Evaluation of Division-based Broadcasting System over Wireless LAN

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Summary. Due to the recent popularization of the Internet, continuous media data broadcasting such as audio or video have attracted great attention. Although continuous media data broadcasting can reduce the more necessary bandwidth than conventional on-demand delivery, waiting time from requesting data to playing it increases. In division-based broadcasting, many researches have proposed scheduling methods to reduce the waiting time. Although a division-based broadcasting system has also been proposed to evaluate these methods, this system does not consider wireless LAN environment. In wireless LAN environment, we have to consider the such problems as a communication protocol and a packet loss. In this paper, we implement and evaluate a division-based broadcasting system to solve the problem in the conventional system. In our proposed system, we confirm the effectiveness for reducing the waiting time using the scheduling method.

1 Introduction

Due to the recent popularization of the Internet, continuous media data broadcasting such as audio or video have attracted great attention [1]. In on-demand delivery, available bandwidth increases in proportion to the number of clients. On the other hand, in broadcasting, the server can deliver the data to many clients with a constant available bandwidth. However, waiting time from requesting data to playing it increases.

In division-based broadcasting, many studies employ the scheduling methods, which reduces waiting time by dividing the data into several segments and frequently broadcasting the precedent segments. These scheduling methods make the broadcast schedule considering the situation in actual network environments. Since most scheduling methods in broadcasting evaluate the waiting time in simulation environments, we need to evaluate the waiting time using the scheduling method in network environments.

We have been proposed a division-based broadcasting system [2], which can evaluate scheduling methods in network environments. Conventional division-based broadcasting systems have implemented the telecommunication protocols based on the wired LAN, and evaluate the scheduling methods. However, since the number of users using a smart phone and a computer increases, many clients connect to the Internet over wireless LAN. Since the bandwidth variation in wireless LAN is larger than that in wired LAN, when clients watch the

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movie in actual network, the packet loss occurs. Therefore, we need to make the broadcasting system considering the case where clients watch the movie over wireless LAN.

In this paper, we propose the solution for this problem and implement the division-based broadcasting system in wireless LAN. In addition, we evaluate our proposed system and confirm the effectiveness for reducing the waiting time using conventional scheduling method.

The remainder of the paper is organized as follows. We explain the division-based broadcasting system in Section 2. Related works are introduced in Section 3. Our proposed system is designed in Section 4. Next, we implement and evaluate our proposed system in Sections 5 and 6. Finally, we conclude the paper in Section 7.

2 Division-based Broadcasting System

2.1 VoD and Broadcasting Systems

In webcasts, there are mainly two types of delivery systems: VoD and broadcasting. In the VoD system, the server starts delivering data sequentially based on client requests. Waiting times under VoD systems are roughly equal to receiving times. When the server repetitively broadcasts continuous media data, clients have to wait until the first portion of the data is broadcast.

When the server broadcasts the continuous media data repetitively, clients have to wait until the first portion of the data is broadcast. To reduce the waiting time, many methods employ the division-based broadcasting technique, which reduces the waiting time by dividing the data into several segments and broadcasting precedent segments frequently.

2.2 Waiting Time in Division-based Broadcasting

In the division-based broadcasting technique, since the data is divided into several segments, it is important to schedule segments without interrupting clients' continuous play. An example of division-based broadcasting technique is shown in Figure 1. The example uses the Fast Broadcasting (FB) [3] method to explain the problem caused by heterogeneous clients easily. In the FB method, the broadcast bandwidth is divided into several channels. Bandwidth for each channel is equivalent to the consumption rate. In this case, the server uses three channels. Also, the data is divided into three segments, S_1, S_2, and S_3. When the total playing time is 7 min., the playing time of S_1 is calculated to be 1 min., S_2 is 2 min., and S_3 is 4 min. The consumption rate is 5.0 Mbps and the available bandwidths for clients are 15 Mbps. Bandwidth for each channel is $15/3 = 5.0$ Mbps, which is the same as the data consumption rate. The server broadcasts S_i $(i = 1,2,3)$ via broadcast channel C_i repetitively. Clients can store broadcasted segments into their buffers while playing the data and can play each segment after receiving them. In this case, since clients can receive broadcasted segments from their midstream, the waiting time is the same as the time needed to receive only S_1 and the average waiting time is 1 min.

3 Related Works

In broadcasting systems, several scheduling methods to reduce waiting time have been proposed. In these methods, by dividing the data into several segments and producing a proper broadcast schedule, waiting time can be reduced.

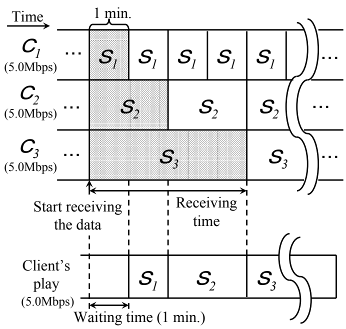

Fig. 1. Example of division-based broadcasting situation without interruption.

Fig. 2. Screenshot in *TeleCaS*.

In the Optimized Periodic Broadcast (OPB) [4], each data is separated into two parts. The server uses several broadcast channels and broadcasts each segment on each channel. When clients complete receiving the precedent parts of the content, they start receiving the rest portions of the data. Since clients can get the sub-segment data in advance, waiting time can be reduced. However, the bandwidth increases as the number of contents increases.

In the Hierarchical Stream Merging (HSM) [5], when the server broadcasts separated data via each channel repetitively, if clients complete receiving the data, the server merges the channel which the client receives. Since the method can reduce the necessary bandwidth, waiting time is reduced under the same bandwidth.

In BroadCatch [6], the server divides the data into 2^{K-1} segments of equal sizes. The server broadcasts them periodically using K channels. The bandwidth for each channel is the same as the data consumption rate. By adjusting K according to the available bandwidth for clients, waiting time is reduced effectively.

In the Layered Internet Video Engineering (LIVE) [7], clients feed back virtual congestion information to the server to assist both bandwidth sharing and transient loss protection. LIVE can minimize the total distortion of all participating video streams and maximize their overall quality. Also, several researches study the reliable broadcasting in wireless network with packet loss probability [8,9].

In the Heterogeneous Receiver-Oriented Broadcasting (HeRO) [10], the server divides the data into K segments. Let J be the data size for the first segment. The data sizes for segments are J, $2J$, $2^2J, \ldots, 2^{K-1}J$. The server broadcasts them using K channels. However, the data size of the K^{th} channel becomes a half of the data, clients may have waiting time with interruptions.

In the BE-AHB method [11], since the data are divided into several segments, segments must be scheduled without interrupting clients' continuous play.

4 Design

4.1 *TeleCaS*

As explained in Section 3, many scheduling methods have been proposed to reduce waiting time in division-based broadcasting. However, most scheduling methods calculate the waiting

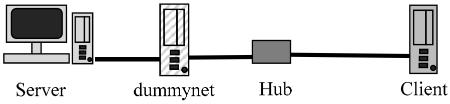

Fig. 3. Network environment for wired LAN in conventional *TeleCaS*.

time by computational simulation. To evaluate the scheduling method in actual network, we have proposed a division-based broadcasting system Telecommunication and BroadCasting System (*TeleCaS*) [2].

The screenshot is shown in Figure 2. We designed our proposed system using the C programming language and Java. Our proposed system can operate with several types of operating systems including the Windows system. In addition, we implemented the function that the server can operate the process by Graphical User Interface (GUI) to set the delivery of several segments easily. This system can make the cooperation of C language and Java by Java Native Interface (JNI). Clients play the data using the media player Totem [12].

4.2 Application to Wireless LAN Environment

Since the scheduling methods described in Section 2 suppose a situation where processing load and packet loss are not considered, they fail to consider the effect on division-based broadcasting systems. To evaluate scheduling methods in actual networks, we proposed a division-based broadcasting system called *TeleCaS*, which introduces several types of conventional broadcasting methods and designs Internet broadcasting systems based on a network environment with a server and clients.

Conventional *TeleCaS* constructs the wired LAN environment by connecting the server and clients. In Figure 3, the connection between the server and the client is routed through Dummynet that controls the bandwidth. With Dummynet, we can construct an evaluation environment to reproduce various network environments.

Due to the recent popularization of smart phones and compact notebooks, many clients can connect to the wireless LAN environment. Therefore, we need to modify the mechanism for wireless LAN environment in *TeleCaS*.

4.3 Packet Loss in Wireless LAN Environment

In division-based broadcasting system, the rate of packet loss in the wireless LAN is more increased than that in the wired LAN. Conventional *TeleCaS* has evaluated scheduling methods in the case of where the packet loss is not occurred. However, since the packet loss in the wireless LAN effects the waiting and interruption times, we modify the protocol considering the packet loss and implement the *TeleCaS* in the wireless LAN environment.

There are two types of packet loss, one is the packet loss for delivering the data by not delivering data to clients and damaging to the packet due to noise. When the client can not receive the packet from the server, or the packet that is received from the server is damaged, clients require a retransmission of the packet to the server. In the case of multicast with User Datagram Protocol (UDP), clients can not require a retransmission of the packet to the server. In wireless LAN environment, the rate of packet loss that is opportunity occurred is less than 1.0%. The packet loss is occurred by three reasons, which are the failure of the communication

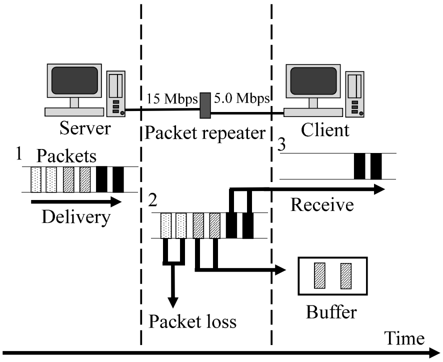

Fig. 4. Packet loss for computing environment.

path in a wired LAN environment, the interference from other radio waves in a wireless LAN environment, and a lack of data due to not reach the client.

The other is the packet loss for computing environment, which is the available bandwidth and the availability of the computer between the server and clients. In broadcasting system with wireless repeater, when the difference between the available bandwidth of the server and that of the client is large, the packet that can not be stored in the buffer area is discarded.

The flow of discarded packets that is not stored in the buffer area is shown in Figure 4. In Figure 4, the server delivers six packets, and the client makes the processing of playing data using two packets. When the rest of packets is four and the buffer size of clients is only two packets, although they can store two packets in four packets in their buffer, the rest of two packets is discarded. In on-demand delivery, clients can receive these packets again by requiring a retransmission of the packet to the server. On the other hand, in broadcasting, the server does not retransmit the packet based on the UDP. Therefore, in the communication protocol for *TeleCaS*, we need to add the mechanism about requiring the retransmission of the packet.

4.4 Rate of Packet Loss

In communication with the client and the radio base station, when the distance from the base station to the client becomes long, the packet loss may occur. In addition, when the client performs a wireless communication by Wi-Fi standard, the packet loss occurs due to the radio interference. Therefore, in the wireless LAN, we need to evaluate the rate of packet loss considering the system environments.

5 Implementation

5.1 *TeleCaS* over Wireless LAN

We implement the *TeleCaS* over the wireless LAN environment. To evaluate the *TeleCaS*, we uses the ethernet converter connected to the client machine via LAN cable. In addition, we use

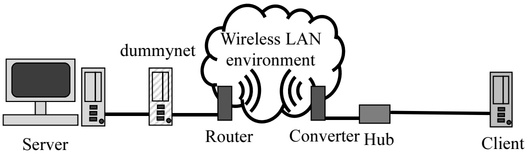

Fig. 5. Wireless LAN environment in *TeleCaS* (dummynet on server side).

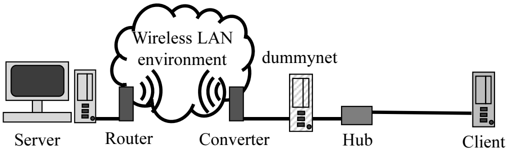

Fig. 6. Wireless LAN environment in *TeleCaS* (dummynet on client side).

the router for the wireless LAN environment The data format of *TeleCaS* in wireless LAN is the same as that in wired LAN.

5.2 Evaluation Items in *TeleCaS*

In *TeleCaS*, the server uses the IP multicast, the communication protocol uses UDP/IP. the server delivers data using UDP that is a connectionless protocol. Since clients have to wait until the data that their packets are lost are broadcast again, the interruption time occurs during playing the data.

We plan three types of evaluation. Firstly, to evaluate the influence of the available bandwidth, we calculate the rate of packet loss under different available bandwidths. Secondly, we evaluate the influence of data size based on the rate of packet loss. Finally, to evaluate the influence of the number of channels, we calculate the rate of packet loss under different available bandwidths.

5.3 Wireless LAN Environment in *TeleCaS*

We explain the structure of wireless LAN environment in *TeleCaS*. To control the available bandwidth, there are two cases in the location of dummynet that has a connection between the server and clients. One is a case of connecting the dummynet on the server side, which is shown in Figure 5. The other is a case of connecting the dummynet on the client side, which is shown in Figure 6. In division-based broadcasting system, the server can reduce waiting time by setting the broadcast scheduling based on the available bandwidth. When the clients receive data in *TeleCaS*, the server sets the available bandwidth. Therefore, we connect the dummynet on the server side.

In this paper, we evaluate the rate of packet loss in *TeleCaS*. The server delivers data to one client in wireless LAN environment.

Table 1. Measurement environment.

Server	CPU	Intel®Core 2 Duo E7500 (2.93 GHz)
	Memory	2.0 GBytes
	OS	Ubuntu 12.10
	NIC	RTL810E / RTL8102E
Client	CPU	Intel®Core 2 Duo E7500 (2.93 GHz)
	Memory	2.0 GBytes
	OS	Ubuntu 12.10
	NIC	RTL810E / RTL8102E
dummynet	CPU	Intel®Core 2 Duo E7500 (2.93 GHz)
	Memory	2.0 GBytes
	OS	FreeBSD 8.2-RELEASE
	NIC1	RTL8169SC
	NIC2	RTL8169SC
Router	Model	CG-WLBARGPXW
	Model	CG-WLBARGPXW
	Standard	IEEE802.11g
	Speed	Max 54 Mbps
Converter	Model	Aterm WG1200HP
	Model	PA-WG1200HP
	Standard	IEEE802.11g
	Speed	Max 867 Mbps

6 Evaluation

6.1 Evaluation Environment

We evaluate the performance in both actual and simulation environments. We use the FB method to evaluate the effectiveness of packet loss in wireless LAN environment, as explained in Subsection 4.3. The performances of machines, wireless LAN router, and Ethernet converter are shown in Table 1. We construct the evaluation environment based on IEEE 802.11g. To evaluate the packet loss in wireless LAN environment, we use the wireless LAN router that has a low spec. In addition, we use a converter for setting the desktop computer as a client in wireless LAN environment.

6.2 Effect of Bandwidth

In wireless LAN environment, we evaluate the interruption time and the rate of packet loss under different available bandwidth. The number of channels is 4, the playing time is 2 min., the data size is 21.4 Mbytes, and the consumption rate is 1.5 Mbps. The interruption time is the average value measured five times. The rate of packet loss in wireless LAN environment is about 1%, and we use a 2.4 Ghz frequently band.

The result of interruption time is shown in Figure 7. The vertical axis is the average interruption time. The horizontal axis is the available bandwidth. In Figure 7, in both wired and wireless LAN environments, interruption time is lengthened according to the available

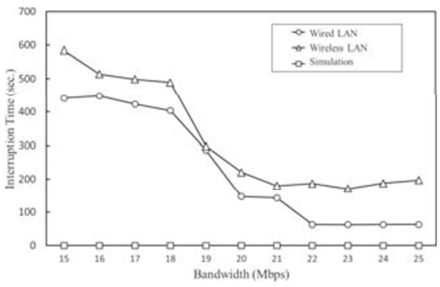

Fig. 7. Bandwidth and interruption time.

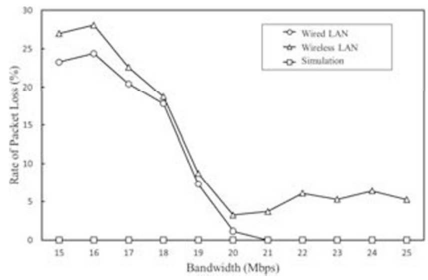

Fig. 8. Bandwidth and rate of packet loss.

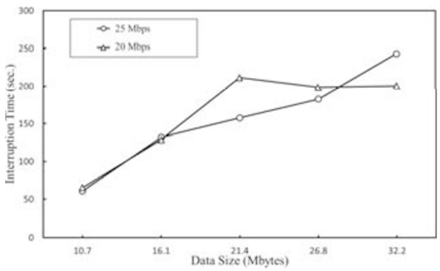

Fig. 9. Data size and interruption time.

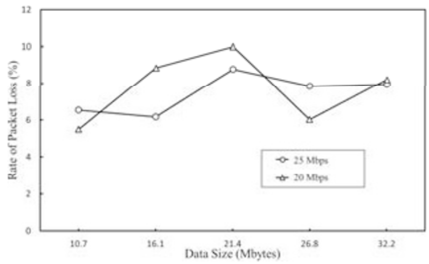

Fig. 10. Data size and rate of packet loss.

bandwidth. When the available bandwidth is less than the necessary bandwidth of server, interruption time occurs by the packet loss while delivering data.

The result of rate of packet loss is shown in Figure 8. The vertical axis is the rate of packet loss. The horizontal axis is the available bandwidth. In Figure 8, when the available bandwidth is more than 22 Mbps, the rate of packet loss is 0% in wired LAN environment, and that is about 5% in wireless LAN environment. In wireless LAN environment, the packet loss is occurred in only the case of where the server delivers data to clients.

6.3 Effect of Data Size

We evaluate the interruption time and the rate of packet loss under different data size. The number of channels is 4 and the consumption rate is 1.5 Mbps. There are five types of data size: 10.7 MBytes, 16.1 MBytes, 21.4 MBytes, 26.8 MBytes, and 32.2 MBytes. Therefore, the playing time of each data size is 60 sec., 90 sec., 120 sec., 150 sec., and 180 sec., respectively. The available bandwidths of server are 25 Mbps and 20 Mbps. The interruption time is the average value measured five times. The rate of packet loss in wireless LAN environment is about 1%.

The result of interruption time is shown in Figure 9. The vertical axis is the average interruption time. The horizontal axis is the available bandwidth. When the data size is 10.7 MBytes or 16.1 Mbytes, since the packet loss is not occurred, interruption time is the same. When the available bandwidth is 20 Mbps and the data size is 21.4 MBytes, interruption time

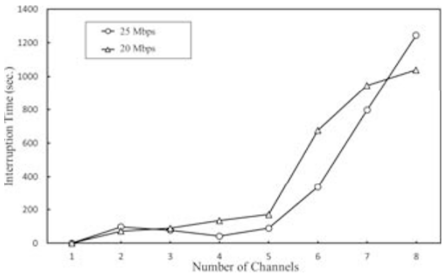

Fig. 11. Number of channels and interruption time.

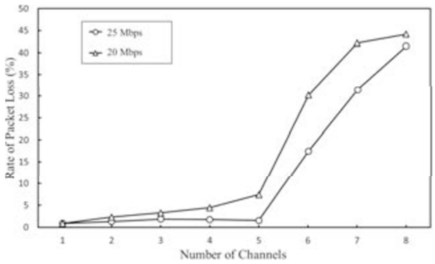

Fig. 12. Number of channels and rate of packet loss.

is greatly lengthened by occurring the packet loss. On the other hand, when the available bandwidth is 20 Mbps and the data size is 26.8 MBytes or 32.2 MBytes, since the rate of packet loss is not changed while delivering data, interruption time is not lengthened. In actual network, since the rate of packet loss is changed according to the data size, the interruption time is also lengthened. Next, when the data size is more than 32.2 MBytes, since the rate of packet loss is changed while delivering data, the interruption time is lengthened as increasing of the available bandwidth.

The result of rate of packet loss is shown in Figure 10. The vertical axis is the rate of packet loss. The horizontal axis is the available bandwidth. The rate of packet loss is from 5 to 10 % in every cases. On the other hand, since the sum of packet loss increases as increasing of the data size, the interruption time is lengthened.

6.4 Effect of Number of Channels

We evaluate the interruption time and the rate of packet loss under different number of channels. The playing time is 120 sec., the data size is 21.4 Mbytes, and the consumption rate is 1.5 Mbps. The available bandwidth of the server is 25 Mbps or 20 Mbps. The interruption time is the average value measured five times. The rate of packet loss in wireless LAN environment is about 1%, and we use a 2.4 Ghz frequently band.

The result of interruption time is shown in Figure 11. The vertical axis is the average interruption time. The horizontal axis is the available bandwidth. In addition, the result of rate of packet loss is shown in Figure 12. The vertical axis is the rate of packet loss. The horizontal axis is the available bandwidth. In Figure 11, interruption time is lengthened as the increasing of the number of channels. In Figure 12, the rate of packet loss becomes high as the increasing of the number of channels. When the number of channels is 2, since the packet loss is not occurred, the rate of packet loss is closely the same in both cases of where the available bandwidth is 25 Mbps or 20 Mbps. When the available bandwidth is 25 Mbps and the number of channels is less than 4, since the packet loss is not occurred, the packet loss is not changed. However, when the number of channels is more than 5, since the packet loss is occurred, the interruption time becomes lengthened.

7 Conclusion

In this paper, we proposed the division-based broadcasting system to evaluate the effectiveness using conventional scheduling method. In division-based broadcasting, many researches have proposed scheduling methods to reduce the waiting time. Although we have proposed the conventional division-based broadcasting system evaluate these methods, this system does not consider wireless LAN environment. In wireless LAN environment, we proposed the solution for the situation which the packet loss is occurred while delivering data. In addition, we implemented the system design and improved the conventional broadcasting system for wireless LAN environment. In our evaluation, we confirmed the effectiveness for reducing the waiting time using the scheduling method.

In the future, we will realize the system that reduces the communication traffic and the processing load of the server.

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