Impact of Buffer Size on UDP Performance for Real-Time Video Streaming Application

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Summary
Live communication which is also known as an RTC (Real-Time Communication) occurs inside the bounds of time. Buffer sizes usually affect the overall network performance. Thus, the impact of homogeneous router buffer size is analyzed in this work. Simulation results show that average throughput and the delay which is from End-to-End does change if different routers' buffer sizes are increased. The increase in End-to-End latency or delay is ordinarily not worthy for applications such as real-time applications but at the same time it enhances throughput. However, throughput diminishes with lower buffer size, but also reduces the delay. Thus, intermediate range of buffer size from 40 to 30 frames is suggested for optimal throughput and lower End-to-End delay.

Key words:
UDP, Video Streaming, Buffer Size.

1. Introduction

Live communication which is also known as an RTC (Real-Time Communication) occurs inside the bounds of time. A buffer is utilized to maintain impermanent congestion. Packets are usually hold by the buffer which is a short time storage memory. Video streaming is one of the real-time applications which are time delicate. In the event that delay increments by a specific threshold, data will be lost. Transmission delay, processing delay, and queuing delay are the main components of End-to-End latency. More packets will hold up in a line with increase in queueing delay which occurs due to expansion in buffer size. One of the cases of an RTC application is the video streaming. In a live communication, if information gets towards the end after its time limit then these communications are pointless and estimated to be lost [1]. For the live communication, there should be low latency and high throughput. This work is based on the buffer size variation at UDP protocol which involves the transport layer of internet protocol suite. Many researchers have proved that UDP is preferable for real-time communication while analyzing different parameters. Thus, the authors in [2] suggested that buffer size should not be used according to the rule or known as rule of thumb as it increases buffer size and it could be much reduced in terms of packets without degradation in performance but with small bandwidth utilization which could be better trade-off for the all optical routers.
In [3], it was demonstrated that the measure of a buffer size can be determined by the loss synchronization and traffic's round-trip times. [4] dissected outcomes for packet loss and throughput by changing buffer sizes of routers; they took both UDP and TCP as the real-time and non-real-time applications (live and non-live) respectively for important two queue managing methods known as DT (Drop Tail) and RED (Random Early Detection). It was brought up by considering the simulation outcomes that throughput and efficiency in regard to TCP application improves accompanied by the buffer size expansion for RED. Moreover, by expanding the size of router buffer, packet loss under UDP application lessens.
Under this research work, End-to-End latency/delay and throughput have been investigated for real-time (live) video streaming under variety of settled sizes of the buffer of packet switches of the network.

2. Performance Indicators

The best execution of the network in terms of keeping up its performance is done by discovering the routing path by utilizing the routing mechanisms. By taking up the accompanying metrics, the ability of the transport protocols can be estimated.

2.1 Average End-to-End Latency/Delay

The delay or latency is a parameter which can be defined as how much amount of aggregate time for the conveyance of packets from the origin to the target is required. It consolidates all types of delays, for example, packet transmission time, queueing time for packet, and the propagation time [5]. Delay is influenced by steady Bit Error Rate (CBR) high rate. Buffers turn out to be quicker in light of the fact that transmission packets should be put away in buffers for quite a while and is estimated in the second [6].

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2.2 Average throughput

Average throughput gauges how expedient the original information is sent by means of a network. Average throughput is ascertained as a proportion of the quantity of aggregate transmitted bits to recipient and sent by a sender and what amount of time is required to receive last packet; bps is its unit.

$$Throughput = \frac{Entire\ Transmitted\ bits}{Duration\ of\ Obsevation} \quad (1)$$

2.3 Router's Buffer Size

In a packet switch network, buffer is a principal component which stores impermanent data at whatever point the system is occupied. The appropriate size estimation of these buffers is a critical and open research predicament [3] [7].

3. Network Architecture

Network Descriptive (NED) language of OMNET++ has been used to develop the network architecture shown in Fig. 1. It consists of twenty-four (24) hosts and nine (09) routers. Video streaming traffic type is considered in the network shown in Fig. 1.

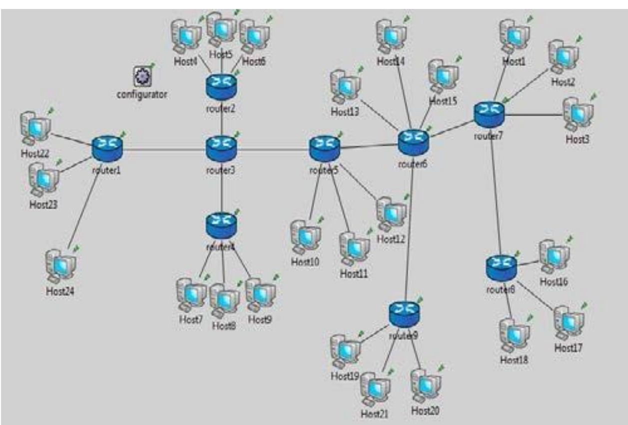

Fig. 1 Network architecture of 24 hosts and 09 routers to view point to point connection.

3.1 Simulation setup for video streaming

One of the application of multimedia is the streaming video technology which is used for audio and video transmission over the web in real-time. Because of the no requirement for the file to be completely downloaded, the document is run while portions of the record are being recognized and decoded only in this mode. Live content, for example, a political debate, any sports match, or a concert can be sent [8]. Simulation parameters for video streaming are shown in Table 1.

By taking into account the simulation parameters shown in Table 1, distinct simulations are carried out for performance assessment using diverse router buffer sizes by sending traffic individually on the similar network i.e., video streaming with buffer size of router for all routers from large- to-small. The results are examined by carrying out simulations for variable buffer size of the router. The developed network consists of 09 routers with 24 nodes and initially 100 frames as a buffer size is set for all routers and then buffer size is reduced up to 20 frames. It ought to be noticed that there are 12 sending and 12 receiving nodes respectively, out of which 12 transmitting nodes are constantly sending video streaming type information packets towards the receiving nodes. Although, the size of router buffer is varied but network topology is fixed for the considered network.

Table 1: Video streaming simulation set up

Parameter	Value
Number of Nodes	24 (12 sender, 12 Receiver)
Video size	1000 MB
No. of frames (Buffer size)	20, 40, 75, 100
Data Rate	2 Mbps
Length of a Packet	10,000 bytes
Simulation time	10,000 Seconds (Net traffic time)
No. of routers	9
Frame capacity	4475
Start time	1 Sec.
Connection Type	PP (Point-to-point)
Propagation delay	100 ms
Send interval	5 Seconds
Bit error rate	0.000001

4. Simulation Results and their Analysis

Firstly, the network architecture for the purpose of simulation was developed with 12 sending and 12 receiving nodes (total 24 nodes) respectively where the nodes having number from 13 to 24 are the transmitting nodes and receiving nodes are numbered from 1 to 12 separately. The network has been developed in the OMNET++ in NED (Network Descriptive) language which contains many libraries built on INET Framework [9] [10]. 100 frames as a fixed size of all buffers of router have been considered. In the simulation set up, communication of video streaming type has been taken into account and primarily screen and assess the service quality with divergent router buffer sizes. Furthermore, the model is made on a similar network, taking into account only the buffer size of middle routers,

for example, router number 3,4,5,6,7, and 9. The sizes of routers under investigation are varied as 100, 75, 40, and 20 frames respectively. There are add up to total 09 routers in a system topology, however, just have been considered 06 fundamental routers to change their buffer sizes. Indicators such as delay (End-to-End) and throughput are investigated for comparing the performance of UDP transport protocol.

4.1 Analysis of delay and throughput with variety of packet switch (router) buffer sizes

All transitional principle switch buffer sizes are settled from 100 frames to 20 frames to examine the delay or latency (End-to-End) in Fig. 2 whereas throughput is shown in Fig. 3 respectively. Clearly, the results show that when 100 frames buffer size is considered, End-to-End delay is quite high and cannot be acceptable in video streaming application, however, throughput is maximum. With the decrease in router buffer size in terms of frames from 100 to 75, there is no effect on both the parameters i.e., throughput and latency and the results are overlapping with each other. It means this buffer size is also not acceptable for video streaming. Furthermore, when the buffer size is further decreased from 75 to 40 frames, still delay is quite high and not to be acceptable, but throughput is still high and acceptable for the video streaming. Finally, when buffer size has been decreased from 40 to 20, maximum delay is 0.9 milli-seconds which is acceptable for the video streaming but at the same time, throughput has decreased to 78.8 bps which is not acceptable in video streaming application.

4.2 Analysis of delay and throughput with variety of packet switch (router) buffer sizes from 08 to 100 frames for all middle routers

In Fig. 4 and Fig. 5, all mentioned transitional buffer sizes of routers are varied from 08 frames to 100 frames and analyzed the throughput and latency or delay in average for video streaming application. In the scenarios which are displayed in Fig. 4 and Fig. 5 respectively, it is quite clear that latency or delay form End-to-End is increasing according to the buffer size of the packet switch (router) but at the same time, throughput is also improving. In these situations, it can be summarized that real-time communication problem cannot be tackled with high buffer size because of the high End-to-End delay, however, then again, one preferred standpoint of high buffer size is the most extreme throughput. Additionally, it is watched that diminishing size of the buffer diminishes both the latency and the throughput. Thus, buffer size should be taken between 40 to 30 frames for getting an acceptable delay/latency from End-to-End and an optimized throughput.

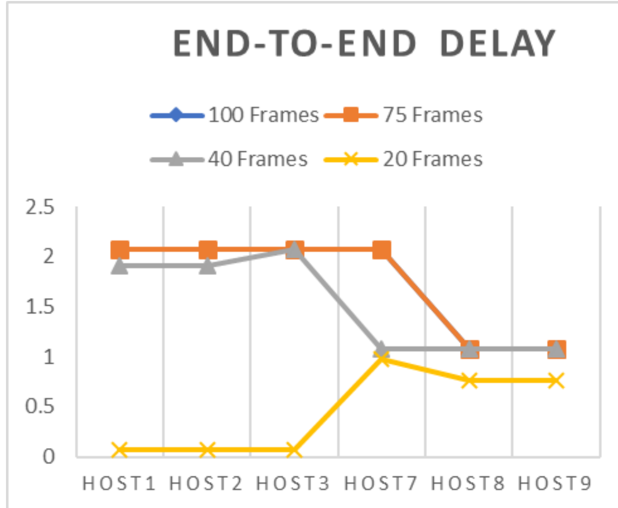

Fig. 2 Buffer size of Router (Packet Switch) vs delay/latency (End-to-End) for an application of video streaming.

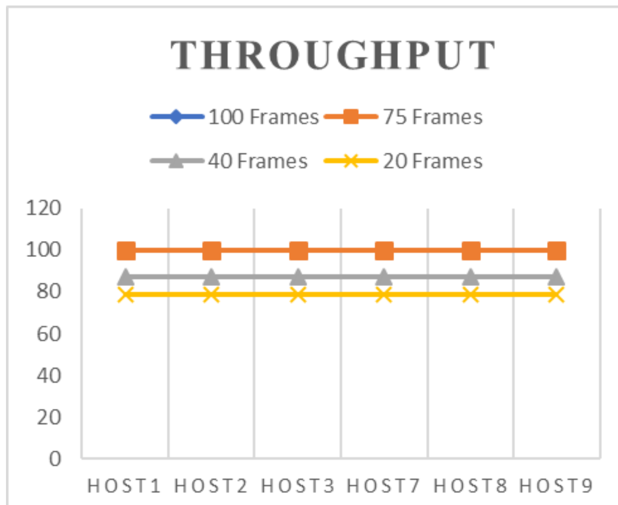

Fig. 3 Buffer size of Router (Packet Switch) vs Throughput for an application of video streaming.

5. Conclusion

In this work, impact of homogeneous router buffer size was analyzed. It was shown through simulation results that buffer sizes usually affect the overall network behaviour in terms of performance. Throughput and the delay or latency (End-to-End) changed if different routers' buffer sizes were increased. By considering large buffer sizes, delay or latency from End-to-End increased and is ordinarily not worthy for real-time applications (RTC) but rather enhanced the throughput. However, throughput diminished in the case of low buffer size but

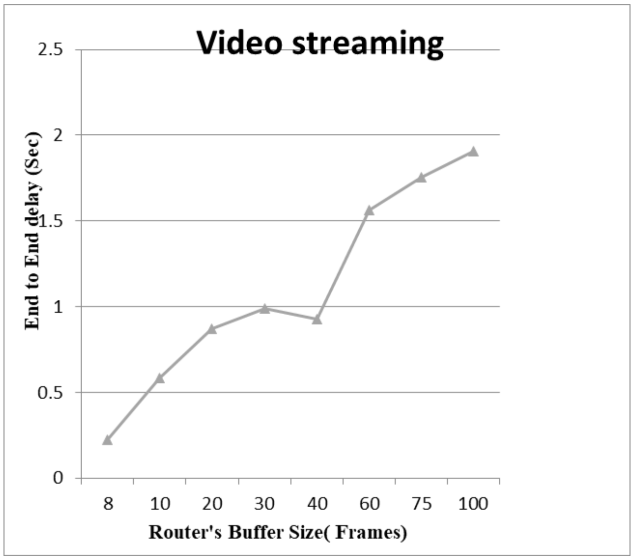

Fig. 4 Packet Switch (Router) buffer sizes vs Average latency/delay (End-to-End) from 08 to 100 frames for video streaming application.

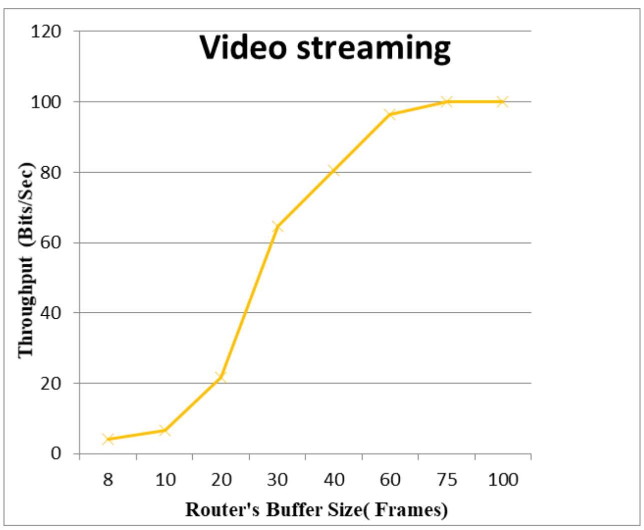

Fig. 5 Packet Switch (Router) buffer sizes vs Average throughput from 08 to 100 frames for video streaming application.

latency or delay (End-to-End) also reduced. Thus, intermediate range of buffer size from 40 to 30 was suggested for an acceptable lower latency/delay (End-to-End) and an optimal throughput.

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