

High-speed transmission and Control plan on high-definition video file using Parallel TCP

Hyoungyill Park*, Sangdo Lee**, Yongtae Shin*

* Department of Computing, Soongsil University

** Computer Engineering, Yonsei University

harrypark58@gmail.com, piterlee@gmail.com, shin@ssu.ac.kr

Abstract— Recently for the transmission of high-capacity video file between sites for manufacturing high-definition used by broadcasting company the Public Network is used a lot, the IP Public Network with QoS(Quality-of-Service) not guaranteed degenerates in performance due to different causes. In order to maximize the effectiveness of such a Network the high efficiency Network can be realized enabling the high-speed of HD video file using the connection with networks between hosts through packet creation and multi-session of Parallel TCP. This thesis is designed to test the high-speed transmission of high-capacity HD video file, and to study the plan for control of relatively fair transmission capacity to many users by the control of Socket from end to end in Public Network by making use of the transmission method using Parallel TCP Application of Open Source.

Keywords— Parallel TCP, Transmission of video file, High speed, Data transmission, TCP performance.

I. INTRODUCTION

As the uncompressed signal or MPEG-2 video with 50Mbps is used due to repeated editing process of video file capacity for actual manufacture for HDTV by the main broadcasting companies, the one hour HD video is more than 23GB very high. For example, though the video file at the stage of manufacture or before distribution to the distributor uses Gigabit class network the transmission between sites with traditional File Transfer method does not exhibit sufficient efficiency. One of the greatest reasons is due to that TCP actually used in FTP Application has re-transmission algorithm on lost segment. Accordingly as the efficiency is far lower with all the considerable development of recent network, the actually operated Network consumes a few times more time by several obstruction factors and considerably lowers the efficiency.

In order to increase the end to end transmission speed, quality target in the sense of user many methods are used. First, there is a method to maximize the resources utilization of network by using the network to expand the Network bandwidth or guarantee QoS. The QoS of Network shall secure the bandwidth of data for transmission, and be transmitted within the time required by the packet, whose loss shall be maintained at less than the regulation[7]. Accordingly

in order to guarantee the QoS of packet several methods such as securement of Device resources and given preference are used. However if the technical factor having to guaranteeing QoS is not limited to the real time, but only to effective use of network and packet loss, it will be a better plan to improve the efficiency of TCP/IP protocol than to secure the network resources for preference of transmission. Especially, the efficiency of Network generating from the Public Network is not very good due to numerous impacts. Above all, as the transmission of large-capacity file for broadcasting material is not a traffic of small data generally used it has an unfavorable environment in the fair network transmission using efficiency with sharing Network or many users. As methods for this the hardware based WAN accelerator is used, the Fast TCP is equipped inside Kernel, or the method to establish the separate private network or CDN network has been used. However as such methods consume lots of cost and time, and are the paralleled ones requiring the hardware improvement following many limitations they are not economical.

In the chapter 2 of this study it is designed to find the factors and causes influencing the transmission efficiency of TCP/IP protocol, and to find which methods are used in the end to end application for improving this. And the chapter 3 covers several methods for maximizing the performance efficiency of large-capacity video file transmission using Parallel tcp and the plan to control the performance through control of stream.

II. 1. CHARACTERISTICS OF TCP TRANSMISSION

A. 1.1. Factor influencing Network transmission performance

In case of large-capacity video file with a few GB's its efficiency relatively greatly lowers in the network transmission to a small data file. The transmission of file by TCP is greatly influenced by RTT and Buffer Size.

TCP decides how much amount of packet it can send at a time using Congestion Window (CWND) called Buffer Size. This means that the more this increases the more amount of packet it can send. TCP "Slow start" and "Congestion avoidance" decide the size of Congestion Window. If the buffer size is too small it cannot exhibit a

sufficient TCP performance, if the transmission buffer is great and the reception is small the packet will overrun, and if the transmission buffer is too small the transmission will be very difficult. In order to keep the maximum transmission efficiency it is necessary to optimize the size of Kernel Socket Buffer for transmission and reception, or Buffer size suited to the network environment.

- The optimized buffer size is two times the Bandwidth * delay of link.
- Buffer size = 2 * Bandwidth * Delay.

Ping program is used for acquiring the end to end delay, multiplying which by two times becomes RTT (Round Trip Time).

- Buffer size = Bandwidth * RTT.

For example, if the Ping time of Network with 100Mbps is 50ms, TCP Buffer Size becomes 0.05 sec * (100 Mbit / 8 bits) = 625 KBytes[1].

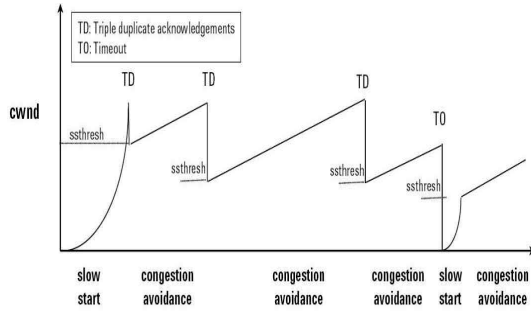


Figure 1. Congestion control form of TCP

Figure 1 is an algorithm to make a congestion control at TCP, which shows how the AIMD (Additive increase / Multiplicative Decrease) works in the Network in a graphic form [2].

Another following equation (1) describes the detailed factors influencing the performance of TCP. It can be found that the lossy-less file transmission TCP protocol is greatly affected by the Loss and RTT of packet. In other words, it can be found that in the communication in a distant place with great RTT and Loss the Bandwidth, or Network is very much lowered. In the network it expresses the single TCP channel performance [3].

$$BW = \frac{MSS}{RTT \sqrt{\frac{2bp}{3}}} \quad (1)$$

p: loss rate
RTT: round trip time
MSS: max segment size
b : constant

It can be found that the bandwidth of network is in inverse proportion to loss and RTT of packet, but in proportion to MSS.

Accordingly if in a distance under a large-capacity Network transmission condition the RTT becomes greater, and the packet loss more, the transmission capacity can be reduced by dramatically decreasing the Congestion Window by more than the multiple of 1/2. This brings such a result as not to sufficiently utilize the available bandwidth of actual network.

B. Network protocol and application for a large-capacity file transmission

In order to effectively transmit a large-capacity high definition video file in a distance different forms of methods are used. Typically the transmission using TCP applied Parallel TCP and UDP protocol transformed Reliable UDP enable the effective use of network at the application class.

Reliable UDP is the one to add the data reliability to the reliable UDP without Congestion Control and to realize the Congestion Control algorithm enabling a high-speed transmission, which has a function of re-transmission of lost packet for the reliable transmission, and Rate-based Congestion Control capability resulting in its being possible to make a properly effective use with other traffics through Congestion Control.

Parallel TCP method enables the rate to reduce the transmission data to be 1/(2N) as the rate to reduce data transmission rate becomes N channels by using the multiple channel when any congestion occurs [4].

III. PARALLEL TCP TRANSMISSION MODEL

A. Parallel TCP's Principle

The equation below describes the relationship with performance of Parallel TCP and Parallel level in Parallel TCP model [3].

$$BW_n = \frac{MSS}{RTT_n} \frac{n}{\sqrt{p_n}} \frac{c_1}{\sqrt{\frac{2b}{3}}} \quad (2)$$

$$BW = \sum_{n=1}^{ParallelLevel} BW_n \quad (3)$$

n: number of parallel flows
p: loss rate
RTT: round trip time
MSS: max segment size
b and c1: constant

Parallel TCP prevents the decrease in performance occurring at a single TCP channel by using N Parallel Flows, and enables to exhibit the maximum End to End performance in the allowed bandwidth. Theoretically it seems to enable the N times performance of a single channel[Figure 2].

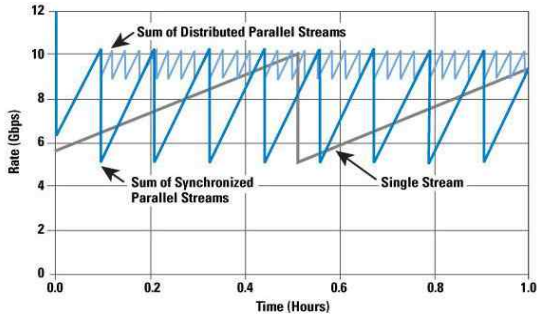


Figure 2. Congestion Control of Parallel TCP

The Application for Parallel TCP transmission in the test controls TCP Flows by use of API of Rapidant to be provided as an Open Source [2].

```
{
String numOfStream =
line.getOptionValue(ParameterInfo.NUM_OF_STREAM);
config.setNumOfStream(Integer.parseInt(numOfStream));
};
```

The setting of API option of Rapidant software Client is as follows[2].

```
rapidantClient.jar -f movie.mov -d movie.mov -host
172.16.0.33 -n 5
-n : the number of streams
-sbuff : socket buffer size
```

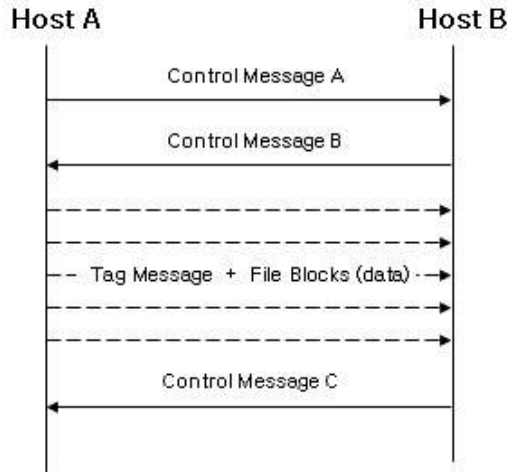


Figure 3. Basic Transfer Procedure.

B. Parallel TCP's Performance Measurement

In order to measure the performance by all sorts of variables of Parallel it was measured by operating the Rapidant at server, and varying the number of flows for the transmission of a large-capacity file at Client, and the transmission rate by change in Buffer Size was measured.

Figure 4 and Table 1 show the values measured by increasing the stream of high-definition video file with

approximately 4GB at Public internet with 90Mbps, RTT 20ms, and Socket Buffer 8MB.

The measured value is the one to have measured the average transmission speed of file by increasing the number (n) of TCP Flows from 1 up to 30.

TABLE 1. TRANSMISSION SPEED BY THE NUMBER OF TCP FLOWS

n	Bandwidth (Mbps)	n	Bandwidth (Mbps)
1	14.866	16	84.423
2	24.181	17	86.384
3	30.985	18	86.085
4	32.816	19	87.325
5	39.232	20	87.434
6	43.097	21	87.181
7	50.246	22	86.915
8	56.168	23	87.816
9	66.424	24	87.365
10	68.919	25	87.309
11	75.248	26	87.236
12	76.371	27	87.184
13	78.571	28	86.857
14	79.865	29	87.451
15	81.183	30	86.847

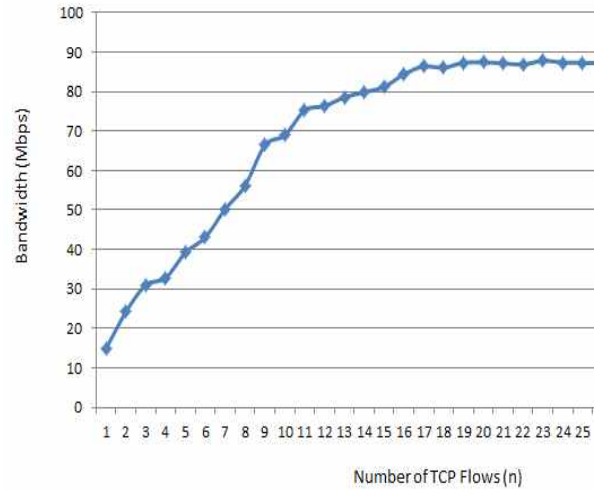


Figure 4. The variation in average transmission speed by control of TCP Stream n

Figure 5 shows the one to have measured the transmission speed by varying TCP Flows and size of Socket buffer. The transmission speed of network is shown to have to be optimized in proportion to the Flows of TCP and size of Socket buffer.

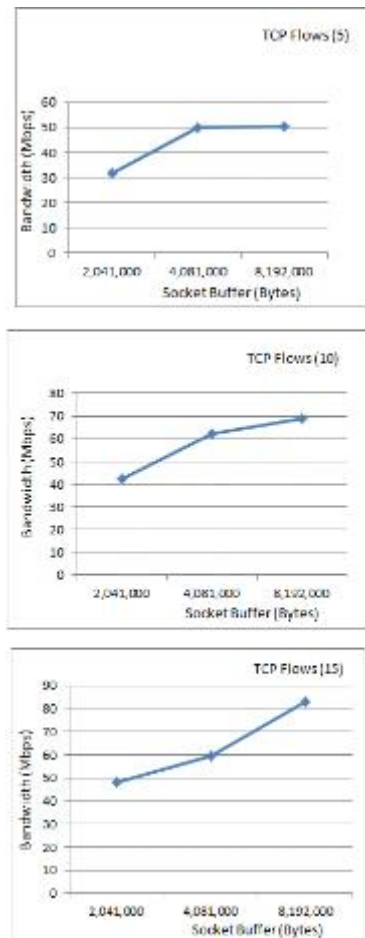


Figure 5. The variation in average transmission speed by coordination of TCP S-Buffer

C. Parallel TCP's Performance Control

As shown in the test in case of Parallel TCP the performance of network transmission can be controlled by the number of Streams and Socket Buffer Size. When the stream $n=1$ in a Public Network with approximately 100Mbps it is average 14Mbps, when $n=15$ it is average 82Mbps, and when more than $n=20$ it reaches the saturated condition to use almost all the network resources.

As another additional method the transmission speed can be improved by increasing Socket Buffer. This can be found that in case of more streams the Parallel performance suitable for its numbers of streams can be exhibited only after the quantity of Socket Buffer is sufficient. Accordingly as Socket buffer and stream have a close relationship with each other the proper tuning of buffer shall be made at the same time in order to more effectively control the efficiency of network performance

IV. CONCLUSIONS

The performance of TCP is largely dependent on RTT. In order to maintain the transmission of a large-capacity file in a

distance with high efficiency network the high efficiency can be maintained by utilizing Parallel TCP. However in order to distribute the fair resources to file transmission host with the limited network it can be found through the test of this study that the quite effective distribution of resources can be made by controlling the number of streams of Parallel TCP.

It is possible to control the bandwidth as maintaining the maximum efficiency in such a network as not to guarantee the actual QoS, which is possible to control in such a network as to measure hundreds of communicable RTT, according to whose environment the number of streams shall be limited and re-coordinated. In order to control a little more reliable transmission speed a little more ideal network shall be formed, the study with still more variables added shall be proceeded as far as aiming at the public network actually used.

The study on the overhead occurring the process of transmitting TCP to Parallel and a little reliable bandwidth control algorithm shall be continued.

And the study on transmission of Parallel TCP for reliable transmission has been made, it is also necessary to make a parallel study on UDP transmission for transmitting HD 1080i encoding video stream without closing like real-time broadcasting.

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