This letter aims to address the problem of wireless resource constraints in wireless robotic networks to enable stable acquisition of map information for a number of robots. In order to reduce wireless resource consumption, we propose a two-step data acquisition scheme for robots. In the proposed scheme, only a few robots directly download map information from the server, most robots acquire map information by means of local data sharing with short range communications among nearby robots. A distributed clustering scheme is utilized to decide which nodes download map information from the server. Evaluation results show that the proposed scheme highly reduces wireless resource consumption, especially when there are many service robots.

Keywords: robotic wireless network, wireless resource efficiency

Classification: Wireless Communication Technologies

References


*(The contents of this manuscript have been partly presented in [3] and [4].)
1 Introduction

A robotic wireless network consists of a map server and a number of robots, which are connected via a wireless network such as wireless local area networks (WLANs) or Long Term Evolution (LTE) [1, 2]. Fig. 1(a) illustrates the conventional wireless robotic network, in which each robot downloads map information directly from the map server. Increasing number of service robots causes large interference among robots and heavy load on the access point (AP) or base station (BS) [3, 4]. This in turn leads to possible failures of stable acquisition of map information for robots.

To overcome this problem, this letter proposes a two-step acquisition of the map information. At first, some of robots directly download the map information from the server. Next, the downloaded map information is shared among robots within the area where the downloaded information is valid. By sharing information with a small transmission power, interference range is reduced, and spectral reuse is enabled. The detail system structure and how to determine the robot downloading the map information and the range of information sharing, i.e. how to make a cluster for local data sharing, are introduced in the next section. Different with the scheme proposed in reference [2], this letter proposed a distributed, simple, and fast clustering scheme that achieves similar effectiveness in saving the consumption of wireless resources.

2 Two-step acquisition of map information with wireless resource efficiency

2.1 System structure

As shown in Fig. 1(b), the key idea of the proposed system is that two-step acquisition of map information is carried out among robots. At first, only a part of robots in a robotic network need to download map information from the server. Then, in the second step, other robots obtain the map information from a nearby cluster head using a low transmission power. This leads to efficient spatial reuse of wireless resource and increase the number of accommodated robots.

In the proposed system, three kinds of radio interfaces are employed at each robot [2], as illustrated in Fig. 1(c) that shows a practical WLAN-based wireless node for the robotic wireless network. One radio interface supports long-range communication to the AP or BS, provided with a large transmission power for a wide coverage. Another interface is for short-range communication to locally share data among nearby robots, and it has a low transmission power to cover only nearby robots. The third radio interface is used as a control interface, which periodically sends beacon signals to its nearby neighbors. Such periodic beacons allow a robot to detect its neighbors and form clusters among nodes for local communications. For easy reconfiguration of cluster and distributing control messages among robots, we suppose to use independent basic service set (IBSS) of WLAN for the second and third radios.
2.2 Distributed clustering of robots

In the proposed robotic wireless system, a distributed-and-paralleled clustering scheme is utilized to determine which nodes should download the map information and which nodes should the map information be shared with. Compared with the scheme proposed in reference [2] that utilizes multiple phases of clustering for local optimization, the proposed scheme is simple, fast for clustering, and does not require strict synchronization among robots. In the distributed clustering procedure, each robot carries out the cluster formation process periodically. The basic cluster formation process at a stand-alone robot X that does not belong to any cluster is described as follows. 1) Robot X becomes a cluster head at a probability $P$; 2) if X becomes a cluster head, it issues cluster head announcements to its neighbors using the radio for sending beacons; 3) if X, which does not become a cluster head, receives a message of cluster head announcement from its local neighbor node, it then joins the cluster and becomes a client of the cluster.

![Diagram](image)

(a) Conventional wireless system. (b) Proposed wireless system. (c) A wireless station. (Functional interfaces can be flexibly configured.) (d) An example of prioritized clustering scheme. “X(n)” means that a robot X has n the number of local neighbors.

**Fig. 1.** An overview of multi-layered wireless system for service robots.

### 2.2 Distributed clustering of robots

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2.3 Prioritized clustering scheme

We further propose an advanced clustering scheme that controls the cluster head generation in a distributed manner. Each candidate of robot has a probability $Prob_{CH}$. $Prob_{CH}$ is designed to be proportional to the number of local neighbours. This is enabled by prioritized generation of cluster-head announcement time. The control algorithm is as follows. Let $Nei[X]$ denote the number of local neighbours of a robot $X$. Given a time pool $P_a[x]$ for a robot to issue the cluster head announcement in each clustering period. The maximum $P_a[x]$ is set to $S$ slot units, where $S$ can be an integer multiple of the number of robots in the network. The announcement time $T[x]$ at each robot is randomly generated in $P_a[x]$. By allowing $P_a[X] = S - Nei[X]$, the robot $X$ with more number of local neighbours has a higher probability to issue a announcement of cluster head than that with less number of local neighbours. An example is shown in Fig. 1(d), in which robot A has a higher probability of being a cluster head than others. To synchronize the start time of clustering, AP periodically issues clustering formation message that triggers the cluster formation each time.

3 Simulation evaluation

To validate the effectiveness of the proposed approach on reducing wireless resource consumption, computer simulation is conducted. We compare the performance of the proposed schemes and with the conventional scheme that uses WLAN to directly download map information from the server. Furthermore, we also investigate the performance difference between the proposed schemes and the local optimized clustering scheme proposed in [2]. The simulation parameter setup is shown in Table I. The basic setup of network environment is shown in Fig. 2(a). We assume that the path loss exponent is 3, supposing the possible building environment. The transmission powers of long-range and short-range communication are set to 14.1 dBm and −3.98 dBm, respectively. Since this letter focuses on the basic features of constructing map-sharing clusters, the movement of robot nodes is not taken into account in the evaluation.

In the evaluation, we measure the wireless resource consumption by

$$WirelessResource = BW \times OccuTime \times InterferenceArea,$$  

(1)

with a unit of MHz·second·m². In this expression, BW stands for the channel bandwidth. Channel occupancy time (OccuTime) is the minimum time that the transmission channel is used to transmit the object map data to a robot. In the simulation, OccuTime is calculated by (Map data size/Data rate) in order to reflect the fundamental time factor to transmit the map data to a robot. The interference area of a robot refers to the size of the area in which the received power is greater than −82 dBm. To explicitly represent the impact of the multi-layered wireless structure on wireless resource consumption, we assume that each robot is able to obtain map information in a separated time regarding to other robot, ignoring the contention effect of channel access among robots.

Clustering results are shown in Fig. 2(b). For the basic distributed clustering scheme that does not employed prioritized selection of cluster head, the ratio of client number to the total number of robots increases from above 28 percent to 79
percent as the total number of robots increases from 5 to 50. The result verifies the effectiveness of data sharing cluster that requires only a small number of robots to be cluster heads for global data communication with the AP in a WLAN. The reason of this result is that the increase of robots causes the increase of the local neighboring robots, leading to more candidate clients of a cluster head. Moreover, the prioritized scheme increases the client ratio by 2.7 percent in average compared with the basic distributed cluster scheme. The reason of this result is that the prioritized clustering scheme allows a robots with more local connected robots to be a cluster head with a higher probability.

Fig. 2(c) shows the results of wireless resource consumption relative to the conventional scheme in downloading map information. The basic distributed clustering scheme results in a significant reduction of wireless resource consumption, compared with conventional scheme that directly downloads the map information from the server. Wireless resource consumption is reduced by 24 percent for the network with 5 robots, and the reduction grows to 71 percent when the number of robots is 50. These results are mainly attributed to the small interference area in data acquisition among robots by using the local communication between client nodes and their cluster head. The prioritized scheme further saves 4.4 percent wireless resource consumption in average compared with the basic distributed cluster scheme. The degradation in wireless resource consumption for the basic clustering scheme, relative to the local optimized scheme, is reduced by 47 percent when utilizing the prioritized scheme.

Meanwhile, local optimized scheme may require multiple rounds of clustering process that attempt to allow each robot to join in an appropriative cluster [2]. As shown in Fig. 2(d), the basic distributed scheme and the prioritized scheme requires only one round of clustering process to allow each robot to finish its clustering process, and local optimized scheme requires up to 1.5 rounds of clustering process in average when the number of robots is large.

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Setup</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number of Robots</td>
<td>5–50</td>
</tr>
<tr>
<td>Network area</td>
<td>40 m × 40 m square</td>
</tr>
<tr>
<td>Positions of robots</td>
<td>Random with uniform distribution</td>
</tr>
<tr>
<td>Number of robot topology settings</td>
<td>100 (each is static)</td>
</tr>
<tr>
<td>Position of the AP</td>
<td>(20, 20)</td>
</tr>
<tr>
<td>Data size of map information</td>
<td>2 Mbits</td>
</tr>
<tr>
<td>Bandwidth</td>
<td>20 MHz</td>
</tr>
<tr>
<td>Data rate</td>
<td>Adaptive (referring to IEEE 802.11g WLAN)</td>
</tr>
<tr>
<td>Path loss exponent</td>
<td>3</td>
</tr>
<tr>
<td>Transmission power (long range radio)</td>
<td>14.1 dBm</td>
</tr>
<tr>
<td>Transmission power (short range radio)</td>
<td>−3.98 dBm</td>
</tr>
</tbody>
</table>
4 Conclusion

This letter studied the efficient use of the limited wireless resources for wireless robotic networks. Reducing wireless resource consumption is significant for a number of service robots to enable stable acquisition of map information. We proposed a robotic wireless system that employs two-step acquisition of map information in order to cope with the wireless resource constraints for a number of service robots. In the proposed wireless system, robots are coordinated by means of distributed clustering in an autonomous manner to download and locally share map information. The locally sharing of map information in each cluster with short-range communication leads to wireless resource reuse and the saving of area-spectral consumption of wireless resources. Evaluation results reveal that the proposed scheme achieves significant reduction of resource consumption especially when there are a large number of robots in a network.

Acknowledgment

This work is supported by the Ministry of Internal Affairs and Communications, Japan.
Mode recoupling between core and cladding modes of cascaded-LPFGs fabricated with heat-shrinkable tube employing a thin confinement layer

Yasuhiro Tsutsumi\textsuperscript{1,2a)}, Takahiro Hase\textsuperscript{1}, Masaharu Ohashi\textsuperscript{1}, Yuji Miyoshi\textsuperscript{1}, Hirokazu Kubota\textsuperscript{1}, and Ikuo Yamashita\textsuperscript{3}

\textsuperscript{1}Department of Electrical and Information Systems, Graduate School of Engineering, Osaka Prefecture University, 1–1 Gakuen-cho, Naka, Sakai, Osaka 599–8531, Japan
\textsuperscript{2}College of Science and Engineering, Ritsumeikan University, 1–1 Nojihigashi, Ksatsu, Shiga 5258577, Japan
\textsuperscript{3}Technical Research Center, Kansai Electric Power Inc., 3–11–20 Nakouji, Amagasaki, Hyogo 6610974, Japan

\textsuperscript{a)} TsutsumiYasuhiro@mem.iee.or.jp

Abstract: Cascaded long period fiber gratings (LPFGs) fabricated with heat-shrinkable tube, screw, and single-mode fiber are proposed herein that employ a thin confinement layer on the bare fiber. The transmittance of a proposed cascaded-LPFG is theoretically and experimentally investigated regarding the cladding mode loss. By employing the confinement layer on the cladding of the proposed LPFGs, mode recoupling between core and cladding modes is successfully induced. Our cascaded-LPFG has potential application for the simple and low-cost optical filters and for the optical sensors monitoring vibration, refractive index, or temperature.

Keywords: mechanically induced long period fiber grating, fiber based Mach-Zehnder interferometer, cladding mode loss, cascaded-LPFG

Classification: Optical Fiber for Communications

References


1 Introduction

Long period fiber gratings (LPFGs) are attractive devices for sensor applications owing to their unique advantages: electro-magnetic immunity, nonexplosiveness, and potential for remote and multiplexing operations [1, 2]. Cascaded-LPFGs that connect two LPFGs in series are promising sensors owing to their high sensitivity for physical parameters such as torsion, temperature, or the refractive index of the surrounding medium [3, 4, 5]. Such high sensitivity for physical parameters arises from a recoupling between core and cladding modes, causing interference between the core and cladding modes [6, 7]. In a cascaded-LPFG, mode coupling between core and cladding modes occurs at the first LPFG, and core and cladding modes are transmitted. At the second LPFG, mode recoupling between cladding and core modes is induced, producing interference between core and cladding modes. Therefore, a cascaded-LPFG acts as a fiber-based Mach–Zehnder interferometer (MZI) [6, 7].

Recently, we proposed a simple fabrication technique for mechanically-induced LPFGs using heat-shrinkable tube [8]. Our LPFGs are fabricated by inserting the fiber between the tube and screw thread [8] and occupying the fiber position where the refractive index periodically changes. As our LPFGs are wrapped with a tube and fixed by the tube and screw thread [8], the LPFG is protected by the tube from the refractive index change of the surrounding medium and the disturbance such as bending of the LPFG. Because of these features, they cannot be used as sensors for measuring the surrounding refractive index and the bending of the LPFG. To overcome these problems, a fiber-based MZI containing cascaded-LPFGs would be effective because the transmittance is sensitive towards the changes in physical
parameters at the interval fiber between the LPFGs [9]. A cascaded-LPFG of fiber-based MZI accomplishes all those using our LPFGs protected by the tube. It can be used as the sensor which doesn’t detect the changes in the physical parameters at the LPFGs, but in the fiber between the LPFGs. However, mode recoupling between cladding and core modes did not occur when our LPFGs fabricated via heat-shrinkable tube were cascaded [10].

In this study, we propose a simple structure of cascaded-LPFGs fabricated with heat-shrinkable tube that causes mode recoupling between core and cladding modes. To reduce the cladding mode loss, a thin metallic layer is employed. Using a transfer-matrix model, we investigate the transmittance of the cascaded-LPFG, focusing on the cladding mode loss at the LPFG [9], and clarify that the mode recoupling of the conventional cascaded-LPFG fabricated with heat-shrinkable tube [10] does not occur due to high cladding mode losses.

2 Theory

Fig. 1 shows the structure of a cascaded-LPFG that connects two LPFGs with the same parameters in series.

The field amplitudes $A_{co}$ and $A_{cl}$ which correspond to respective the core mode and cladding mode can be written as follows by using the transfer matrix $F_L$ of the LPFG and the transfer matrix $F_F$ of the fiber between the LPFGs based on the transfer matrix analysis [9].

$$
\begin{bmatrix} A_{co} \\ A_{cl} \end{bmatrix} = F_L F_F F_L \begin{bmatrix} 1 \\ 0 \end{bmatrix}
$$

(1)

The transfer matrix $F_L$ of the LPFG is expressed as follows [9].

$$
F_L = \begin{bmatrix}
\cos(\gamma_c L) + i \frac{\delta}{\gamma_c} \sin(\gamma_c L) & i \frac{\kappa}{\gamma_c} \sin(\gamma_c L) \\
- \frac{i \kappa}{\gamma_c} \sin(\gamma_c L) & \cos(\gamma_c L) - i \frac{\delta}{\gamma_c} \sin(\gamma_c L)
\end{bmatrix},
$$

(2)

$$
\delta = \pi \Delta n_{\text{eff}} \left( \frac{1}{\lambda} - \frac{1}{\lambda_p} \right) \cong \pi \frac{\lambda_p}{\Lambda} \left( \frac{1}{\lambda} - \frac{1}{\lambda_p} \right), \quad \gamma_c = \sqrt{\kappa^2 + \delta^2},
$$

(3)

where $L$ and $\Lambda$ denote the length and pitch of the LPFG, respectively. $\Delta n_{\text{eff}}$ gives the effective index difference between core and cladding modes. $\lambda_p$ is a resonance wavelength of the LPFG. $\kappa$ and $\delta$ are the coupling coefficient and detuning factor of the LPFG, respectively [9].

Fig. 1. Structure of a cascaded-LPFG fabricated with heat-shrinkable tube and an aluminum foil layer.
For LPFGs fabricated with heat-shrinkable tube, the propagation losses of cladding modes occur due to the microbending. Moreover, when the refractive index of the material surrounding the cladding is sufficiently greater than that of the cladding, then the propagation losses of cladding modes occur due to coupling with leaky modes [6, 7].

To express the cladding mode loss in the LPFG, the transfer matrix of the LPFG with cladding mode loss is defined as follows.

\[
F_L = \begin{bmatrix}
\cos(\gamma_L L) + i \frac{\delta}{\gamma_L} \sin(\gamma_L L) & i \frac{\kappa}{\gamma_L} \sin(\gamma_L L) \\
\frac{\kappa}{\gamma_L} \sin(\gamma_L L) & e^{-2} \left[ \cos(\gamma_L L) - i \frac{\delta}{\gamma_L} \sin(\gamma_L L) \right]
\end{bmatrix},
\]

(4)

where \( \alpha \) represents the cladding mode loss in the LPFG.

The transfer matrix of the fiber between the LPFGs is expressed as follows.

\[
F_F = \begin{bmatrix}
\exp \left( \frac{2\pi \Delta n_{\text{eff}} D}{\lambda} \right) & 0 \\
0 & \exp \left( \frac{-2\pi \Delta n_{\text{eff}} D}{\lambda} \right)
\end{bmatrix},
\]

(5)

where \( D \) is the fiber length between the LPFGs.

The transmittance \( T \) of the cascaded-LPFG can be calculated by the following equation.

\[
T = |A_{co}|^2
\]

(6)

We calculated the transmittance of the cascaded-LPFG with a length \( L \) of 5.5 cm, a grating period \( \Lambda \) of 0.7 mm, and a resonance wavelength \( \lambda_r \) of 1607 nm using the above model for the case of no attenuation loss in the fiber between the LPFGs.

Figs. 2(a)–(d) show the transmittance of the cascaded-LPFG with no cladding mode loss \((\alpha = 0)\) for various distances \( D \) between the LPFGs. To simplify the calculations, we ignored the wavelength dependence of the coupling coefficient \( \kappa \) and the effective index difference \( \Delta n_{\text{eff}} \) between core and cladding modes, and we set \( \kappa \) at 19 and \( \Delta n_{\text{eff}} \) at 0.0022956 considering experimental results [8]. We found

![Fig. 2. Calculated transmittance of cascaded-LPFGs for various distances \( D \) (a) \( D = 0 \) cm, (b) \( D = 15 \) cm, (c) \( D = 30 \) cm, and (d) \( D = 50 \) cm. Calculated transmittance of cascaded-LPFGs with \( D = 50 \) cm for various values \( \alpha \), (e) \( \alpha = 0.5 \) Np, (f) \( \alpha = 1.0 \) Np, (g) \( \alpha = 1.5 \) Np, and (h) \( \alpha = 2.0 \) Np.](image-url)
that when the distance $D$ is 0, the transmittance of the cascaded-LPFG equals that of a united-LPFG which is connected the LPFGs with a length of $L$. The wavelength spacing between the adjacent peaks of the cascaded-LPFG decreases as the distance $D$ increases, and therefore the attenuation bandwidths decrease as the distance $D$ increases.

We also calculated the transmittance of the cascaded-LPFG for various values $a$ of cladding mode loss, as shown in Figs. 2(e)∼(h). The cladding mode loss $a$ (Np) was set to each of four values: 0.5, 1.0, 1.5, and 2.0. These correspond to 2.17 dB, 4.34 dB, 6.51 dB, and 8.69 dB, respectively. In the simulations, we set the distance $D$ between LPFGs at 50 cm. We found that the fringe visibility [6, 9] decreases as the cladding mode loss increases. For high cladding mode losses, beats caused by the interference between core and cladding modes could not been observed. It is seen from Figs. 2(a) and (e)∼(h) that the attenuation bandwidth of cascaded-LPFGs broadens compared with a united-LPFG as the cladding mode loss increases. The measured spectrum of the united-LPFG fabricated with a heat-shrinkable tube shows the similar one calculated for the high cladding mode loss [8, 10]. Therefore, we clarified that the mode recoupling of the conventional cascaded-LPFG fabricated with the heat-shrinkable tube cannot occur due to the high cladding mode loss, and that the cladding mode loss is an important factor for controlling the transmittance of the cascaded-LPFG.

### 3 Experimental results

As illustrated in Fig. 1, to confine the cladding mode into the fiber, the bare fiber was covered with a thin layer of aluminum foil which reduce the microbending loss and change the refractive index of the surroundings. We fabricated the cascaded-LPFG in the following way. First, we put a bare optical fiber on a sheet of aluminum foil, then wrapped a screw thread with the foil. We inserted the wrapped thread and fiber into the heat-shrinkable tube and heated the tube. By heating the tube, the tube shrinks and the fiber contacts the thread. After cooling the tube to room temperature, the periodic refractive index modulation was induced by the photo-elastic effect, thereby creating a LPFG [8]. In the same manner, we fabricated a 2nd-LPFG with aluminum foil at a distance $D$ from the 1st-LPFG.

The cascaded-LPFG was fabricated with the heat-shrinkable tube (Hagitech-NF040), standard-optical telecommunication fibers (ITU-T. G.652), aluminum foil of thickness 75 μm, and screw threads with a pitch $A$ of 0.7 mm and length $L$ of 5.5 cm. The distance $D$ between the LPFGs was 50 cm. The conventional cascaded-LPFG with high loss of cladding mode [8] was fabricated as a reference sample using the same components but excluding the aluminum foil. Figs. 3(a) and (b) show the measured transmittances of the cascaded-LPFGs with and without an aluminum foil layer, respectively. The measured transmittance of the cascaded-LPFG with aluminum foil showed a lot of narrow attenuation bandwidth. The measured spectrum in Fig. 3(a) was in good agreement with the simulation one for the low cladding mode loss in Fig. 2(g). In contrast, the transmittance of the cascaded-LPFG without the aluminum foil shows a broader attenuation bandwidth. The measured spectrum in Fig. 3(b) was in good agreement with the simulation one
for the high cladding mode loss in Fig. 2(h). It is found that the confinement layer is effective for reducing the cladding mode loss. However, the fringe visibility of the cascaded-LPFG with the aluminum foil is small, so the cladding mode loss is slightly higher. This slightly higher loss is attributed to an imperfect interface surrounding cladding, which may cause the scattering loss [6]. In addition, since we wrapped only a part of the fiber with aluminum foil, losses may also have arisen from the part of the fiber in contact with the air and screw thread inside the tube. The fringe visibility could be improved by optimizing the structure of the confinement layer.

4 Conclusions

We proposed a simple fabrication method for a cascaded-LPFG fabricated with heat-shrinkable tube and a thin confinement layer of aluminum foil. We investigated the transmittance of the cascaded-LPFG as a function of the cladding mode loss using a transfer matrix analysis. We clarified from our simulation results that the attenuation bandwidth of the cascaded-LPFG can be determined by the cladding mode loss. Considering the similarity of the spectra, we also clarified that the mode recoupling of the conventional cascaded-LPFG cannot occur due to the high-cladding mode losses. We confirmed experimentally that the proposed structure of the cascaded-LPFG employing a thin confinement layer on the cladding is effective for confining the cladding mode into the fiber and for inducing the mode recoupling between the core and cladding modes. As the result, controllability of transmission of cascaded-LPFG are improved significantly.

Cascaded LPFGs fabricated with a heat-shrinkable tube have potential uses for the simple and low-cost optical filters and for the optical sensors monitoring vibration, refractive index, or temperature.

Acknowledgments

This work was supported by JSPS KAKENHI Grant Number 16K06307.
Digital-analog hybrid transmitter equalizer for multi-valued signaling

Yasushi Yuminaka\textsuperscript{1a)} and Yosuke Iijima\textsuperscript{2}

\textsuperscript{1} Graduate School of Science and Technology, Gunma University, 1–5–1 Tenjin-cho, Kiryu 376–8515, Japan
\textsuperscript{2} National Institute of Technology (KOSEN), Oyama College, 771 Nakakuki, Oyama 323–0806, Japan
a) yuminaka@gunma-u.ac.jp

Abstract: High-speed data transmission over electric wiring in a VLSI system is achieved by employing equalization circuitry for waveform shaping. This letter presents a novel transmitter that incorporates waveform shaping techniques especially for multi-valued signaling. The combination of digital and analog equalizers and adjustment of the digital parameter can improve the waveform distortion of rising and falling edges. Simulation and experimental results of multi-valued data transmission are shown to demonstrate the feasibility of the proposed transmitter, which is capable of controlling the received waveforms flexibly and improve the signal integrity.

Keywords: multi-valued logic, PAM-4, equalization
Classification: Transmission Systems and Transmission Equipment for Communications

References


1 Introduction

A high-speed serial link beyond several tens of Gbps is required for chip-to-chip, backplanes, and data center transmission. However, at high-speed data rates, electric wires behave as low-pass filters that destroy the high-frequency characteristics of the transmission lines. The limited bandwidth of the channel, therefore, causes intersymbol interference (ISI) at the receiver; hence, waveform shaping techniques are required to remove the ISI. Moreover, multi-valued (MV) signaling is widely used to increase the information capacity per symbol [1, 2]. Compared with binary signaling, which transmits the digits 0 or 1, MV signaling can transmit more than double the information capacity in each symbol. However, because the waveform distortion becomes complex owing to the many data transition patterns of MV signaling, controlling the received waveforms becomes difficult.

To solve these problems, we previously proposed waveform shaping techniques based on Tomlinson-Harashima Precoding (THP) at the transmitter to double the operating frequency compared with the symbol rate, which can sharpen the transition edge of the received waveform [3, 4]. However, these techniques incur additional hardware costs as a high-speed digital-to-analog converter (DAC) is required because of the doubled symbol rate.

This letter proposes a new waveform shaping transmitter that combines digital and analog equalizers to lower the DAC operating frequency. This transmitter operates at the frequency of the symbol rate, obviating the need for DACs that operate at twice the frequency of the symbol rate. The transmitter cooperates with analog pre-emphasis at half the symbol rate that can improve the edges of received signals by setting the parameters of the digital circuits. A simulation and experiments are used to demonstrate the feasibility of the proposed transmitter and its ability at the architecture level in order to control the received waveforms flexibly and improve the signal integrity.

2 Waveform shaping technique for PAM-4 signaling

2.1 PAM-4 data transmission

In 4-level pulse amplitude modulation (PAM-4) data transmission, 2-bit binary data are converted to 4-level MV data, thereby enabling PAM-4 to transmit 2-bit data as one symbol. Compared with binary signaling, PAM-4 is advantageous in that it uses half the Nyquist frequency and twice the throughput for the same Baud rate, and hence the lowpass effects of channels are reduced. However, as shown in Fig. 1(a), the PAM-4 eye shape becomes nonuniform in each symbol owing to the 12-types different transition patterns of MV signaling [5]. The eye width of the symbols between 1–2 is restricted by both the transition patterns of 0 → 2 and 3 → 1. Fig. 1(b) shows the 1 ns pulse responses of a micro-strip line (MSL) 1 m for each symbol level. Because the waveforms of the pulse responses are different, the edge slope changes in accordance with the transition pattern. The different edge patterns have a profound effect on the eye diagrams of received MV signaling.
2.2 Waveform shaping technique

The signal distortion due to ISI can be improved using signal-processing techniques known as equalization. Feed-forward equalizer (FFE) at the transmitter can be realized by digital filter circuits composed of adders, multipliers, and delay circuits at the symbol rate $T_s$. Output signals are generated by the product sum of the FFE coefficients $a_1, a_2, \ldots, a_n$ and input data. The number of taps is determined by the transmission line characteristics, and the delay time of $D$ is set the same as the symbol rate $T_s$. Although the FFE can remove the ISI effect, shaping the waveform such that the eye shape is modified is difficult, especially to improve the rising and falling edges of waveforms. This is because the frequency characteristics of FFES are restricted by the operating speed of FFE circuits, which corresponds to the inverse of the delay time in the FFE.

To improve the edges and expand the eye, we previously proposed double-rate THP (DTHP) equalization techniques that employ $T_s/2$ delay operation [4]. The technique can control the eye shape more flexibly compared with a conventional FFE by operating at double the symbol rate. However, the DTHP needs a high-speed DAC that operates at double the symbol rate, which increases the hardware cost.

3 Hybrid transmitter combining digital and analog equalizer

To control the received waveform flexibly to improve the rising and falling edges, this letter proposes new waveform shaping transmitters for MV signaling by combining digital and analog equalizers. This transmitter does not require a DAC that operates at double the symbol rate, avoiding an increase in the hardware cost. Fig. 2(a) shows an overview of the transmitter. The proposed transmitter is realized by operating the FFE at symbol rate $T_s$, and is equipped with DACs and analog circuitry. Transmitter signals are generated by subtracting the FFE output signals from the PAM-4 signals. The PAM-4 input data are converted by 2-bit resolution DAC1, and data processed by the FFE is converted by DAC2. The output signal is then generated by subtracting the output signals of DAC1 and DAC2 using the analog circuitry. In the transmitter, DAC2 is operated by inverse clock timing compared to DAC1 to realize delay with a doubled symbol rate ($T_s/2$).
The proposed transmitter generates the high-frequency components of signals by subtracting signals with an inverse timing. The high-frequency component contributes to the improvement of rising and falling edges of signals. Moreover, the output waveforms can be controlled digitally, by tuning the FFE parameters $a_1, a_2, \ldots, a_n$. The parameters, therefore, can be changed easily according to the environment after hardware implementation. Fig. 2(b) shows the simulated output waveforms of the proposed transmitter. The frequency of the transmitter signal is double that of the PAM-4 signal and FFE output. The high-frequency signal generated by the proposed transmitter can control the received waveform flexibly and sharpen the rising and falling edges of the signals.

4 Experimental results

As proof-of-concept at the architecture level, an arbitrary waveform generator (AWG) AWG70001A was used to emulate the DAC operation with FFE and an analog subtractor (Fig. 3(a)). The imported FFE-modulated random signal data were calculated by numerical simulation considering a 4-tap FFE with 2-bit and
6-bit resolution for DAC1 and DAC2, respectively. Figs. 3(b) and (c) show the results of the simulation and measurement of the PAM-4 eye diagrams with conventional FFE and the proposed transmitter at 2 Gsps. In conventional FFE, fine tuning the delay time is difficult. As a result, it is difficult to adjust the shape of the eye. In contrast, because the proposed transmitter is composed of digital circuitry, digital fine adjustment is possible, thereby achieving flexible waveform shaping by setting the FFE coefficients. Compared with conventional FFE, the proposed transmitter can improve the rising and falling edges of the received waveform to widen the eye width from 0.46 UI to 0.68 UI (UI: unit interval).
Because the processing speed limits conventional FFE to the high-frequency characteristics, it is difficult to emphasize the edges of the received waveform. In contrast, the proposed transmitter can realize high-frequency characteristics simply by subtracting the PAM-4 and FFE delayed signal to obtain $T_s/2$ delay. The effect of boosting the high-frequency components can especially widen the time axis direction of the eye diagram. The extension of the time axis direction significantly contributes to the improvement of the jitter tolerance, especially for PAM-4 data transmission.

The non-ideal duty ratio of the clock signal and the nonlinearity of the analog subtraction circuit cause a timing mismatch between the outputs of DAC1 and DAC2. To evaluate the imperfection, these effects were simulated for the case of a $T_s/2$ delay with an error of plus or minus 5% (in this case, ±12.5 ps). As shown in Figs. 3(d) and (e), the timing mismatch does not have a significant impact on the output eye diagram. Although the analog subtraction circuit can operate faster than the DACs, the operating frequency and nonlinearity of the analog circuitry deteriorate the shape of the transmitter signal. Because this limitation and effect are strongly dependent on the topology of the analog circuit, evaluations of the imperfection using transistor-level circuit simulation are needed as a future task.

5 Conclusion

This letter presented a new waveform shaping transmitter combined with digital and analog equalizers. The proposed transmitter can control the received waveform flexibly and is suited to PAM-4 data transmission without using high-speed equalizers and DACs. The experimental results showed that the proposed equalizer can compensate for the effect of waveform distortion, especially by expanding the eye width opening.

Acknowledgments

This work was supported by JSPS KAKENHI Grant Numbers JP18H01488 and JP18K11232.
Adaptive transmission probability for CSMA/CA-based consensus control of multi-agent systems

Kentaro Kobayashi1a), Shunsuke Noro2, Hiraku Okada1, and Masaaki Katayama1

1 Institute of Materials and Systems for Sustainability, Nagoya University, C3–1(631) Furo-cho, Chikusa-ku, Nagoya 464–8603, Japan
2 Department of Electrical Engineering and Computer Science, Graduate School of Engineering, Nagoya University

a) kobayasi@nuee.nagoya-u.ac.jp

Abstract: This paper deals with consensus control for multi-agent systems over CSMA/CA(Carrier Sense Multiple Access with Collision Avoidance)-based wireless networks. In consensus control, the number of agents within the communication range changes with time according to the movement of the agents and affects the packet collision rate. In this paper, we propose a new method to reduce the packet collision rate considering consensus dynamics of the agents. In the proposed method, each agent distributedly estimates the priority of its own position information on the basis of position information received from its adjacent agents and then adjusts transmission probability according to the estimated priority. Simulation results show that the proposed method improves the control quality.

Keywords: wireless control, multi-agent systems, consensus control, CSMA/CA, packet collision

Classification: Wireless Communication Technologies

References


1 Introduction

Coordinated control of autonomous mobile robots has been attracted attention in many industrial fields. Multi-agent system is one of control systems to realize such a coordinated control system, and is a distributed control system that consists of autonomous agents and whose behavior is determined by interaction between the agents. Consensus control that converge each agent’s state to a certain consensus state only by exchanging information with adjacent agents has been treated as a basic control problem of multi-agent system. When wireless communication is used for exchanging information among the agents in consensus control, the degradation of control quality due to communication failure becomes a problem.

Most research have been conducted to design control methods to reduce the influence of communication failure, e.g. [1, 2, 3]. The communication failure in these studies is assumed to occur with a constant probability. However, considering that the agents move toward a consensus position, the communication failure probability changes because the number of agents in the communication range changes with time, and the probability of collision of transmitted packets also changes. Therefore, for the improvement of control quality, it is necessary to reduce packet collision considering the movement of the agents. From this view point, [4] has proposed a slotted ALOHA-based communication method and shown that setting of transmission probability based on the number of adjacent agents can reduce packet collision and improve the control quality.

In this paper, for CSMA/CA-based wireless consensus control, we propose a method to set transmission probability on the basis of estimated importance of agent’s position information, and show that the proposed method can reduce the packet collision rate and improve the control quality.

2 System model

This paper deals with an average consensus control problem [5] in which multiple autonomous mobile agents move on the basis of position information acquired from adjacent agents within a communication range and get together to a consensus position, namely, the average of their initial positions.

2.1 Multi-agent system

Agent $i (\in \{1, 2, \ldots, N\})$ is represented by the following state equation.

$$ x_i[k+1] = x_i[k] + u_i[k], \quad x_i[0] = x_{0i}, $$

where $x_i[k]$ and $u_i[k]$ are two-dimensional vector and represents position information and control input at time $t = kT_s$ ($T_s$: control period), respectively. $x_{0i}$ is the
initial position. Each agent linearly moves in a two-dimensional plane at a constant speed not exceeding the maximum moving speed $v_{\text{max}}$.

The network among the agents changes with time because of the movement of the agents and communication failure due to packet collision. The network at time index $k$ is defined by a directed graph $\mathcal{G}[k] = (\mathcal{V}, \mathcal{E}[k])$, where $\mathcal{V} = \{1, 2, \ldots, N\}$ is a set of agents that compose the graph and $\mathcal{E}[k] = \{(i, j) \in \mathcal{V} \times \mathcal{V}\}$ is a set of agents with a communication path. The communication range of each agent is defined by radius $R$. When communication from agent $j$ to $i$ is successful within the control period, i.e., $(j, i) \in \mathcal{E}[k]$, agent $j$ is defined as adjacent to $i$. All adjacency relationships are represented by an adjacency matrix $\mathbf{A}[k] = [a_{ij}[k]] \in \{0, 1\}^{N \times N}$, where $a_{ij}[k]$ is 1 if the agent $j$ is adjacent to $i$ and 0 otherwise. Adjacent agents of agent $i$ are defined by $\mathcal{N}_i[k] = \{ j \in \mathcal{V} | (j, i) \in \mathcal{E}[k], i \neq j \}$, which is a set of agents where the agent $i$ is adjacent.

### 2.2 Consensus control

The control input $u_i[k]$ of agent $i$ is given by (2) using the deviation of its own position $x_i[k]$ and the position $x_j[k]$ received from adjacent agents.

$$u_i[k] = -\frac{1}{|\mathcal{N}_i[k]| + 1} \sum_{j=1}^{N} a_{ij}[k](x_i[k] - x_j[k]), \quad (2)$$

where $|\mathcal{N}_i[k]|$ is the number of adjacent agents. This results that each agent moves to the center of gravity of the positions. A consensus is achieved by distributedly determining the control input every control period. As described in (3), the consensus means that from any initial positions $x_{01}, x_{02}, \ldots, x_{0N}$, positions of all agents asymptotically converge the average $x_0$ of the initial positions.

$$\lim_{k \to \infty} x_i[k] = x_0 = \frac{1}{N} \sum_{i=1}^{N} x_{0i} \quad (3)$$

### 2.3 CSMA-based transmission

Agents transmit and receive position information to and from adjacent agents by broadcast communication using CSMA/CA. Each agent sends its own position information every control period. Then, the control input is determined on the basis of the position information received within the control period, and the agent moves according to the control input.

The operation of CSMA/CA is as follows according to the specification of IEEE802.11DCF broadcast communication. First, carrier sensing is performed, and if a signal is not detected on a channel for a $DIFS$ duration, data is transmitted after a backoff time elapses. This backoff time is given by $Random \times \text{SlotTime}$, where $Random$ is an integer random number distributed uniformly in $[0, CW]$, and $CW$ is a contention window value. The backoff suspends when the channel is sensed busy and resumes only after the channel is sensed idle for $DIFS$ duration again. Packet collision occurs when agents with the same backoff time simultaneously transmit position information. In addition, packet collision may occur due to hidden terminal problem. Such packet collisions cause degradation of the control quality.
In order to reduce packet collision, we propose a method in which each agent estimates the importance of its own position information at each control period, and permits transmission of the position information with a probability according to the importance. It is expected that the packet collision rate is reduced by less important agents stopping transmission, and thus the control quality can be improved.

3.1 Priority of position information
The agent’s position information close to the consensus position is highly important and should be sent. However, each agent can only obtain position information from adjacent agents and can not know where it is located in the whole system. The closeness to the consensus position needs to be estimated. For this reason, each agent uses distance $d_i[k]$ between the center of gravity of the adjacent agents in the previous control period and its position as a measure of the closeness to the consensus position.

$$d_i[k] = x_i[k] - \frac{1}{|N_i[k]|} \sum_{j=1}^{N} a_{ij}[k-1]x_j[k-1]$$  \hspace{1cm} (4)

This takes advantage of the system dynamics of (1) and (2) and the fact that agents far from the consensus position often have a biased distribution of adjacent agents. As shown in Fig. 1, as $d_i[k]$ is smaller, the importance is higher because it is closer to the consensus position, and as $d_i[k]$ is larger, the importance is lower because it is farther from the consensus position.

3.2 Setting of transmission probability
The proposed method allows transmission with probability according to the measure $d_i[k]$ of the importance of position information. The transmission permission probability $P_i[k]$ is set as follows so that less important agents stop transmission of position information.

$$P_i[k] = 1 - \frac{d_i[k]}{R}$$  \hspace{1cm} (5)

This means that the transmission probability is set linearly according to $d_i[k]$ normalized with the the maximum value $R$. Within the control period, only agents
that are probabilistically permitted to transmit will transmit its own position information.

4 Numerical examples

4.1 Simulation setup

The effectiveness of the proposed method is evaluated by computer simulation. The simulation is set to control mobile agents like drones and parameters of IEEE802.11DCF are set with reference to [6]. The simulation parameters are shown in Table I. $N = 100$ agents are uniformly randomly arranged in an area of $300 \times 300$ m. Initial positions where the graph of network is disconnected and there is no possibility of achieving consensus are excluded beforehand. Performance evaluation has been conducted on several simulation parameters for the number of mobile agents, communication range; however, due to the limitation of space, here we show only numerical results where the effect of packet collision is remarkable.

The communication quality is evaluated by the average of packet collision rate. The quality of consensus control is evaluated by the average of the number of convergence agents. The number of convergence agents is defined as the number of agents located at a distance $\delta$ from each agent. In the evaluation, we compared the proposed method and the conventional method that permits transmission of all agents. In addition, in control quality evaluation, we also compared with the ideal performance where all agents in the communication range communicate without packet collision and therefore obtain all position information of adjacent agents.

4.2 Simulation results

Fig. 2(a) shows temporal changes of the packet collision rate. In the initial position, the packet collision rate is high because the number of hidden terminal agents is large. Since the number of agents in the communication range increases as the agents approach the consensus, the number of hidden terminal agents decreases, and thus the packet collision rate decreases. The proposed method can reduce the packet collision rate in the range from the initial position until all agents are within the communication range of each other. Packet collision occurs even in the area after all agents have entered each other’s communication range, but this is caused by setting the same backoff time and transmitting simultaneously.

Fig. 2(b) shows temporal changes of the number of convergence agents. As shown in Fig. 2(b), the proposed method converges faster than the conventional

<table>
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<tr>
<td>Control period ($T_s$)</td>
<td>20 [ms]</td>
<td>Data rate</td>
<td>11 [Mbps]</td>
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<td>Maximum speed limit ($v_{max}$)</td>
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<td>Slot Time</td>
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<td>The number of simulation</td>
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method. This is because the packet collision rate can be reduced and the number of agents that can communicate is increased. Further performance improvement is expected by a more appropriate function instead of the simple function \((5)\).

5 Conclusion

In this paper, for consensus control of multi-agent system, we propose a method to reduce the packet collision rate by estimating the importance of agent’s position information and setting the transmission probability on the basis of the estimated importance. By computer simulation, it was shown that by suppressing the transmission of less important agents, the packet collision rate can be reduced and the control quality can be improved.

Acknowledgments

The authors would like to thank Prof. Takaya Yamazato of Nagoya University for his variable suggestions. A part of this work is supported by JSPS KAKENHI Grant Number 18K04134.
Strongly secure ramp secret sharing with more participants based on Reed-Solomon codes

Ryutaroh Matsumoto

1 Department of Information and Communication Engineering, Nagoya University, Furo-cho, Chitane-ku, Nagoya, Aichi 464–8603, Japan
2 Department of Mathematical Sciences, Aalborg University, Denmark
a) ryutaroh.matsumoto@nagoya-u.jp

Abstract: The number of participants in the McEliece-Sarwate strongly secure ramp secret sharing scheme is at most \( q - L \), where \( q \) is the size of each share and \( L \) is the number of symbols in the secret. We propose another construction of strongly secure ramp secret sharing that can support \( q \) participants also based on the Reed-Solomon codes.

Keywords: ramp secret sharing, strong security, Reed-Solomon code

Classification: Fundamental Theories for Communications

References


*This research was partly funded by JSPS grant number 17K06419.
1 Introduction

Secret sharing is a scheme to share a secret among multiple participants so that only qualified sets of participants can reconstruct the secret, while forbidden sets have no information about the secret [1]. A piece of information received by a participant is called a share. A set of participants that is neither qualified nor forbidden is said to be intermediate. The access structure of a secret sharing scheme is the set of qualified sets, that of intermediate sets and that of forbidden sets.

It is well-known that the size of classical shares cannot be smaller than that of the classical secret in a perfect secret sharing scheme, where perfect means that there is no intermediate set, while ramp or non-perfect means that there exist intermediate sets [2, 3, 4]. An advantage of ramp schemes is that the size of secrets can be arbitrarily large for a fixed size of shares.

Ordinary ramp schemes have the following security risk: Suppose that classical secret is \( \bar{m} = (m_1, \ldots, m_L) \), and an intermediate set has \( \ell \) (\( \geq 1 \)) symbols of information about \( \bar{m} \). Then that intermediate set sometimes knows \( m_i \) explicitly for some \( i \). This insecurity was mentioned in [5, 4]. Iwamoto and Yamamoto [6] explicitly constructed such an example.

In order to address this security risk, Yamamoto [4] introduced the notion of strong security into ramp schemes: A secret sharing scheme with secret \( \bar{m} = (m_1, \ldots, m_L) \) is said to be strongly secure if any \((L - \ell)\) symbols in \( \bar{m} \) is always statistically independent of shares in an intermediate set that has \( \ell \) symbols of information about \( \bar{m} \), for \( \ell = 1, \ldots, L - 1 \).
The first ramp secret sharing scheme was proposed by McEliece and Sarwate [5]. It was based on the Reed-Solomon codes [7], and can support up to \( q - L \) participants, where \( q \) is the size of shares and \( L \) is the number of symbols in the secret. Much later the McEliece-Sarwate scheme was proved to be strongly secure [8]. Yamashita and Ogata [9] also proposed a strongly secure ramp secret sharing scheme that can support \( q - 1 \) participants with \( L = 2 \). Martínez-Peñas [10] studied the communication efficiency and the strong security simultaneously.

Often we can increase the size of shares to support more participants. However, if we cannot increase the size of secrets, the storage space of shares are wasted more. Thus, it is desirable to have another scheme that can support more participants.

The purpose of this short paper is to provide another construction of strongly secure ramp secret sharing schemes with more participants. After reviewing relevant definitions of secret sharing in Section 2, we will propose our new ramp secret sharing and will prove its strong security in Section 3.

### 2 Preliminaries

Let \( \mathbb{F}_q \) be the finite field with \( q \) elements. In this paper we assume that each share belongs to \( \mathbb{F}_q \).

**Definition 1** [2, 4] A \((k, L, n)\)-threshold ramp secret sharing scheme distributes a secret in \( \mathbb{F}_q^L \) to \( n \) participants. Each share is one symbol in \( \mathbb{F}_q \). \( k \) or more participants can reconstruct the secret, while \( k - L \) or less participants have no information about the secret. By “no information” we mean the statistical independence between the secret and a set of shares.

**Definition 2** [4] Assume that the probability distribution of secrets is uniform. A \((k, L, n)\)-threshold ramp secret sharing scheme is said to be strongly secure, if any \( L - \ell \) symbols in the secret and any set of \( k - L + \ell \) shares are statistically independent of each other for \( \ell = 1, \ldots, L - 1 \).

Iwamoto and Yamamoto [6] generalized Definition 2, and the generalized definition was mentioned in the introduction. The McEliece-Sarwate secret sharing [5] is a strongly secure \((k, L, n)\)-threshold scheme.

### 3 Proposed construction and its strong security

#### 3.1 Proposed construction

Let \( n \leq q \) and \( a_1, \ldots, a_n \) be distinct elements in \( \mathbb{F}_q \). We assume that \( a_1, \ldots, a_L \) are nonzero. We will construct a strongly secure \((k, L, n)\)-threshold scheme, with \( n \leq q \) and

\[
  k \geq 2L. 
\]

Define an \([n, k]\) Reed-Solomon (RS) code as

\[
  \text{RS}(n, k) = \{ (f(a_1), \ldots, f(a_n)) : f(x) \in \mathbb{F}_q[x], \deg f(x) < k \}. 
\]

Hereafter we assume that secrets are uniformly distributed in \( \mathbb{F}_q^L \). For a given secret \( \tilde{m} = (m_1, \ldots, m_L) \in \mathbb{F}_q^L \), find \( g_1(x) = a_0x^{q0} + \cdots + a_{L-1}x^{qL-1} \) and such that \( g_1(a_j) = m_j \).
When we view generalized Reed-Solomon code. Define 

\[ g_1(x) = \frac{m_j}{x^{k-L}} \] 

for all \( j = 1, \ldots, L \). Such \( g_1(x) \) always exists because computation of \( g_1(x) \) is just the inverse mapping of the encoding of \( \text{RS}(L, L) \) for the codeword \( (m_1/x^{k-L}, \ldots, m_L/x^{k-L}) \). Let 

\[ g_2(x) = x^{k-L}g_1(x), \] 

Observe that \( g_2(a_j) = m_j \).

Randomly choose \( b_0, \ldots, b_{k-L} \in \mathbb{F}_q \) and let 

\[ g_3(x) = g_2(x) + b_0 + b_1x + \cdots + b_{k-L-1}x^{k-L-1}. \]

The dealer sends \( g_3(a_i) \) as a share to the \( j \)-th participant, for \( j = 1, \ldots, n \).

Let \( \tilde{x}_1, \tilde{x}_2 \in \mathbb{F}_q^n \) be two vectors of \( n \) shares, and assume that \( \tilde{x}_i \) corresponds to a secret \( \tilde{m}_i \in \mathbb{F}_q^n \) for \( i = 1, 2 \). A secret sharing scheme is said to be linear if the linearly combined share vector \( \beta_1\tilde{x}_1 + \beta_2\tilde{x}_2 \) corresponds to the linearly combined secret \( \beta_1\tilde{m}_1 + \beta_2\tilde{m}_2 \). It is known that any linear secret sharing scheme can be expressed by a nested pair of linear codes \( \mathcal{C}_2 \subset \mathcal{C}_1 \subset \mathbb{F}_q^n \) with \( \dim \mathcal{C}_1 - \dim \mathcal{C}_2 = L \) \([11, 12, 13, 14, 15]\). In our proposed scheme we have \( \mathcal{C}_2 = \text{RS}(n, k - L) \) and \( \mathcal{C}_1 = \text{RS}(n, k) \).

The coset distance of \( \mathcal{C}_1 \supset \mathcal{C}_2 \) is defined as \([13]\) 

\[ d(\mathcal{C}_1, \mathcal{C}_2) = \min \{ \text{wt}(\tilde{x}) \mid \tilde{x} \in \mathcal{C}_1 \setminus \mathcal{C}_2 \}, \]

where \( \text{wt}(\tilde{x}) \) is the Hamming weight of \( \tilde{x} \). It was shown \([11, 12, 13, 14]\) that any \( n + 1 - d(\mathcal{C}_1, \mathcal{C}_2) \) shares can reconstruct the secret and that any \( d(\mathcal{C}_2^\perp, \mathcal{C}_1^\perp) - 1 \) shares are statistically independent of the secret, where \( \mathcal{C}_1^\perp \) is the dual code of \( \mathcal{C}_1 \).

Since \( d(\text{RS}(n, k), \text{RS}(n, k - L)) = n - k + 1 \) and \( d(\text{RS}(n, k - L), \text{RS}(n, k)) = k - L + 1 \), we know that the proposed ramp scheme is a \((k, L, n)\)-threshold scheme.

### 3.2 Strong security

Our remaining task is to examine the strong security of the proposed scheme. This subsection is devoted to a proof of its strong security. Without loss of generality we can consider the statistical independence between \( m_1, \ldots, m_{L-\ell} \) and a set of \( k - L + \ell \) shares.

In our proposed scheme, \( b_0, \ldots, b_{L-1} \) serve as dummy randomness hiding \( \tilde{m} \). When we consider the secrecy of \( m_1, \ldots, m_{L-\ell} \), the rest \( m_{L-\ell+1}, \ldots, m_L \) of the secret \( \tilde{m} \) also serves as dummy randomness hiding \( m_1, \ldots, m_{L-\ell} \).

For 

\[ g(x) = b_0x^0 + \cdots + b_{L-\ell-1}x^{k-L+\ell-1}, \]

define 

\[ \tilde{g}(x) = b_{k-L+\ell}x^{k-L+\ell} + \cdots + b_{k-1}x^{k-1} \]

such that 

\[ \tilde{g}(a_j) = -\sum_{i=k-L}^{k-1} b_ia_i \]

for \( j = 1, \ldots, L - \ell \). Such a \( \tilde{g}(x) \) is uniquely determined because it is the inverse of encoding of the \([L - \ell, L - \ell]\) generalized Reed-Solomon code. Define a linear code

\[ D = \left\{ (g(a_1) + \tilde{g}(a_1), \ldots, g(a_n) + \tilde{g}(a_n)) : \deg g(x) \leq k - L + \ell - 1 \right\}. \]

When we view \( m_1, \ldots, m_{L-\ell} \) as the secret and the rest \( m_{L-\ell+1}, \ldots, m_L \) as dummy randomness, the secret sharing scheme can be described by the nested pair of linear codes \( D \subset \mathcal{C}_1 \), where \( \mathcal{C}_1 = \text{RS}(n, k) \) as defined before.

For a subset \( S \subset \mathbb{F}_q^n \) and \( A \subset \{1, \ldots, n\} \), we mean 

\[ P_A(S) = \{ (x_i)_{i \in A} : (x_1, \ldots, x_n) \in S \}. \]
Lemma 3

\[
\dim P_A(\text{RS}(n, k)) - \dim P_A(D) = \begin{cases} 
0 & \text{if } 0 \leq |A| \leq k - L + \ell, \\
|A| - k - L + \ell & \text{if } k - L + \ell \leq |A| \leq k, \\
L - \ell & \text{if } k \leq |A| \leq n.
\end{cases}
\] (2)

Proof: Since the minimum Hamming distance of \(\text{RS}(n, k)\) is \(n - k + 1\), we have [7]

\[
\dim P_A(\text{RS}(n, k)) = \begin{cases} 
|A| & \text{if } 0 \leq |A| \leq k, \\
L - \ell & \text{if } k \leq |A| \leq n.
\end{cases}
\] (3)

The codeword in \(D\) is the sum of a codeword in \(\text{RS}(n, k - L + \ell)\) and the codeword defined by \(\tilde{g}(x)\). The latter can be seen as a codeword in a generalized Reed-Solomon code of length \(n\) and dimension \(L - \ell\). So, the Hamming weight of a codeword defined by \(\tilde{g}(x)\) is \(\geq n + 1 - L + \ell\). There exists a codeword in \(\text{RS}(n, k - L + \ell)\) of Hamming weight \(n - k + L - \ell + 1\). Since we have assumed \(k \geq 2L\) in (1) and \(\ell \geq 1\), we always have \(n - k + L - \ell + 1 < n + 1 - L + \ell\). Under this condition, the minimum weight codeword in \(\text{RS}(n, k - L + \ell)\) cannot be canceled by a codeword defined by \(\tilde{g}(x)\). Therefore, the minimum Hamming distance of \(D\) is \(n - k + L - \ell + 1\), which implies [7]

\[
\dim P_A(D) = \begin{cases} 
|A| & \text{if } 0 \leq |A| \leq k - L + \ell, \\
k - L + \ell & \text{if } k - L + \ell \leq |A| \leq n.
\end{cases}
\] (4)

Combining Eqs. (3) and (4) gives the claim of this lemma. \(\square\)

The mutual information between \(m_1, \ldots, m_{L-\ell}\) and the shares in \(A\) is [15, Eq. (16)]

\[
\dim P_A(C_1) - P_A(D).
\] (5)

By Lemma 3, \(|A| \leq k - L + \ell\) implies that (5) is zero, which proves the strong security of the proposed ramp secret sharing scheme.