Towards a Fairness Multimedia Transmission Using Layered-Based Multicast Protocol

Heru SUKOCO^{†,††a)}, Yoshiaki HORI^{†b)}, Members, Hendrawan^{†††c)}, Nonmember,
and Kouichi SAKURAI^{†d)}, Member

SUMMARY The distribution of streaming multicast and real time audio/video applications in the Internet has been quickly increased in the Internet. Commonly, these applications rarely use congestion control and do not fairly share provided network capacity with TCP-based applications such as HTTP, FTP and emails. Therefore, Internet communities will be threatened by the increase of non-TCP-based applications that likely cause a significant increase of traffics congestion and starvation. This paper proposes a set of mechanisms, such as providing various data rates, background traffics, and various scenarios, to act friendly with TCP when sending multicast traffics. By using 8 scenarios of simulations, we use 6 layered multicast transmissions with background traffic Pareto with the shape factor 1.5 to evaluate performance metrics such as throughput, delay/latency, jitter, TCP friendliness, packet loss ratio, and convergence time. Our study shows that non TCP traffics behave fairly and respectful of the co-existent TCP-based applications that run on shared link transmissions even with background traffic. Another result shows that the simulation has low values on throughput, vary in jitter (0-10 ms), and packet loss ratio > 3%. It was also difficult to reach convergence time quickly when involving only non TCP traffics.

key words: TCP friendly, Receiver-driven Layered Multicast, network simulator, Pareto, quality of service, jitter, throughput, packet loss ratio, fairness, convergence time

1. Introduction

1.1 Background

The usage of Internet development has been increased rapidly including on the number of users, infrastructures, and applications. Thus, no wonder if there are so many new trend applications developing in the Internet, especially UDP-based multimedia streaming applications such as video services and broadcasting services. Media streaming has been popular and widely used by Internet communities due to its capability to transfer from one server to many clients, called IP multicast [1].

The earliest services for video distribution over the Internet combined multi-rate or layered traffic flows with multicast transmission. This mechanism offers the ability to satisfy users with many quality levels [2]. The increase of these services could raise the Internet data traffics that can trigger a congestion which is the most common issue when they converge with a bottleneck link. Congestion and bottleneck may cause packet loss, low throughput, high queuing delay and even cause a network collapse. Those are because of error control mechanisms that retransmit more packets to networks. Another major issue related to real time transmission is the TCP-friendly behavior of the underlying transport protocols [3].

TCP-based applications use traffic flow control mechanisms that can respond to congestion by increasing or decreasing congestion window size. The mechanisms in [4]–[6] are known as slow-start and additive-increase and multiplicative-decrease (AIMD). Typically, the Internet protocol and architecture have been designed for sending only data packet, not the real time packet. Due to sophistication and rapid development of Internet, it could be used for real time packet such as audio, still images, and videos delivered by multimedia applications [4], [7]. The use of these applications dramatically may cause flooding in the Internet due to they have no congestion control, fairness mechanisms while occupying the networks, and bandwidth consumers. If this happened, TCP-based applications will lead to congestion collapse and experience a starvation so that users could not surf the websites and send emails.

1.2 Motivation

Some of the proposed TCP-friendly congestion controls have been introduced in the Internet both window-based schemes [7]–[9] and rate/equation-based schemes [7], [8], [10], [11]. The main objectives of these schemes are to reduce variations in transmission rate [12] and to control UDP-based multimedia streaming applications that interact well with TCP [6], [13]. The proposed protocols should maintain their fairness and aggressiveness in the networks across connections to get TCP-fairness ratio (F) close to 1 ($F \approx 1$) [7], [14].

Fairness is an important aspect of any transport protocol so that traffic is equally shared and distributed between applications in a network [3], [7]. $F \gg 1$ means that non TCP-based protocols are very dominant and unfair with

Manuscript received March 6, 2010.

Manuscript revised June 20, 2010.

[†]The authors are with the Department of Informatics, Graduate School of Information Science and Electrical Engineering, Kyushu University, Fukuoka-shi, 819-0395 Japan.

^{††}The author is with the Department of Computer Science, Faculty of Mathematics and Natural Sciences, Bogor Agricultural University, Indonesia.

^{†††}The author is with School of Electrical Engineering and Informatics, Bandung Institute of Technology, Indonesia.

a) E-mail: heru@itslab.csce.kyushu-u.ac.jp, hsrkom@ipb.ac.id

b) E-mail: hori@inf.kyushu-u.ac.jp

c) E-mail: hend@stei.itb.ac.id

d) E-mail: sakurai@inf.kyushu-u.ac.jp

DOI: 10.1587/transinf.E93.D.2953

TCP traffics on the networks, which can lead to collapse, whereas $F \ll 1$ means that non TCP-based protocols are ineffective to use in the networks.

This paper will examine the performance evaluation of the layered traffic control in [15] with given Internet traffics such as HTTP, FTP and constant bit rate (CBR). Our research evaluates the basic concept of TCP fairness by simulating the protocol for unicast and multicast flows competing between TCP and UDP protocol with background traffics and DropTail queue management. We use RLM protocol for delivering multicast and layered encoding transmission and evaluate the performance metrics such as throughput, delay/latency, jitter, TCP friendliness (F), Packet Loss Ratio (PLR), and convergence time (CT) when it competes with background traffics.

1.3 Related Works

The study of the layered video transmission [16]–[19] was pioneered by Receiver-driven Layered Multicast (RLM) [15] protocol. Multimedia streaming applications have become a new trend in delivering information such as news, lecture materials, advertisement, songs, and movies. Most of them are UDP-based traffic [12] that could lead to congestion collapse and TCP starvation in the IP-based networks such as Internet. Some of the multimedia streaming flows are uncontrolled traffics and need a smoothness variation in transmission rate.

Jacobson in [4] introduced AIMD protocol for TCP traffic and other proposed protocols for controlling non TCP-based traffics have been also introduced including TCP-friendly schemes classified in [7]. Widmer, Denda, and Mauve in [7] classified by single/multi rate and rate-based/window-based schemes as follows (1) RAP, LDA+, TFRCP, TFRC, LTRC, TRAM, and TEAR are single rate and rate-based schemes; (2) MTCP, NCA, and PGMCC are single rate and window-based schemes; (3) RLC, FLID-DL, TopoSense, LVMR, LTS, TFRP, MLDA, and PLM are multi rate and rate-based schemes; and (4) Rainbow is multi rate and window-based schemes.

Vicisano, et.al and Wang, et.al in [20], [21] describe a typical approach to multi rate congestion control using layered multicast, which is a sender should divide the data transmission into several layers and transmits them to different multicast groups. Each receiver can individually select to join as many groups as permitted by the bandwidth bottleneck between that receiver and the sender. The previous research explored and improved the properties of RLM, but neither of them exploited fairness value nor performance evaluation with background traffics such as HTTP and Pareto.

1.4 Problem Statement

Layered traffic is suitable for a video transmission to improve the video quality and layered multicast. Congestion control is performed indirectly by the group management

and routing mechanisms of the underlying multicast protocol. Receiver-driven congestion control is usually used together with layered congestion control approaches that the receivers decide whether to subscribe or to unsubscribe from additional layers, based on the congestion situation of the network.

1.5 Contribution

Our research proposed 8 scenarios, as discussed in Sect. 2.1, to act friendly between both TCP applications and multicast applications. Those scenarios use background traffics [22], [23] and DropTail queue management, which is used by a typical router for managing incoming and outgoing packets in the buffers [24]. In addition, we proposed a set of mechanisms such as a simple class for creating a static wired network topology that can be used in NS2 simulation, providing various data rates with background traffics, a simple class of creating static wired network topology, various scenarios to represent the real internet environment. Finally, we evaluated RLM performance in various conditions and data rates by evaluating some performance metrics such as fairness, throughput, jitter, convergence time and PLR when competing with each other in a network.

2. Simulation Model

We first describe the topologies used by this research and evaluate some performance metrics. We denote a scenario by $S_{N \in \text{integer}} : L_{i,j} \{(d, b) \mid d \in D, b \in B\}$ where S_N , $L_{i,j}$, D , and B refer to a set of scenarios, bidirectional links, delay (in ms), and bandwidth (in mbps), respectively. We use subscript N and i, j to refer n -th simulation and a link from i -th node to j -th node, respectively, unless otherwise specified.

2.1 Simulation Models and Properties

Simulation model comprises 8 scenarios to evaluate the behaviors and traffic patterns both TCP and non-TCP applications in unicast and multicast data streams. The first three scenarios of the simulation, $S_{1,2,3}$, describe the basic concept of unicast flow traffics and the next five scenarios, $S_{4,5,6,7,8}$, describe multicast flow traffics. The models illustrated the flows of multimedia data that involved several parameters such as bit rate, layered streams for multicast, queue management, queue length, delay/latency, topology, application protocols, and traffic sources. We use network simulator 2 (NS2) version 2.34 for implementing the scenarios. The RLM protocol is available in NS2 from version 2.1bx to 2.34 and it does not have a module to evaluate the performance, so that we create our own scenarios.

First, Fig. 1 (a) illustrates the simple topology used by the Scenario 1, 2, and 3 when networks received TCP and UDP unicast data streams. These scenarios consist of four senders, one receiver, and one router and denoted by S_N model as follows: $S_1 = S_2 = S_3 : \{L_{0,1}(10, 2), L_{1,2}(10, 2), L_{1,3}(30, 1), L_{1,4}(20, 0.256),$

Fig. 1 Network topology for the simulation scenarios.

$L_{1,5}(20, 0.256)\}$. In Scenario 1, the senders N_0 and N_2 send FTP traffics and the senders N_4 and N_5 send Pareto traffics [25]–[27] with shape factor 1.5 ($\alpha = 1.5$). In Scenario 2, we changed the sending traffics on N_0 and N_2 to constant bit rate (CBR) flows, and in Scenario 3, we changed the nodes to FTP and CBR, respectively. The link $L_{1,3}$ is designed as a bottleneck link and the router uses a drop-tail queue as a typical queue management. These scenarios will evaluate the network performances and characteristics of simple TCP-based and UDP-based unicast flows.

Second, the Scenario 4 and 5 illustrate the transmission of multicast traffics delivered by both a network that has not a flow control and has a receiver-driven flow control (RLM protocol), respectively. They are denoted by S_N model as $S_4 = S_5 : \{L_{0,1}(50, 1), L_{1,2}(50, 0.1), L_{1,3}(50, 1), L_{3,4}(50, 0.25), L_{3,5}(50, 1), L_{5,6}(50, 0.05)\}$. In addition, the Scenario 6 and 7 are similar with the scenario 4 and 5 but with the addition of background traffic sent by node 7 and node 8. They can be denoted by S_N model as $S_6 = S_7 : \{L_{0,1}(50, 0.1), L_{1,2}(50, 0.1), L_{1,3}(50, 1), L_{3,4}(50, 0.25), L_{3,5}(50, 1), L_{5,6}(50, 0.05), L_{1,7}(50, 0.256), L_{1,8}(50, 0.256)\}$.

Finally, the last Scenario 8 involved RLM protocol, background traffics and TCP applications such as FTP sent by node 5, node 6, and node 13, respectively, and can be denoted by S_N model as $S_8 : \{L_{0,1}(40, 5), L_{1,2}(30, 5), L_{1,3}(50, 1), L_{1,4}(50, 1), L_{1,5}(20, 0.256), L_{1,6}(20, 0.256), L_{3,7}(20, 1), L_{3,8}(30, 5), L_{3,9}(30, 5), L_{3,10}(30, 5), L_{3,11}(30, 5), L_{3,12}(30, 5), L_{3,13}(10, 1)\}$. The S_4 to S_7 will use to evaluate multicast flows without and with, respectively, congestion control by delivering multi rate and background traffic, and The scenario S_8 evalu-

ates multicast flows with congestion control (multi rate) and background traffic Internet such as FTP. The $S_{4,5}, S_{6,7}, S_8$ and their properties can be shown as in Fig. 1 (b), 1 (c), 1 (d), and Table 1, respectively.

2.2 Performance Metrics

The Internet is an interconnection of the various types of heterogeneous networks and supports a wide range of applications. Problems arise when there is a huge gap in capacity between local networks and the Internet, called the bottleneck, that lead to congestion collapse. There are 3 ways to prevent congestion collapse, e.g. monitoring of traffic flow and network capacity, especially a bottleneck router, managing of transmission rate between sender and receiver, and congestion avoidance.

The data flows are said as TCP-friendly if their arrival rates do not exceed the arrival rate of TCP flows in the same circumstances, and we define TCP friendliness value as in Eq. (1).

$$F = \frac{T_{non-tcp}}{T_{tcp}} \quad (1)$$

where F =Fairness ratio, $T_{non-tcp}$ =non-TCP throughput (bps), and T_{tcp} =TCP throughput (bps). Floyd and Fall in [6] estimate a simple TCP throughput and Wang and Schwartz in [21] estimate a more complex TCP throughput, respectively, as shown in Eq. (2) and Eq. (3).

$$T(t_{RTT}, s, p) = \frac{1.5 \sqrt{\frac{2}{3}} \times s}{t_{RTT} \sqrt{p}} \quad (2)$$

Table 1 Properties of simulation models.

Parameters	Scenario							
	1	2	3	4	5	6	7	8
t_{sim}	60s							
Q_{mgmt}	DropTail							
Proto	tcp	udp	tcp, udp	mcast	mcast, rlm	mcast	mcast, rlm	tcp, mcast, rlm
Rate	vary							
L_{queue}	20 packets							
PktSize	1KB							
Traf _{src}	ftp, pareto	cbr, exp	ftp, cbr	cbr	cbr	cbr, pareto	cbr, pareto	ftp, cbr, pareto
Latency	vary							
#Layers	na ⁺				6*	na ⁺	6*	6*

⁺na=not available, * 32×2^m kbps; m=0,1,...,6

where t_{RTT} =round-trip time, s=packet size, p=packet loss rate.

$$T(t_{RTT}, t_{RTO}, s, p) = \min \left[\frac{W_m \times s}{t_{RTT}}, \frac{s}{t_{RTT} \sqrt{\frac{2bp}{3}} + t_{RTO} \min \left(1, 3 \sqrt{\frac{3bp}{8}} \right) p(1 + 32p^2)} \right] \quad (3)$$

where b=number of ACK packet, W_m =congestion window size, t_{RTO} =request time out

We calculate both instant throughputs and system throughputs in this paper. The instant throughput in Eq. (4) is a value that calculated by dividing the bytes of each packet by the time required to send the packet while the system throughput in Eq. (5) is a value that calculated by dividing the total bytes sent up to a given time by the total time in seconds. For calculating instant throughput, we use low pass filter EWMA (Exponentially Weighted Moving Average) in Eq. (6) with smoothness factor (α)=0.9.

$$T_i = \frac{PktSize_i}{t_{i(received)} - t_{i(sent)}} \quad (4)$$

$$T_{system} = \frac{\sum_i PktSize_{i(received)}}{\Delta t} \quad (5)$$

$$\bar{y}_i = \alpha \cdot \bar{y}_{i-1} + (1 - \alpha) \cdot x_i \quad (6)$$

where $\alpha = smoothnessfactor = \frac{n}{n+1} (n \geq 0)$

Besides the above performance metrics, we also calculated others such as the convergence time (CT), latency, jitter, and packet loss ratio (PLR). Convergence time is time needed by the receiver to achieve a steady state condition when receiving the data streams from a server due to natural behavior of non-deterministic flows. We calculated a convergence time as a constant jitter that leads to 0. CT, jitter (Eq. 7), and PLR are performance metrics needed by multimedia applications.

$$jitter = \frac{[t_{received}(j) - t_{sent}(j)] - [t_{received}(i) - t_{sent}(i)]}{j - i} \quad (7)$$

where:

t_{sent} = time when packet is sent from a sender/source (second)

$t_{received}$ = time when packet is received by a receiver/client (second)

j, i = packet sequence number, i -th and j -th, where $j > i$

The simulations use FTP, CBR, and Pareto as traffic source distributions and also 6 different transmission rates/layers (formulated by 32×2^m kbps, where m=number of layers {0,1,...,6}) when transmitting multicast traffics using layered-based multicast protocol, such as RLM. The protocol works based on congested links and capability on the receivers, known as receiver-driven control.

3. Our Result

We evaluate the performance of TCP fairness and the RLM when interacts with TCP applications and use 6 layered multicast transmission with background traffic Pareto with $\alpha = 1.5$.

3.1 Experiment

The simulation properties in this paper were run with Drop-Tail queue management, unicast and multicast flows combined with TCP-based and non-TCP based applications, background traffics, and 6 layers in multicast flows as shown in Table 1. We use 8 scenarios involving both unicast and multicast flows that use the network topologies as shown in Fig. 1

3.2 Throughput

The system throughput is very important to know the power of our transmission in the network systems. This means that how many bytes we are transferring the data by the unit of time in our networks. Simulation 1 to 3 shows the basic characteristics of controlled and uncontrolled unicast flows of TCP and UDP traffics, respectively. Figure 2(a) shows that TCP flow generated by FTP traffics competes fairly

Fig. 2 System throughput of scenario 1 to 8.

($F \approx 1$, as shown in Fig. 3 (a)) with other TCP flows and in Fig. 2 (b) and 2 (c) UDP flows generated by CBR traffics competes unfairly ($F \gg 1$, as shown in Fig. 3 (a)) with TCP flows.

This is due to that TCP has a flow control mechanism to monitor congested link while UDP has not. The CBR traffic is uncontrolled, unidirectional, and user-defined in packet size, packet interval and stream duration. Most multimedia applications such as audio and video streams are based on UDP and thus, they are also unreliable and compete unfairly with FTP streams as shown in Fig. 2 (a), Fig. 2 (b), Fig. 2 (c), and Fig. 3 (a). In our experiments, the scenarios have low values on throughput, especially when they compete with background traffics and multicast traffics as shown in Fig. 2 and the summary of network performance conducted by this research is shown in Fig. 3.

We used 6 layers of transmission rate as shown in Fig. 4 with the lowest rate is 32 kbps and the highest rate is 2,048 kbps. In RLM, a client should have 256 kbps of its link capacity when it subscribes 3rd layer otherwise it only get a lower layer or congestion. Figure 4 shows a member in scenario 8 that it reached the highest layer and then dropped

to the 3rd layer due to Pareto traffics flows.

3.3 Packet Loss Ratio (PLR)

PLR is important to evaluate the quality of services (QoS) and link efficiency in the network system, especially for multimedia streaming. Boyce and Galianello in [28] evaluate the MPEG video performance that 3% of packet loss could cause 30% of the video frame cannot be used. Figure 3 shows the result of PLR on all scenarios when transmitting TCP and non-TCP data flows. Most of the unicast traffics both TCP flows and UDP flows have PLR range from 0% to 4% as shown in Fig. 3, Scenario 1 to 3.

A network that comprises RLM mechanisms for managing or controlling flows on each multicast member also has PLR value range between 0.35% to 6.69% as shown in Fig. 3, Scenario 5,7,8. However, a network that comprises only UDP multicast flows, the PLR is relatively high between 17% to 61% as shown in Fig. 3, Scenario 4 and 6. This is happened due to a heavy load of uncontrolled multicast traffics on the network.

Fig. 4 Joining a layer on scenario 5, 7, and 8.

Measurements	Scenario I		Scenario II		Scenario III		Scenario IV		
	Unicast + BgTraf	FTP2	Unicast + BgTraf	EXP	Unicast + BgTraf	FTP	M2	M4	M6
BW _{Link} (kbps)	1,000	1,000	1,000	1,000	1,000	1,000	100	256	50
T _{min} (kbps)	2.22	0.54	16.36	22.19	7.72	131.61	23.13	39.46	20.35
T _{max} (kbps)	606.91	367.87	224.93	249.05	507.06	492.77	87.99	215.48	48.08
T _{avg} (kbps)	284.93	314.32	158.88	240.90	172.29	481.42	72.82	167.63	35.62
Efficiency	59.92		39.98		65.37		72.82	65.48	71.24
CT (sec)	0.10	0.50	0.43	0.30	0.10	0.30	9.21	3.02	1.00
PLR (%)	1.29	0.82	0.00	0.00	2.44	2.35	23.35	17.75	18.29
F	1.10		1.52		2.79		-	-	-

(a) Scenario 1 to 4

Measurements	Scenario V			Scenario VI			Scenario VII		
	MCast + RLM			MCast + BgTraf			Mcast + BgTraf + RLM		
	M2	M4	M6	M2	M4	M6	M2	M4	M6
BW _{Link} (kbps)	100	256	50	100	256	50	100	256	50
T _{min} (kbps)	36.91	37.94	20.35	41.56	31.53	20.18	39.08	35.28	3.88
T _{max} (kbps)	92.08	202.68	47.16	86.98	135.00	48.28	92.71	166.64	30.37
T _{avg} (kbps)	65.10	94.65	39.39	62.12	93.34	43.13	69.99	97.00	19.26
Efficiency	65.10	36.97	78.79	62.12	36.46	86.26	69.99	37.89	38.52
CT (sec)	24.19	24.56	4.38	4.22	3.41	1.00	21.79	*)	4.38
PLR (%)	1.86	0.35	3.03	51.74	60.78	22.13	1.68	0.32	6.69
F	-	-	-	-	-	-	-	-	-

(b) Scenario 5 to 7

Measurements	Scenario VIII					
	Unicast+Mcast+BgTraf+RLM (Shared-Link: R1-R2)					
	TCP/FTP	M4	M8	M9	M10	M11
BW _{Link} (kbps)	1,000	1,000	1,000	1,000	1,000	1,000
T _{min} (kbps)	3.45	78.44	37.27	37.27	34.20	36.65
T _{max} (kbps)	858.18	704.81	417.06	417.06	371.83	371.83
T _{avg} (kbps)	795.00	133.53	98.20	87.98	89.93	92.70
Efficiency	79.50	13.35	9.82	8.80	8.99	9.27
CT (sec)	0.10	0.10	0.10	0.10	0.10	0.10
PLR (%)	0.09	1.30	3.85	3.85	3.85	3.85
F	1.00	0.17	0.74	0.90	1.02	1.03

(c) Scenario 8

Fig. 3 Simulation results of 8-scenarios. BgTraf is a background traffic that has a self-similar traffic pattern based on Pareto distribution with $\alpha = 1.5$ and $\beta = 500$ ms and the *) symbol in Fig. 3 (b) indicates the links have experienced a congestion collapse due to heavy traffics to Node 4 exceeding the link capacity.

3.4 Convergence Time and Jitter

Figure 3 shows that most scenarios have achieved a convergence time in less than 2 seconds in average except scenario

5 that has more than 20 seconds in average. This means both TCP and RLM could work together and achieved a steady state condition in a short time. Besides convergence time, we also calculated another metric related to convergence time that is a jitter.

Jitter is a delay variation needed by multimedia services as a constraint to improve its quality of service in delivering the data transmission from the provider to subscribers. It also can be used to determine the convergence time of the protocol. UDP flows are quicker than TCP flows to achieve their convergence time because of using constant rate as shown in Fig. 6 and Fig. 7 while TCP flows are hard to achieve a convergence time due to its characteristic to react when another flows occupying in the network. TCP has a variety of jitter values in a range from 0 to 20 milliseconds and UDP has jitter values in a range from 0 to 5 milliseconds as shown in Fig. 5.

3.5 Fairness

Fairness shows whether a protocol dominating other protocols or not on the networks in the same circumstances. As shown in Eq. 1, each application should occupy the networks fairly. These circumstances could not be reach without a flow control and monitoring mechanism. Figure 3 (Scenario 1) and Fig. 2 (a) show that the maximum throughput of TCP is more than 600 kbps and react fairly when other flows occupied in the networks on the same circumstances, the maximum throughput of UDP is approximately 249 kbps, as shown in Fig. 3 (Scenario 2) and 2 (b), when the networks only occupied by UDP flows, and TCP flows dropped more than 100% when competing with UDP flows due to it responds to UDP flows.

Figure 3 shows TCP and RLM have fairness values from 1.0 to 2.7 in Scenario 1, 2, 3, and 8. We concern on both TCP-based and non TCP-based applications working together in the Internet because most of data traffics in the Internet are TCP-based. It is almost impossible that the Internet only consists non TCP data traffics except in particular case such as in private networks. This is a reason why we did not examine the fairness values in Scenario 4 to 7 due to all traffics involving in these scenarios are non TCP flows.

Fig. 5 Jitter of unicast flows.

Fig. 6 Jitter of scenarios 4 to 7: Multicast flows.

4. Discussion

This paper proposed some scenarios to act friendly between both TCP applications and multicast applications. We focused on the works how to improve and evaluate QoS and link efficiency for the real time applications. Some of performance evaluations such PLR, fairness, jitter, throughput, and convergence time have been evaluated. They are impor-

Fig. 7 Jitter of scenarios 8: Multicast flows.

tant factors to improve QoS on real-time applications.

Our study showed that most scenarios could reach TCP friendliness even they have low throughputs, difficult to reach CT, and PLR more than 3%. We use background traffics Pareto and DropTail queue management, which can be a subject to be discussed in the future to overcome these results by determining the appropriate distribution function for background traffics. RLM also does not support error recovery mechanisms that can be a problem for reliability issue.

5. Conclusion

TCP-based applications, such as email and http, use a reliable and controlled protocol. A network that contains uncontrolled data packet streams will be more prone to congestion. RLM protocol works well enough to push the packet loss ratio and keep traffic flows from congestion incidents of non-TCP data packets even it takes longer than other proto-

cols to reach convergence time (steady state).

In our study of 8-simulation scenarios, RLM that delivers 6 layers of multicast transmission still has a low network utilization when using a channel capacity. This is due to the RLM has a subscription mechanism to a specific layer which can be adapted to network condition. Its mechanism is fair enough to TCP traffics ($F \approx 1$) when use a channel in the networks in the same circumstances which means that RLM could control non-TCP traffics on TCP traffics fairly. Other results show the convergence time difficult to reach quickly, vary in jitter (0-10 ms), low values on throughput, efficiency, and PLR ($PLR > 3\%$) in the same circumstances.

RLM has low values on throughput, efficiency, and PLR, but it works well together with TCP applications. RLM can also reduce loss packet, but it does not have an error recovery mechanism. It cannot be used to the applications that need high reliability, such as database replications.

Acknowledgments

The authors would like to thank Directorate General of Higher Education (DGHE), Department of National Education, Indonesia for supporting Ph.D scholarship, the members of Information Technology and Security Laboratory, Department of Informatics, Kyushu University, Japan and the members of Net-Centric Computing Laboratory at Department of Computer Science, Bogor Agricultural University, Indonesia for valuable help and support.

References

- [1] S. Jagannathan, K.C. Almeroth, and A. Acharya, Topology Sensitive Congestion Control for Real-Time Multicast, Department of Computer Science, University of California Santa Barbara, CA, 2000.
- [2] P.A. da S. Goncalves, J.F. de Rezende, O.C.M.B. Duarte, G. Pujolle, and M. Curie, Improving Layered Video Multicast Using Active Networks. Available at <http://citeseerx.ist.psu.edu/viewdoc/summary?doi=10.1.1.25.738>
- [3] C. Bouras, A. Gkamas, and G. Kioumourtzis, "Simulcast transmission for video applications: Performance evaluation with an integrated simulation environment," International Symposium on Performance Evaluation of Computer and Telecommunication Systems (SPECTS 2009), Istanbul, Turkey, July 2009.
- [4] V. Jacobson, "Congestion avoidance and control," Proc. ACM SIGCOMM, Stanford, CA, Aug. 1988.
- [5] Y.C. Lai and R.S. Cheng, Some Improvement of TCP Congestion Control, ICOIN-Korea, 2003.
- [6] S. Floyd and K. Fall, "Promoting the use of end-to-end congestion control in the Internet," IEEE/ACM Trans. Netw., vol.7, no.4, pp.458-572, 1999.
- [7] J. Widmer, R. Denda, and M. Mauve, "A survey on TCP-friendly congestion control (extended version)," Technical Report, TR-2001-002, Department of Math. and Comp. Science, University of Mannheim, 2001.
- [8] S. Floyd, M. Handley, and J. Padhye, A Comparison of Equation-Based and AIMD Congestion Control, ACIRI, May 2000. Available at <http://www.icir.org/tfrc/aimd.pdf>
- [9] Y.R. Yang and S.S. Lam, "General AIMD congestion control," Proc. ICNP, Osaka, Japan, Nov. 2000.
- [10] J. Padhye, J. Kurose, D. Towsley, and R. Koodli, "A model-based TCP-friendly rate control protocol," Proc. NOSSDAV, 1999.
- [11] S. Floyd, M. Handley, J. Padhye, and J. Widmer, "Equation-based congestion control for unicast applications," Proc. ACM SIGCOMM, May 2000. Available at <http://www.icir.org/tfrc/tcp-friendly.pdf>
- [12] J. Feng and L. Xu, "On the time scale of TCP-friendly admission control protocols," Proc. ICC, pp.45-51, 2008.
- [13] S. Jin, L. Guo, I. Matta, and A. Bestavros, TCP-Friendly SIMD Congestion Control and Its Convergence Behavior, Computer Science Department, Boston University. Available at <http://www.cs.bu.edu/fac/best/res/papers/icnp01.pdf>
- [14] H.A. Wang and M. Schwartz, "Achieving bounded fairness for multicast and TCP traffic in the Internet," Proc. ACM SIGCOMM, 1998.
- [15] S. McCanne, V. Jacobson, and M. Vetterli, "Receiver-driven layered multicast," Proc. ACM SIGCOMM, pp.117-130, Palo Alto, CA, 1996.
- [16] J. Byers, M. Frumin, G. Horn, M. Luby, M. Mitzenmacher, A. Roetter, and W. Shaver, "Flid-dl: Congestion control for layered multicast," Proc. Second International Workshop on Networked Group Communication (NGC 2000), Nov. 2000.
- [17] J. Byers, M. Luby, M. Mitzenmacher, and A. Rege, "A digital fountain approach to reliable distribution of bulk data," Proc. ACM SIGCOMM, Sept. 1998.
- [18] T. Turletti, S. Parisi, and J. Bolot, "Experiments with a layered transmission scheme over the Internet," Technical Report RR-3296, INRIA, France, Nov. 1997.
- [19] W. Tan and A. Zakhor, "Error control for video multicast using hierarchical FEC," Proc. International Conference on Image Processing, Oct. 1999.
- [20] L. Vicisano, J. Crowcroft, and L. Rizzo, "TCP-like congestion control for layered multicast data transfer," Proc. IEEE INFOCOM, vol.3, pp.996-1003, March 1998.
- [21] H.A. Wang and M. Schwartz, "Achieving bounded fairness for multicast and TCP traffic in the Internet," Proc. ACM SIGCOMM, 1998.
- [22] N. Cranley and M. Davis, Performance Evaluation of Video Streaming with Background Traffic over IEEE 802.11 WLAN Networks, Oct. 13, 2005.
- [23] Z. Kenesi, Z. Szabo, Z. Belicza, and S. Molnar, "On the effect of the background traffic on TCP's throughput," Proc. ACM COMM, pp.631-636, ISSN 1530-1346, June 2005.
- [24] R. Stanojevic, R.N. Shorten, and C.M. Kellett, "Adaptive tuning of drop-tail buffers for reducing queueing delays," Proc. IEEE Communications Letters, vol.10, no.7, pp.570-572, ISSN 1089-7798, 2006.
- [25] A.B. Downey, "Lognormal and pareto distributions in the Internet," Proc. ACM Computer Communications, vol.28, pp.790-801, 2005. ISSN:0140-3664
- [26] K. Park and W. Willinger, "Self-similar network traffic: An overview," in Self-Similar Network Traffic and Performance Evaluation, pp.1-34, Wiley-Interscience, New York, 2000.
- [27] V. Paxson and S. Floyd, "Wide area traffic: The failure of poisson modeling," Proc. IEEE/ACM Transactions on Networking (TON), v.3 n.3, pp.226-244, June 1995. Available at <http://dx.doi.org/10.1109/90.392383>
- [28] J.M. Boyce and R.D. Gaglianella, "Packet loss effects on MPEG video sent over the public Internet," Proc. ACM SIGMULTIMEDIA, pp.181-190, 1998.

Heru Sukoco was received Bachelor of Computer Science and Master of Engineering from Bogor Agricultural University (IPB) in 1999 and Bandung Institute of Technology (ITB) in 2005, respectively. He is interested in net-centric areas such as network design, network and database programming, network analysis and performance, network simulation, high speed computing, and wireless and mobile technology. Now, He is studying as PhD student in Department of Informatics, Graduate School of

Information Science and Electrical Engineering, Kyushu University, Japan.

Yoshiaki Hori is an Associate Professor of Department of Computer Science and Communication Engineering, Kyushu University, Japan since 2004. He was received B.E., M.E., and D.E. of Computer Science from Kyushu Institute of Technology, Iizuka in 1992, 1994, and 2002, respectively. He was also a Research Associate with Common Technical course of Faculty of Design, Kyushu Institute of Design in 1994. He is interested in Computer Network and Distributed Systems. He is a member of IPSJ,

IEEE and ACM.

Hendrawan is lecturer and researcher at Telematics Laboratory of School of Electrical Engineering and Informatics, Bandung Institute of Technology (ITB), Indonesia. He was received B.Eng in Electrical Engineering from ITB in 1985, MSc. in Telecommunication and Information Systems in 1990, and PhD. in Electronic System Engineering in 1994 both from University of Essex, UK. Now, He is also a Head of Doctoral and Master Programs of School of Electrical Engineering and Informatics, ITB and a member of IEEE and ACM. His research interest includes Telecommunication Performance Engineering, Computer Network and Multimedia.

ics, ITB and a member of IEEE and ACM. His research interest includes Telecommunication Performance Engineering, Computer Network and Multimedia.

Kouichi Sakurai is a Professor of Department of Computer Science and Communication Engineering, Kyushu University, Japan since 2002. He was received B.E., M.E., and D.E. of Mathematics, Applied Mathematics, and Computer Science from Kyushu University in 1982, 1986, and 1993, respectively. He is interested in Cryptography and Information Security. He is a member of IPSJ, IEEE and ACM.