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User Datagram Protocol Parametric Analysis for Maximum Transmission Unit in High-Definition Video Transmission for 1080p and 720p

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ABSTRACT

This article analyses the parametric used for Maximum Transmission Unit (MTU) in User Datagram Protocol (UDP) for online video streaming based on 1080p HD video and 720p HD video. Using a one-to-one node network design, this article has offered an investigation of the performance of UDP in NS3. UDP experiences rather considerable packet losses during the simulation procedure by contrasting the pertinent performance indicators. The selection of the video streaming applications over UDP as an example relates the requirements for group video conferencing and the computation of packet sizes. Understanding the simulation's results can help network administrators deal with network problems involving video streaming, video conferencing, or any other applications that might be connected to the associated protocol.

Keywords:
User datagram protocol; maximum transmission unit; video streaming; NS3; throughput; delay; packet loss

1. Introduction

In today's digital era, high-definition video became a fundamental part of everyone’s daily lives by engaging in remote video conferences or enjoying online gaming [1]. The seamless transmission of high-quality video created new challenges for networking researchers. One of which is the optimization of data packet sizes for Maximum Transmission Unit (MTU) [2,3]. This significant parameter directly impacts the efficiency and quality of video streaming and conferencing. Therefore, it is important to analyse the MTU of HD video transmission to highlight the importance of finding the perfect balance between packet size and transmission efficiency [4,5].

The primary objective of this paper is to investigate and simulate the performance comparison of transport layer protocols, specifically focusing on the User Datagram Protocol (UDP) [6]. To achieve this, a network simulation was used to examine UDP's behaviour and efficiency. The outcomes of this simulation aim to provide deeper insights into the dynamics of UDP data transfer concerning various

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configurable parameters. The UDP is a widely adopted protocol for handling substantial data volumes, particularly in applications like video streaming. UDP is a connectionless protocol and simply does not have any mechanism for error-corrections [7]. However, UDP's simplicity and minimal overhead make it suitable for applications that prioritize low latency and high throughput, such as real-time audio and video streaming [8,9]. Being widely used by todays' streaming applications, it is necessary to understand UDP performance behaviour to comprehend the rate of failure and performances. In this review, a real scenario from conference videos application was selected to be simulated to comprehend the relations between performance and network requirements such as bandwidth allocation [10].

To investigate the network performance Network Simulator 3 was used in this study [11]. NS3 is an open-source software licensed under GNU GPLv2 license which means that it can be used, modified, and distributed for free [12]. This software is mainly used to simulate network activity. It is a discrete-event network simulator that helps to simulate network activity in sequences events of time and primarily used for research and educational purposes. It provides a flexible platform for simulating a wide range of network technologies, including wired and wireless networks, and supports various protocol stacks and traffic models [13]. To review and compare the behaviour of UDP in NS3, three network performance metrics were used during simulation process. The metrics elements are throughput, delay, and packet loss.

2. Design and Define the Parameters for the UDP Simulation

In order to analyse and to understand the performance of UDP transmission in NS3, this paper simulate a simple network topology consisting of two nodes connected by a link. The simulation process involves sending data packets of varying sizes and rates between the nodes using UDP with fixed parameters setup, and measure the resulting throughput, delay, and packet loss. The details on the parameters or setup that are executed in NS3 are as the following:

2.1 Parameters for UDP on 1080p HD 3.8 Mbps:

i. <u>Time:</u> To measure the simulation packets transmission on 3 different set of times. 2 minutes (120 seconds), 5 minutes (300 seconds) and 10 minutes (600 seconds).
ii. <u>Number of Packets:</u> represent the packets that actually send through the network for 3.8Mbps data streams. Table 1 shows the packet numbers which must be calculated based on the total bytes per second divided by MTU size.

Table 1
Number of Packets for 1st Test Environment

#	Time	Number of Packets
1	600	198,260
2	300	99,130
3	120	39,652

iii. <u>Bandwidth/Data Rate:</u> represent the medium of transmission based on Ethernet speeds available on today's infrastructure. 1000Mbps represents the 1Gbps Ethernet speeds transmission medium such as Cat6 cable, and 10,000Mbps is refer to 10Gbps ethernet speeds transmission such as Cat6a/Cat7 [14].
iv. <u>MTU:</u> represents the Maximum Transmission Unit (MTU) for each packet size being transmitted over the sequences of simulation in NS3 [15].

2.2 Parameters for UDP on 720p HD 2.6Mbps:

i. Time: To measure the simulation packets transmission on 3 different set of times. 2 minutes (120 seconds), 5 minutes (300 seconds) and 10 minutes (600 seconds).
ii. Number of Packets: represent the packets that actually send through the network for 2.6Mbps of data streams. Table 2 shows the packet numbers which must be calculated based on the total bytes per second divided by MTU size.

Table 2
Number of Packets for 2nd Test Environment

#	Time	Number of Packets
1	600	135,652
2	300	67,826
3	120	27,130

iii. Bandwidth/Data Rate: represent the medium of transmission based on Ethernet speeds available on today's infrastructure. 1000Mbps represents the 1Gbps Ethernet speeds transmission medium such as Cat6 cable, and 10,000Mbps is refer to 10Gbps ethernet speeds transmission such as Cat6a/Cat7.
iv. MTU: represents the Maximum Transmission Unit (MTU) for each packet size being transmitted over the sequences of simulation in NS3.

2.3 Simulation Network Topology in NS3

The network topology used for this simulation consists of two nodes (Node 0 and Node 1) connected by a point-to-point link between the NetDevice [16]. The NS3 simulator will transmit data packets from Node 0 to Node 1 using User Datagram Protocol with different parameters as mentioned, and the performance for the specific time are being analysed. Figure 1 illustrate the node-to-node topology for the UDP simulation.

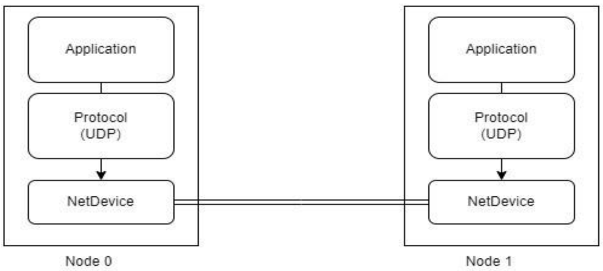


Fig. 1. Node to Node Topology for UDP Simulation

The NS3 configuration for the network topology design are as the following code-snippet [17]:

i. Defining port number for UDP protocol, total of packets and the size of packets for each transmission. The total of PacketCount is set according to the environment setup:

```
uint16_t port = 4000; uint32_t
PacketCount = 39652; uint32_t
PacketSize = 1472;
```

ii. Defining time taken to simulate the packets in minutes:

```
float duration = 120.0;
```

iii. Creating node 0 and node 1:

```
NodeContainer n;
n.Create (2);
```

iv. Building ethernet like channel with data rate value for 1Gbps and 10Gbps:

```
CsmaHelper csma;
csma.SetChannelAttribute       ("DataRate",       StringValue       ("10000Mbps"));
csma.SetChannelAttribute      ("Delay",      TimeValue      (MicroSeconds      (2)));
csma.SetDeviceAttribute ("Mtu", UintegerValue (1500));
NetDeviceContainer d = csma.Install (n);
```

2.4 UDP Simulation Implementation Setup

For the simulation setup, there 12 different C++ source files are created with specific parameters. Table 3 below shows the information regarding the simulation setup:

Table 3
Simulation Setup

#	Test Environment	Parameters	Objectives
1	Test1-720-120-1G	720p HD 120 seconds 1000 Mbps/1Gbps	To simulate 720p HD packets file for 120 seconds via 1Gbps Ethernet channel.
2	Test2-720-300-1G	720p HD 300 seconds 1000 Mbps/1Gbps	To simulate 720p HD packets file for 300 seconds via 1Gbps Ethernet channel.
3	Test3-720-600-1G	720p HD 600 seconds 1000 Mbps/1Gbps	To simulate 720p HD packets file for 600 seconds via 1Gbps Ethernet channel.
4	Test4-720-120-10G	720p HD 120 seconds 10000 Mbps/10Gbps	To simulate 720p HD packets file for 120 seconds via 10Gbps Ethernet channel.
5	Test5-720-300-10G	720p HD 300 seconds 10000 Mbps/10Gbps	To simulate 720p HD packets file for 300 seconds via 10Gbps Ethernet channel.
6	Test6-720-600-10G	720p HD 600 seconds 10000 Mbps/10Gbps	To simulate 720p HD packets file for 600 seconds via 10Gbps Ethernet channel.
7	Test7-1080-120-1G	1080p HD 120 seconds 1000 Mbps/1Gbps	To simulate 1080p HD packets file for 120 seconds via 1Gbps Ethernet channel.
8	Test8-1080-300-1G	1080p HD 300 seconds 1000 Mbps/1Gbps	To simulate 1080p HD packets file for 300 seconds via 1Gbps Ethernet channel.

9	Test9-1080-600-1G	1080p HD 600 seconds 1000 Mbps/1Gbps	To simulate 1080p HD packets file for 600 seconds via 1Gbps Ethernet channel.
10	Test10-1080-120-10G	1080p HD 120 seconds 10000 Mbps/10Gbps	To simulate 1080p HD packets file for 120 seconds via 10Gbps Ethernet channel.
11	Test11-1080-300-10G	1080p HD 300 seconds 10000 Mbps/10Gbps	To simulate 1080p HD packets file for 300 seconds via 10Gbps Ethernet channel.
12	Test12-1080-600-10G	1080p HD 600 seconds 10000 Mbps/10Gbps	To simulate 1080p HD packets file for 600 seconds via 10Gbps Ethernet channel.

2.5 Tracing Implementation

To collect performance data during the simulation, there are additional source code written to perform and calculate the total of Throughput, Packets size, Average Delay and Packet Losses [18]. The simulation output files are also being logged to the NS3 root directory for references. The log files for each test varying size from 405KB to 17,571KB. Below is the example of additional run command added to create the simulation logs:

Example Command:

./waf --run scratch/(simulationfilesname) 2>&1 | tee log-udp-sim2-6001G.txt

3. Results

After the test scenarios has been successfully conducted on the 12 different test environments that is mainly for 1080p HD with 3.8Mbps transfer rate data, and 720p HD with 2.6Mbps transfer rate data. The results for both of these tests are shown in Table 4 and Table 5 below.

The UDP simulation results for 1080p HD:

Table 4
UDP simulation test on 1080p HD 3.8Mbps over 1G and 10G ethernet

#	Simulation Test	Time (s)	Total of Bytes	MTU	# Packets (Bytes)	Bandwidth / Data Rate = 1000 Mbps			Bandwidth / Data Rate = 10000 Mbps		
						Throughput (Mbps)	Average Delay (s)	Loss packets (Bytes)	Throughput (Mbps)	Average Delay (s)	Loss packets (Bytes)
1	1080p HD 3.8 Mbps	600	291,480,000	1472	189,260	776.336	0.133	148,049	3073.71	0.032	30,093
		300	145,920,000	1472	99,130	776.435	0.119	69,091	3071.92	0.027	10,844
		120	58,368,000	1472	39,652	776.511	0.097	21,715	3073.52	0.013	2,334

The UDP simulation results for 720p HD:

Table 5
UDP simulation test on 720p HD 2.6Mbps over 1G and 10G ethernet

#	Simulation Test	Time (s)	Total of Bytes	MTU	# Packets (Bytes)	Bandwidth / Data Rate = 1000 Mbps			Bandwidth / Data Rate = 10000 Mbps		
						Throughput (Mbps)	Average Delay (s)	Loss packets (Bytes)	Throughput (Mbps)	Average Delay (s)	Loss packets (Bytes)
2	720p HD 2.6 Mbps	600	199,680,000	1472	135,652	776.378	0.126	98,182	3071.74	0.03	17,972
		300	99,840,000	1472	67,826	776.518	0.111	44,158	3073.64	0.022	4,732
		120	39,936,000	1472	27,130	776.506	0.088	11,741	3074.41	0.009	2,333

From the tables, logs and results shown in the NS3, the data has been analysed and several comparisons on the performance metrics visually translated to the graph as below:

3.1 Throughput

Figure 2 shows that the lower data streaming (720p HD) utilize the 1000Mbps bandwidth more efficiently on 600 seconds and 300 seconds during the simulation time. While for 120 seconds, there are slightly different where it shows the 1080p throughput rates is higher than 720p.

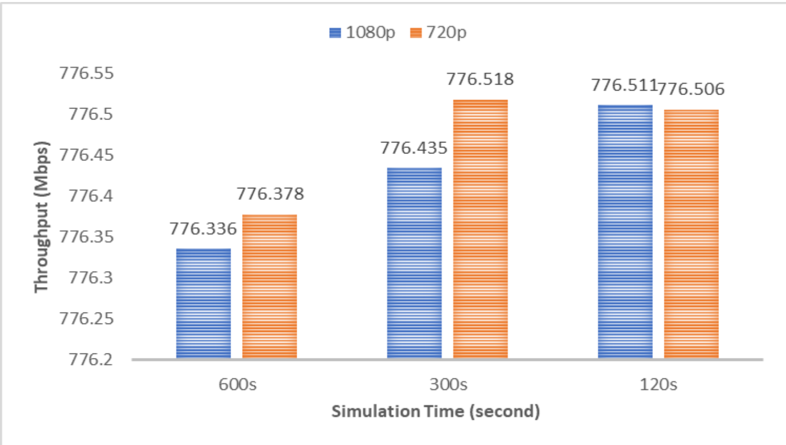


Fig. 2. Throughput rates comparison between 1080p HD vs 720p HD on 1000Mbps bandwidth

Figure 3 shows that the lower data streaming (720p HD) again utilizes the 10,000Mbps bandwidth more efficiently on 300 seconds and 120 seconds during the simulation time. While for 600 seconds, there are inconsistence of throughput rates that shows 1080p has a better throughput than 720p.

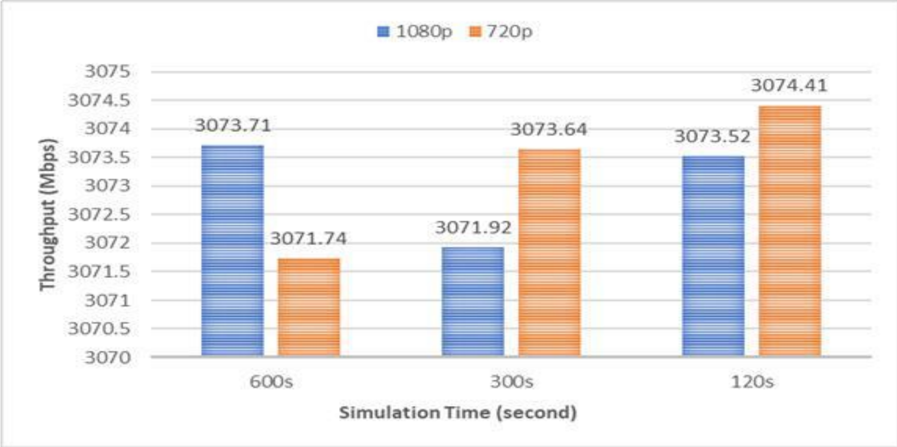
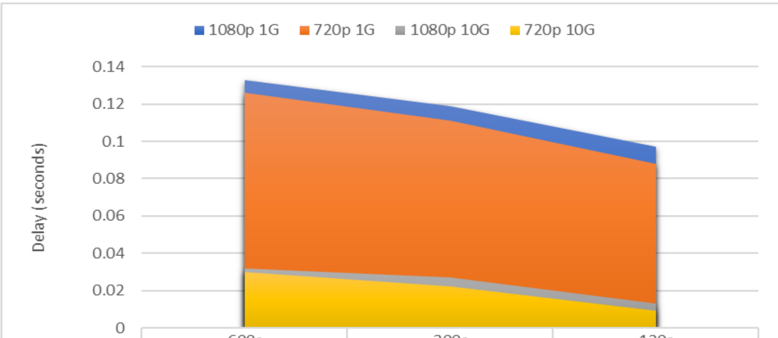


Fig. 3. Throughput rates comparison between 1080p HD vs 720p HD on 10,000Mbps bandwidth

Based on the two findings above, it shows that lower data streaming 720p have a good throughput rate over two different ethernet bandwidth. There are two inconsistencies shown in the graph that need to be further study.

3.2 Delay

Figure 4 shows the average delay in seconds for corresponding totals of packets after transmitted. The smallest delay is shown by 720p data being transmitted over 10,000Mbps bandwidth. While the same number packets for 720p being the third ranks if transmitted over 1000Mbps bandwidth. The biggest delay in these simulations is shown by 1080p data transmitted over 1000Mbps bandwidth with 0.133 seconds. By analysing the data, it shows that bigger bandwidth can accelerate the data transfer and reduce the latency and average delay.



	600s	300s	120s
1080p 1G	0.133	0.119	0.097
720p 1G	0.126	0.111	0.088
1080p 10G	0.032	0.027	0.013
720p 10G	0.03	0.022	0.009

Fig. 4. Average delay for different bandwidth size

3.3 Packet Loss

Figure 5 shows that a significant of packet loss happens in the NS3 simulation. The most packet loss is on 1080p over 1Gbps bandwidth simulation. The smallest packet loss is on 720p over 10Gbps bandwidth.

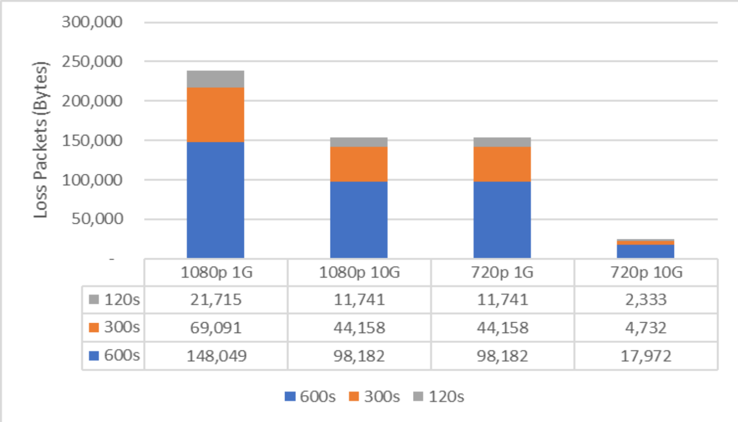


Fig. 5. Packet Loss Comparison for 1080p and 720p on different time sets and bandwidth

It is clearly shows that in the simulation over the ethernet channel conducted, the packet losses are certain and undeniable. In this figure, it also indicates that:

i. Larger packets transfer over small bandwidth will result in more packet losses.
ii. Packet losses are accumulated over a time period. Longer simulation times will accumulate more packet losses.
iii. Packet loss is significantly improved over a bigger bandwidth with less packets transfer over a shorter time.

This unreal situation happens to packet losses lead to another parameter changes in the simulations. Two more test scenarios have been executed by tweaking the value of Maximum Transmission Unit (MTU) as the following:

Table 6
UDP simulation test over 10G ethernet with 600 seconds parameter and MTU size reduction.

#	Simulation Test	Time (s)	Total of Bytes	MTU	# Packets (Bytes)	Bandwidth / Data Rate = 1000 Mbps		
						Throughput (Mbps)	Average Delay (s)	Loss packets (Bytes)
1	1080p HD 2.6 Mbps	600	291,480,000	1050	227,942	2429.41	0.029	23,381
2	720p HD 2.6 Mbps	600	199,680,000	1050	190,171	2429.23	0.026	13,383

Refer to the Table 6, for the first simulation on 1080p HD, total packets for the simulation are being increases by 40%. This is because of the total of bytes is divided by the new value of MTU (1050) will create smaller packets size thus increases the totals of packets for the simulation. The value of the existing packets is 198,260 increased to 277,942 bytes after the MTU size adjustment. The calculation to get the number of packets for the NS3 simulation can be determine by using this formula:

$$Number\ of\ Packets\ = \frac{Total\ of Bytes}{MTU\ Size}$$

After the simulation run-test, the results after MTU reduction by 30% shows some a great improvement on total of packet losses. Percentage of losses for Simulation Test #1 (as referred to

Table 6) has been reduced to 8% from existing 15%. The total of packets losses is reduced by 6,712 packets. The following Figure 6 shows the existing percentage of losses in details.

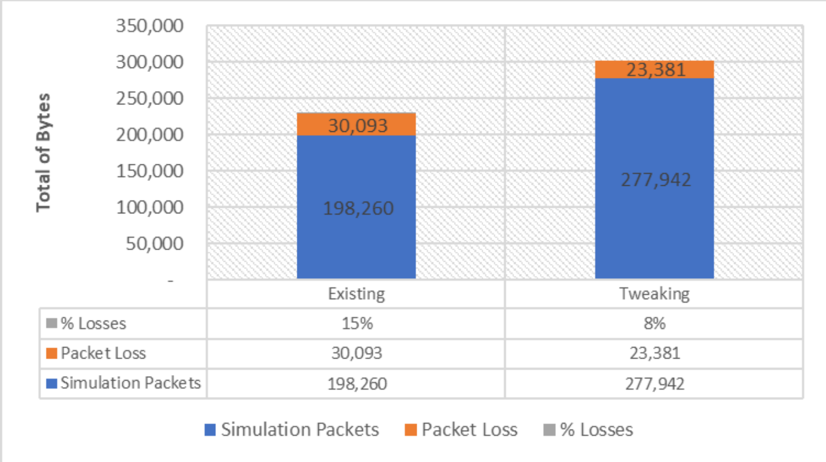


Fig. 6. Total of Packet Loss Reduction for 1080p HD over 10Gbps after MTU size tweak

As the results show from the NS3 simulation, UDP transmission data losses can be reduced significantly by providing the correct MTU size. In this example, the MTU size is reduced by 30% and will help to lower the packet losses by 7%.

Based on the results of the simulation, the following conclusions about the performance metrics of UDP applications in NS3 are:

i. Throughput: Better throughput can be achieved by minimizing the total of data packets. For example, in this simulation, UDP for 720p HD has higher throughput rather than 1080p HD.
ii. Delay: Higher bandwidth will help to provide a small delay of time. A small bandwidth with higher data rates over the network will cause higher delay or latency.
iii. Packet Loss: UDP experiences higher packet loss as it has no collision detection and avoidance, it does also not guarantee the reliable delivery of the data packets. In this simulation, minimal or reduced rates of data losses can be achieved by reducing the packet size transmitted over the ethernet for example by tweaking the MTU size.

Overall, in this simulation it shows that UDP provides higher throughput and lower delay if the total of packets transmitted over the ethernet is reduced. It also shows that an efficient packet sizing mechanism such as the determination of MTU size for specific applications is necessary to improve packet losses in UDP.

4. Conclusions

In conclusion, this paper has conducted an evaluation of UDP performance within the NS3 framework, utilizing a one-to-one node network topology. Through a comparative analysis of relevant performance metrics, it becomes evident that UDP experiences significant packet losses during the simulation. To illustrate the implications of these findings, we have chosen video streaming applications over UDP as an example, highlighting their relevance to group video conferencing requirements [19,20] and packet size calculations. The insights gained from this simulation are invaluable for Network Administrators when addressing network issues that pertain to video streaming, video conferencing, or any applications associated with this protocol.

The outcomes of this simulation underscore the critical importance of a high-performance infrastructure for contemporary applications, enabling organizations to seamlessly execute their daily business processes and tasks.

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