

Comparison of Real-time Streaming Performance Between UDP and TCP Based Delivery Over LTE

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Abstract. Video traffic becomes dominant in mobile networks since users use mobile networks for enjoying more instant and handy video service. Currently, TCP and UDP are major transport layer protocols for video streaming service. This paper compares performance of those protocols over LTE in terms of Maximum Available Bitrate (MAB). The results will become the key clue to select one between DASH (Dynamic Adaptive Streaming over HTTP) and MMT (MPEG Media Transport).

Keywords: Mobile network · UDP · TCP · Video streaming service · LTE · Maximum available bitrate · MMT · DASH

1 Introduction

Mobile market is gradually growing in the size, and the ratio of the mobile data traffic occupied by the video is 55 % in 2012. The part occupied by the video traffic is expected that more than 69 % in 2019 [1]. Using mobile is more comfortable than other equipment (laptop or desktop) to use consumer goods. This causes the result of be offered the real-time service. The video streaming service is a type of real-time service and there are two types of video streaming services which are Video on Demand (VOD) Streaming Service and Real-time Streaming Service. In case of VOD streaming, the server has various media files and sends the requested media file to the client. Another case which is the real-time streaming service, the server does not have whole of video file but has fragments of the filming video. Both the VOD service and the real-time streaming service must have buffer for smooth video playback, because of the network delay which is caused by various factors. To create a robust system in delay, the de-jittering buffer must be existed. Presence of the buffer causes the buffering which induce that the video played after a period time as well as the larger the buffer, the longer the delay to be axiomatic. Therefore, we think that it is important to measure the accurate available bitrate to decrease the buffering delay. We measure Maximum Available Bitrate (MAB) of UDP and TCP which one can be the main transport layer protocol to transfer streaming video data as well as analyze which protocol is more suitable for the estimate available bitrate.

2 Related Work

In this chapter, we present features of UDP and TCP as well as the estimate available bitrate algorithm to improve End-to-End QoS.

2.1 Features of UDP and TCP

UDP is the best-effort protocol which does not guarantee reliable transmitting. It causes the need to add Forward Error Correction (FEC) code. TCP is the connection-oriented protocol which guarantees reliable transmitting with congestion and flow control. For this reason, TCP is a reliable protocol than the UDP. Data rate, on the other hand, is higher UDP than TCP since UDP does not control the rate for reliability. Because of the trade-off in reliability and data rate, the service provider is used to select the protocol suitable for the service that it provides. To transmit video stream to contents consumer, UDP and TCP are both commonly used in recent years.

2.2 Estimate Available Bandwidth

In this section, we briefly describe the existing estimate available bitrate algorithm.

Pathload uses a one-way delay of periodic packet stream and observes that the stream rate is how much increase as compared to the available bitrate [2]. It can apply the amount of bandwidth similar with the estimate available bandwidth to the adaptive application, there is a small disadvantage to large time convergence.

pathChirp is the method using the concept of “self-induced congestion”. Exponentially time spaced packets are used and available bandwidth is estimated by using that the amount of delay is decreased or increased [3].

TOPP use the analysis method by segmented regression. The well-separated probe packet pair is used, the available bitrate is measured using a delay, regardless of the network condition [4].

pathQuick2 utilizes the probe slope model, including the concept of the probe rate model [6]. PSM-based method measures the effective UDP throughput quickly, and estimates the available bandwidth at the same time. Each packet in the packet train is transmitted in the same time interval, *pathQuick2* uses a delay of time interval [5].

[7] presumes available bitrate using the one-way delay distribution. The Bayesian mechanism is proposed to model bandwidth variations. The algorithm can estimate the available bandwidth very fast, because it does not use the RTT, but uses the one-way delay which is receiver-based mechanism. However, service provider sets some parameters to estimate the available bandwidth, but those parameters are not shown on.

3 Experiment Method and Mathematical Model

In reality, the change of the MAB is irregular and difficult to predict due to various factors. The irregular characteristics make it difficult to adapt streaming service application to the current channel, which immediately appears to the degradation of

QoS and QoE. In this chapter, we propose some parameters which are the criterion that which one is more appropriate between the UDP and the TCP, also those are the value of applicability on adaptive streaming service.

3.1 Measure Method

To analyze network statement, one PC sender and one laptop receiver were used. Laptop computer connect Internet using LTE network by the tethering using mobile phone. Sender transmits more than 1 Mbytes using 10 kbytes packet over best-effort on UDP. If receiver takes 1 Mbytes size data, it sends a signal which uses flag and measuring RTT to server. Sender receives this signal and sends 1 Mbytes segment immediately. Repeating this simple test for a period of time, MAB is measured. The method is as follows Fig. 1.

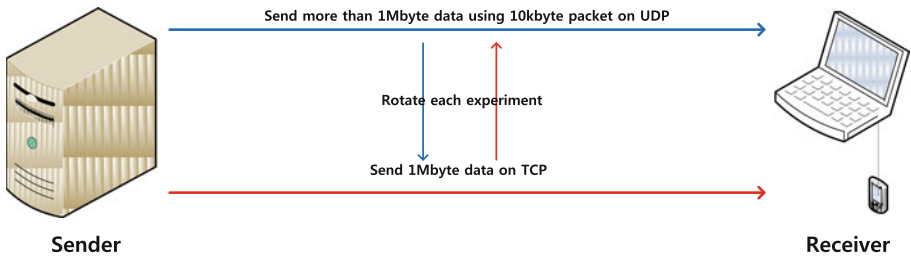


Fig. 1. Experimental method

1. UDP MAB

Receiver measures UDP MAB using the getting started time (T_{1st}) and the finishing received time (T_{last}). The equation is as follows (1).

$$MAB_{UDP}(t) = \frac{1Mbyte \times 8(bit/byte)}{(T_{last_i} - T_{1st_i}) \times 10^{-3}s} (Mbps) \tag{1}$$

Also, $T_{last_i} - T_{1st_i}$ is defined as the inter arrival time J_{i_UDP} .

2. TCP MAB

Receiver measures TCP MAB using the getting started time (T_{T1st}) and the finishing received time (T_{Tlast}). The equation is as follows (2).

$$MAB_{TCP}(t) = \frac{SegmentSize(byte) \times 8(bit/byte)}{(T_{Tlast_i} - T_{T1st_i}) \times 10^{-3}s} (Mbps) \tag{2}$$

Also, $T_{Tlast_i} - T_{T1st_i}$ is defined as the inter arrival time J_{i_TCP} .

3. Markov model

The Markov model is a valuable model that can be used to “predict the future through the past”. In this paper, bitrate is how sustainable and changeable between a good environment and a bad environment. The threshold of two probability variable is 14 Mbps for UDP and 9 Mbps for TCP. Value to be obtained through the model are shown in the following (3) ~ (4).

$$P_{g \rightarrow b} = \frac{P_{gb}}{P_{gg} + P_{gb}} \tag{3}$$

$$P_{b \rightarrow g} = \frac{P_{bg}}{P_{bg} + P_{bb}} \tag{4}$$

S.Time is a formula on how much each state is maintained. The equations are shown in the following (5) ~ (6).

$$S.Time_{gb} = \frac{1}{P_{g \rightarrow b}} \times PacketTimeInterval(sec) \tag{5}$$

$$S.Time_{bg} = \frac{1}{P_{b \rightarrow g}} \times PacketTimeInterval(sec) \tag{6}$$

UDP and TCP is interpolated every 500 ms to ease the calculations, because each experiments run at random time space. So, *PacketTimeInterval* is 0.5 s in this paper.

4. Second Order Differential

Second-order differential of bitrate means instantaneous rate of change of bitrate, using equations shown in (7) ~ (8).

$$f'_{MAB}(t_k) = \frac{f_{MAB}(t_{i+1}) - f_{MAB}(t_i)}{t_{i+1} - t_i} \quad \text{when, } t_k = \frac{t_{i+1} + t_i}{2} \tag{7}$$

$$f''_{MAB}(t_l) = \frac{f'_{MAB}(t_{k+1}) - f'_{MAB}(t_k)}{t_{k+1} - t_k} \quad \text{when, } t_l = \frac{t_{k+1} + t_k}{2} \tag{8}$$

5. Correlation

If the random variable has more than two, the joint PDF can be used to perform various calculations. Among them, covariance and correlation coefficient is used to learn correlation between random variable X and Y. The formula for covariance is the same as (9).

$$\sigma_{XY} = Cov(X, Y) = E[(X - \mu_X)(Y - \mu_Y)] = E(XY) - \mu_X\mu_Y \tag{9}$$

μ_X and μ_Y is $E(X)$ and $E(Y)$, the correlation coefficient through covariance is defined by the following (10).

$$\rho = \frac{Cov(X, Y)}{\sqrt{Var(X)Var(Y)}} \tag{10}$$

ρ has a value between -1 and 1, if $\rho = 0$, X and Y is independent. In this paper, we analyze correlation using MAB and RTT.

4 Comparison Between UDP and TCP

4.1 Maximum Available Bitrate

This experiment carried out aboard the bus around Gangnam of Seoul, Korea as shown in Fig. 2. It measured during about 1000 s. The result of experiment is represented in Fig. 3. The blue line is the MAB of UDP and the red spotted line is the MAB of TCP. Generally, the MAB of TCP is higher than the MAB of UDP. In addition to, we can confirm that the variation of TCP is larger than the UDP. If the MAB is low, the client cannot receive high quality video. Large variance means that quality of service is often changed. It force to use larger buffer, it leads to longer delay that the time was taken to play video. It is not desirable for real-time streaming service, because real-time streaming service is sensitive for delay. Especially, when offered service is sports, client was spoiled crucial moments, because of delay.

Additionally, the loss packet and the bitrate are closely related. About 20,000 of the UDP packets were sent and only 5 packets were lost. The loss caused by the congestion and the fading is substantially not occur, this part is ignored in this paper.



Fig. 2. Gangnam, Seoul, Korea, the map which is place of experiment

How long network conditions maintain is shown in Table 1. At good statement, UDP are kept during 33.93 s, TCP are maintained just 1.3 s. However, at bad statement, UDP are kept during 9.3 s, TCP are maintained 6.23 s. It is the best that good statements are maintained longer and bad statements are maintained shorter. But, the threshold of UDP is 14 Mbps is higher than the one of TCP 9 Mbps.

Table 1. Sustainment Time of each condition

			Second
UDP	Avg. Bitrate: 17.2 Mbps	$S.Time_{gb}$	33.93 s
	Threshold: 14 Mbps	$S.Time_{bg}$	9.3 s
TCP	Avg. Bitrate: 11.7 Mbps	$S.Time_{gb}$	1.3 s
	Threshold: 9 Mbps	$S.Time_{bg}$	6.23 s

4.2 Distribution of J_i

Measured J_i fits to the Erlang distribution. In general, J_i of UDP is smaller than TCP, because UDP has higher MAB. J_{i_UDP} and J_{i_TCP} are both right-skewed, but J_{i_TCP} has heavy-tail. The heavy-tail means that probabilities were not concentrated on average, and were spread both sides. This characteristic makes hard to estimate MAB, and is represented in Fig. 4. In this graph, inter arrival times of UDP are concentrated from 380 ms to 526 ms, but inter arrival times of TCP don't have appropriately concentrated point. We show that J_{i_UDP} and J_{i_TCP} are fit $k_{UDP} = 3, \mu_{UDP} = 25$ Erlang distribution and $k_{TCP} = 3, \mu_{TCP} = 80$ Erlang distribution. Consequently, the UDP protocol is suitable for the adaptive streaming, because of easier estimation.

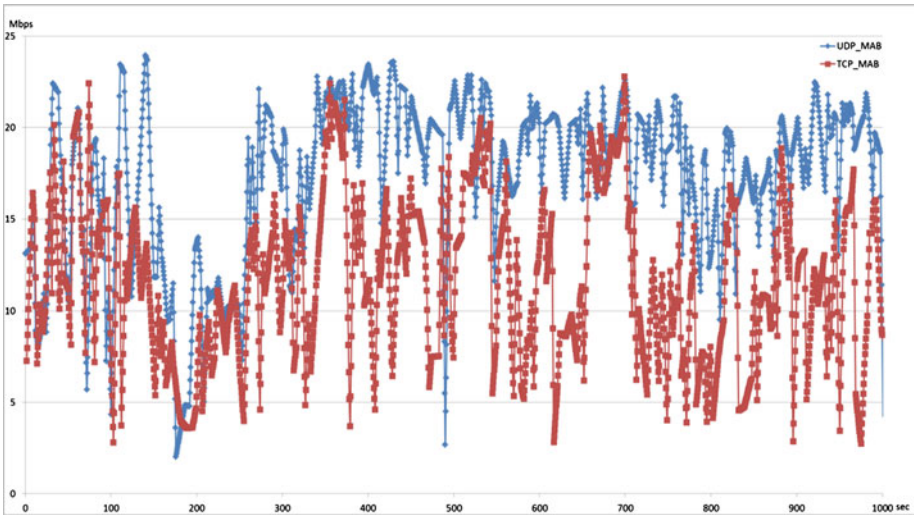


Fig. 3. Maximum available bitrate of UDP and TCP

4.3 Instantaneous Change of Rate

If the MAB maintains steady slope, it means that channel condition is gradually bad or good. Especially the slope is kept zero, it means that channel condition does not change. So when the slope is retained some values, estimating MAB using previous values is not difficult. However, when the slope is changeable, estimating MAB is more difficult. Then estimation error probability is increase.

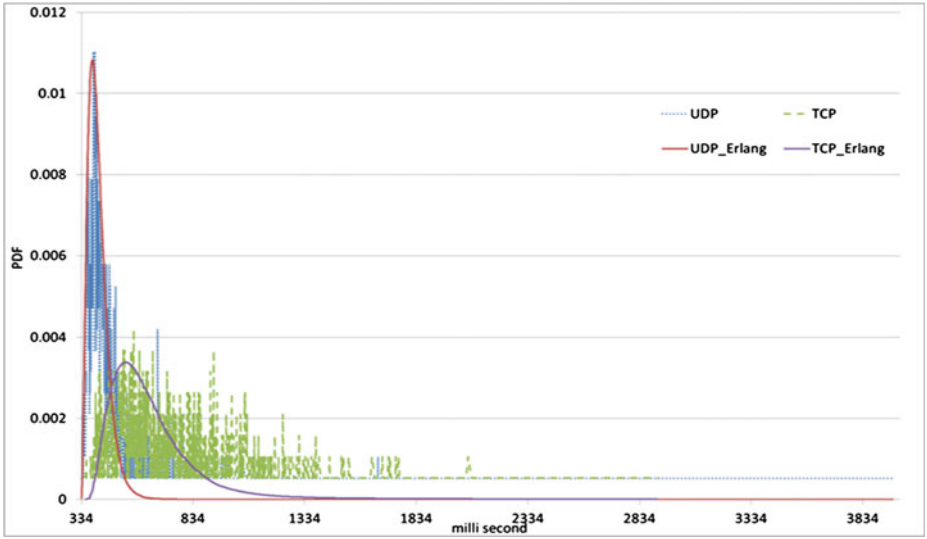


Fig. 4. A PDF of the inter arrival time

In this section, we propose second order differential of MAB for analysis which is easier to estimate MAB TCP or UDP. By using measured MAB, we draw second order differential of MAB graph at Fig. 5. for analysis

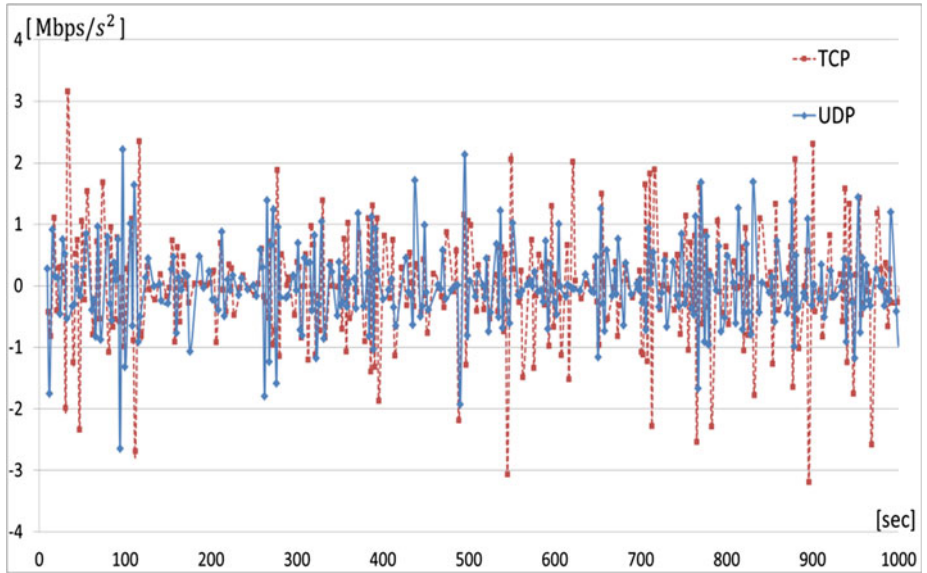


Fig. 5. Second order differential of MAB

For easy to view, the sum of absolute second order differentials at interval 100 s are represented in Table 2.

Table 2. Second order differential of each time section

TIME Section	100	200	300	400	500	600	700	800	900	1000	SUM
UDP	18.75	11.06	13.59	17.07	12.59	10.61	9.385	16.72	15.1	17.91	142.8
TCP	26.24	12.85	11.98	23.72	14.52	20.39	12.39	28.58	23.28	24.28	198.6

We can acquire that the sum of absolute second order differentials of UDP is smaller than one of TCP, except for TIME Section from 300 ms to 400 ms, also total the sum of UDP is smaller too.

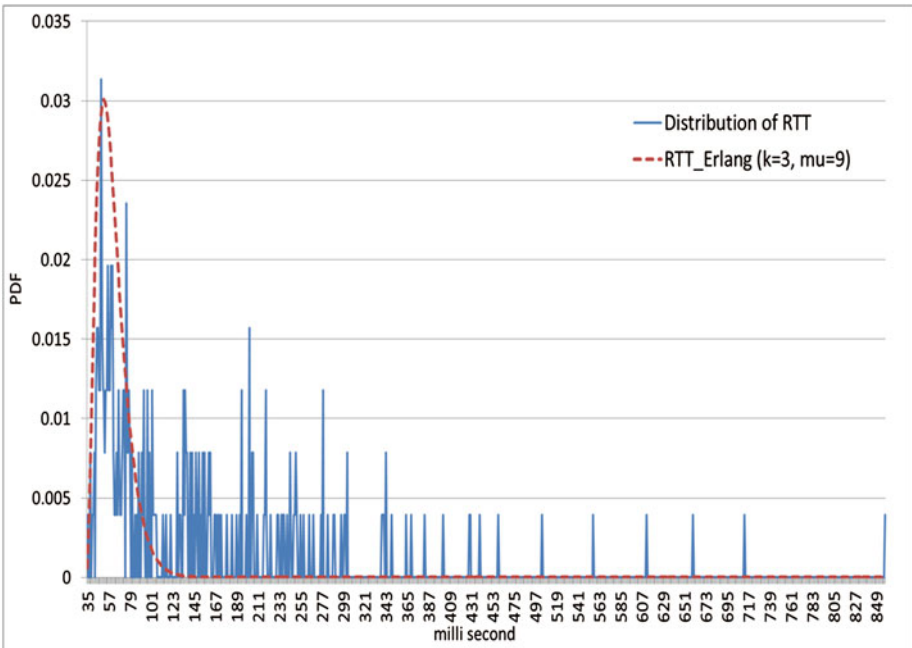


Fig. 6. A PDF of Round Trip Time following the Erlang distribution

4.4 Correlation

Before analyzing correlation, Fig. 6. is distribution of RTT with fitting graph following the Erlang distribution ($k = 3, \mu = 9$). Most of value is not more than 200 ms.

We analyze correlation between the MAB and the RTT. First of all, covariance $\text{Cov}(MAB_{UDP}, MAB_{TCP})$ and correlation $\rho_{MAB_{UDP}, MAB_{TCP}}$ are calculated by the equation in correlation part of Sect. 3.1 (9) and (10).

$$\text{Cov}(MAB_{UDP}, MAB_{TCP}) = 6.062, \rho_{MAB_{UDP}, MAB_{TCP}} = 0.29$$

And correlation values between RTT and TCP as well as UDP are calculated below.

$$\text{Cov}(MAB_{TCP}, \text{RTT}) = 5.254, \rho_{MAB_{TCP}, \text{RTT}} = 0.009$$

$$\text{Cov}(MAB_{UDP}, \text{RTT}) = -30.026, \rho_{MAB_{UDP}, \text{RTT}} = -0.055$$

We expected that the MAB has correlation with the RTTs, but correlations almost zero. So we can conclude the MAB is scarcely correlated with the RTT.

5 Conclusion

In this paper, we compare end-to-end performance of the UDP and the TCP, by using LTE, which is the most commonly used. The UDP has four advantages in comparison with TCP. First, the MAB of UDP is higher than the one of TCP, so the UDP can transmit higher quality video. Second, the MAB estimation is easy at the UDP, because the 2nd order of differential of the UDP is smaller than the TCP. Third, maintenance of state at the UDP is longer than the TCP. It means that the video quality of UDP is maintained. It is comfortable to watch for user. Last, Distribution of the MAB_{UDP} is right-skewed.

In conclusion, above advantages represent that UDP is suitable protocol for real-time streaming service. In future works, we will design the MAB estimation algorithm, and using the algorithm with Forward Error Correction (FEC), confirm the UDP-based streaming service is better than TCP.

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