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Performance Evaluation of Unicast Networks

Using Different Queuing Protocols

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Abstract—**This study is conducted to analyze the performance of unicast networks by evaluating different queuing protocols under different criteria using multimedia data. That was achieved by using the NS-2 tool. The data traffic used in the simulation was CBR over UDP traffic because it is representative of multimedia data in the NS-2. Protocols tested in the simulation were FIFO, FQ, RED, DRR and SFQ. These protocols, including its performance, were evaluated and compared to each other with respect to the packet loss ratio, dropping fairness, end-to-end delay and delay jitter. Also, this research provides background and some related literature for the performance of unicast networks including queuing algorithms, performance metrics, congestion avoidance and quality of service. Overall results of the study show that FQ, DRR and SFQ were providing better performance compared to other protocols. However, the best three protocols had different orders based on the performance metric conducted. The scope of the research focuses only on unicast wired networks, so this work is not applicable for multicast or wireless networks. The proposed model shows how results were collected and evaluated. Finally, the purpose of using multimedia data is to observe the behavior of each protocol under heavy traffic.**

I. INTRODUCTION

The use of the Internet has become part of our regular life. For example, it is used for buying and selling products, making reservations for hotels and flights, and communicating with other people. [8] [9]. Since the Internet consists of smaller networks that are connected to each other, the basic idea of any network is point-to-point communication, also known as unicast [21]. In other words, a message goes from one node to another, finding its way through the network. Also, in 1988, the IP multicast was invented, which involves sending data from one source to multiple destinations [40]. The performance of unicast and multicast networks is very challenging because many aspects are involved [34] [35] [36]. Each application has its own requirements that need to be met in order to reach the maximum available performance; for instance, mail servers can be fixable in case of delaying messages [10], whereas real-time applications must have assurance of delay, or they will perform poorly. So, quality of service, performance metrics and congestion avoidance are very effective and must be taken into consideration [3] [6].

II. UNICAST

Unicast is simply point-to-point communication. It is when one source node transmits traffic to one destination node [28]. One characteristic of unicast is that it moves in one direction at a time; that is, if a node is sending some traffic to another node, the channel will be occupied so the receiver node will be unable to respond to the sender until the sender finishes transmitting its data [36].

III. NETWORK PERFORMANCE

Network performance is a very important aspect because some new technologies cannot be used or perform poorly due to the huge amount of data they need in order to achieve their targets [4]. Moreover, network performance can be divided into several categories including queuing algorithms, TCP congestion control, congestion avoidance, quality of service and performance metrics [39].

A. Queuing Protocols

Queuing protocols are how packets are treated inside routers' buffers and how they are sent to their destinations. Also, the decision of which packet will be sent and which packet will be dropped is another mission for the protocol [11], including how long the packet will wait inside the buffer before it is transmitted. The most famous queuing protocols are FIFO, FQ, SFQ, DRR and RED. FIFO: first in, first out, meaning that the first flow of packets received by the router is the first flow that will be transmitted to its destination. FQ: fair queuing is using round-robin service to transmit packets from different flows in turn. SFQ: stochastic fairness queuing is based on FQ but uses a fixed number of flows. DDR: Deficit round robin, it uses a special mechanism that calculates the deficit counter for a flow of packets. If it is greater than the packet size, the credit of the flow will be decreased by the value of the packet size; otherwise, the packet is going to be skipped and the credit of the flow will be increased by quantum value.

B. TCP Congestion Control

The idea here is to send packets to the destination without any notice. So, if the sender is getting acknowledgment from the receiver, that means the links still have more capacity; therefore, more packets can be sent. If no acknowledgment is received, that means packets are dropping, so no more traffic can take place. Theoretically, this approach is applicable, but in reality it is difficult to implement [25].

C. Congestion Avoidances

TCP congestion control reduces the congestion inside the network after it occurs, but congestion avoidance involves trying to prevent congestion from happening. RED: random early detection is the most famous protocol for congestion avoidance. It works by dividing the responsibility for control avoidance among routers and sources. So, if congestion is going to occur soon, the router is going to notify the sender by dropping one packet. So, the sender will reduce its sending rate to avoid dropping more of its packets later on [22].

D. Quality of Service

Quality of service is the ability of the network to meet the requirements and resources needed by applications. For example, real-time applications need their data to arrive on time, whereas some applications, such as email servers, do not require this kind of restriction [30]. So, quality of service is responsible for making decisions about providing resources to applications and how much of these resources will be used. The network should then respond to the application requirements by indicating whether it is able to provide the resources [27]. Resources of any network generally are divided between two aspects: bandwidth of links and buffer space in routers. The end nodes will communicate with routers, asking them for resources. Then routers will answer back to the end nodes by telling them the available resources. For instance, the service used on the Internet is called best effort service, and it does not guarantee any kind of resources assurance, which means all data or flows of packets are treated in the same manner and can be hit by network congestion [18]. So, this model is suitable for tolerant applications, such as mail servers, that do not require any assurance. This limitation led to the invention of new services to provide and enhance resource assurance. The most famous services are integrated services and differentiated services that are developed to improve quality of service.

E. Performance Metrics

Performance metrics indicate how reliable the performance of the network is. Performance metrics include bandwidth, throughput, utilization, delay, round-trip time, delay jitter and packet loss.

- Bandwidth: If there is a link that has a bandwidth of 10 Mbps, any data above this number will be dropped. In other words, bandwidth is how fast data can be transmitted per second in a link moving data [12].
- Throughput: How fast actual data can be transmitted per second in a link moving data [46]. Therefore, throughput is always less or equal to bandwidth.
- Utilization: Utilization is simply throughput over bandwidth. For example, if throughput is 6 Mbps and bandwidth is 10 Mbps, the utilization is 60%, meaning 40% of the link is still available [48].
- Delay : It is the total time duration needed by a packet to be transmitted from sender to receiver including queuing delay, propagation delay and finally transmission delay [1].
- Round-trip Time (RTT): It is the time needed for a packet to be received by a receiver plus the time needed for acknowledgment of the same packet to be received by the sender [38].
- Delay Jitter: Delay jitter is the variation between packets' delay. So if the maximum delay of a packet is 500 ms and the minimum delay of a packet is 300 ms, then the delay jitter will be 500 – 300, which is 200 ms [13].
- Packet Loss: It is when a packet is unable to reach its target for any reason; for example, not enough bandwidth, connection issues, human interference, or link failure, or a combination of two or more of these reasons [44].

IV. NS-2 (NETWORK SIMULATOR)

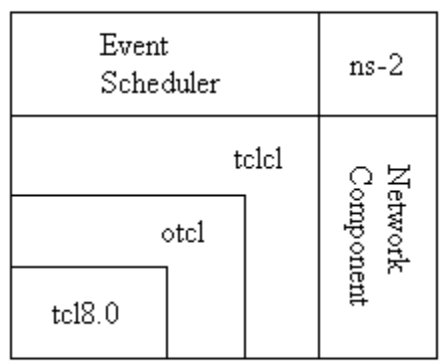

Fig. 1. NS-2 structure.

NS (version 2) is the network simulator that was used to collect the results of this study. It is object-oriented. It uses two programming languages, C++ and OTcl. It can be used for simulating either local or wide networks. Also, it has the ability to simulate wired and wireless networks, as well as simulating unicast and multicast networks. NS-2 is very difficult for a first-time user. There is a lot of documentation about how to use NS-2, but not many of them are friendly manuals. Moreover, NS-2 is not easy to install on the Windows platform, but it runs perfectly on Linux. It implements network protocols, such as TCP and UDP. Also, it implements traffic sources, such as FTP and CBR. Finally, it implements queuing mechanisms, such as Drop Tail (FIFO) and RED. In order to run NS-2 simulation, an OTcl Script must be written to create topology using network objects and functions in the library Fig. 1. Also, it can be adapted to set the start and end times for traffic sources. In addition, there is a very important object in NS-2 called event schedule that keeps a track for each single packet during the simulation time including the unique packet ID, which node transmits the Packet and which node receives the packet, as well as the timing of this event (see Fig. 1). Plumbing is another important factor. This makes objects in the network distinguish each other so traffic can reach its destination. C++ language is used by NS-2 to create the network components and compile them to save more processing time. It is also used to separate data path implementation from control data implementation [7].

event	time	from node	to node	pkt type	pkt size	flags	fid	src addr	dst addr	seq num	pkt id

```
r : receive (at to_node)
+ : enqueue (at queue)          src_addr : node.port (3.0)
- : dequeue (at queue)          dst_addr : node.port (0.0)
d : drop    (at queue)
```

```
r 1.3556 3 2 ack 40 ------- 1 3.0 0.0 15 201
+ 1.3556 2 0 ack 40 ------- 1 3.0 0.0 15 201
- 1.3556 2 0 ack 40 ------- 1 3.0 0.0 15 201
r 1.35576 0 2 tcp 1000 ------- 1 0.0 3.0 29 199
+ 1.35576 2 3 tcp 1000 ------- 1 0.0 3.0 29 199
d 1.35576 2 3 tcp 1000 ------- 1 0.0 3.0 29 199
+ 1.356 1 2 cbr 1000 ------- 2 1.0 3.1 157 207
- 1.356 1 2 cbr 1000 ------- 2 1.0 3.1 157 207
```

Fig. 2. Trace format and trace file (Chung and Claypool, 2005).

Trace Analysis and AWK Command: Trace file has the event type of every single packet in the simulation; that is, receive, send or drop. It also includes time of event, source node, destination node, packet size, number of the flow that packet belongs to, address of source and destination, packet sequence number (to make sure that packet will be in its correct order between other packets at receiver side), and finally, packet ID (see Fig. 2). Information included in this file can be analyzed to provide many statistics for the network, such as end-to-end delay, number of packets dropped, delay jitter, throughput and utilization [7]. AWK Command is a programming language that runs on files including the trace file. Since it is very complicated to get information due to the huge amount of data of the trace file, AWK provides statistics and calculates results. Moreover, it is easy to use and flexible, regardless of whether the trace file is for unicast, multicast, wired or wireless networks.

I. PROPOSED MODEL

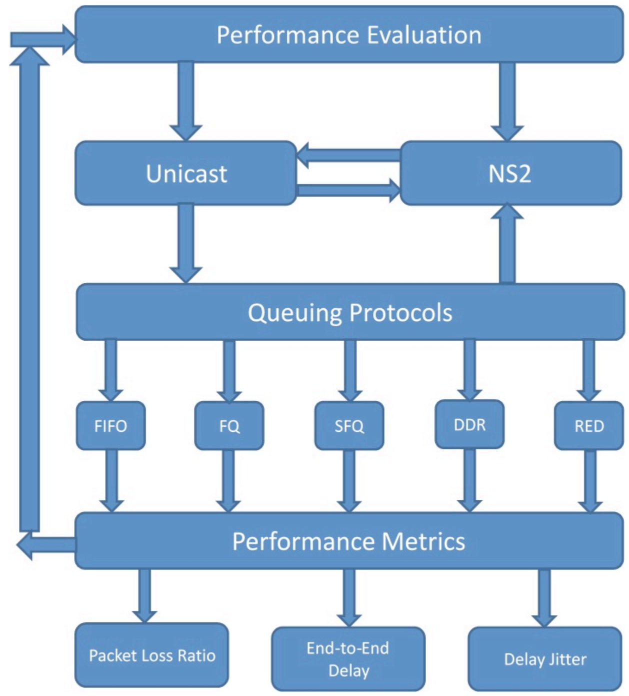

Fig. 3. Diagram represents the proposed model

All formulas established for packet loss ratio, end-to-end delay, delay jitter and drop fairness in the porposed model will be applied on results in the trace file that was created by NS-2.

A. *Packet Loss Ratio*

It is the number and percentage of packets that will be dropped when the size of data exceeds the available bandwidth of the link [5] [20] [23] [37] [47].

Calculation:

$$DP = TP - AP$$
$$PLR = (DP / TP) \, X \, 100$$

Where:

DP: Number of dropped packets at bottleneck router.
TP: Total number of packets sent by a node.
AP: Number of arrived packets at receiver.
PLR: Packet loss ratio at bottleneck router.

B. *End-to-End Delay*

It is the average time duration needed by a flow of packets to be transmitted from sender to receiver, and it indicates the network's speed and reliability [2] [14] [42] [44] [45].

Calculation:

$$EED = TR - TT$$
$$AEED = (EED1 + EED2 + EED3 + ... + EEDi) / TP$$

Where:

EED: End-to-End Delay for a packet.

TR: Received time of a packet at the receiver.
TT: Sent time of a packet at the sender.
TP: Total number of packets sent by the sender.
AEED: Average end-to-end delay.

C. Delay Jitter

It is the difference between maximum end-to-end delay for a packet and minimum end-to-end delay for a packet in the same flow. And it provides the reliability of the network [13] [15] [34] [41].

Calculation:

$$DJ = MXEED - MIEED$$

Where:
DJ: Delay Jitter.
MXEED: maximum End-to-End Delay.
MIEED: minimum End-to-End Delay.

D. Drop Fairness

This is to compare the fairness between different queuing protocols in the case of which packets are going to be dropped and which packets will be sent to their destinations, as well as which flow has the first priority to not be dropped. Normally, fairness value is between 0 and 1; 0 means no fairness at all and 1 means complete fairness [19] [24] [26] [29] [31].

Calculation:

$$f(F1,F2,F3,..........,Fn) = (\ )\verb|^|2 / n \quad (0)$$

Where:
f: Fairness.
F: Number of packets dropped for a specific flow.
n: number of flows in the network.
i: flow number.

II. RESULTS AND DISCUSSION

A. Network Topology

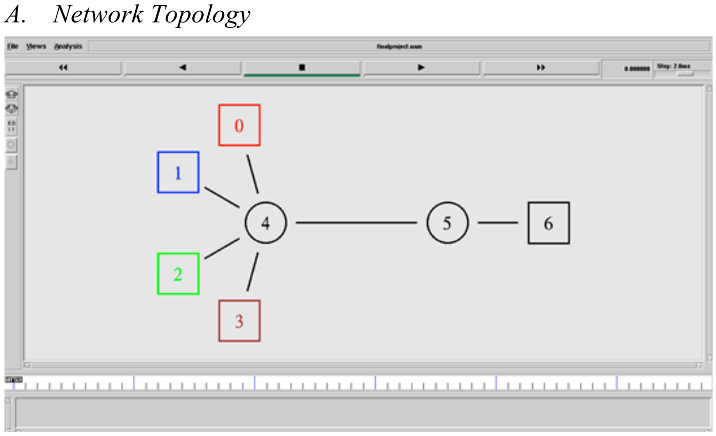

Fig. 4. Topology used in unicast evaluation. Circles represent routers, whereas boxes represent nodes. The black box is the receiver, whereas red, green, yellow and brown boxes are senders (only two senders used in this simulation).

B. Case Description

In this case, traffic was sent from two different CBR sources node 0 and node 1 to node 6, through a bottleneck router node 4 using UDP agent for each source. All links have a bandwidth of 1 Mbps with 10 ms delay except for the link between node 4 and node 5, which has a bandwidth of 1 Mbps with delay of 30 ms. In addition, the packet size is 500 bytes and the interval is 0.005 seconds for both CBR source results in 800 Kbps sent in each link. All results in the coming tables were taken from the trace file using AWK command then applying the proposed formulas in the previous section.

C. Packet Loss Ratio and Dropping Fairness

Flow Number	**Protocol Used**				
	FIFO	***FQ***	***RED***	***DRR***	***SFQ***
one	0	325	359	350	365
two	702	326	370	351	366
Total Packets Lost	702	651	729	701	731
Packet Loss Ratio	0.35	0.325	0.364	0.35	0.365

TABLE 1.

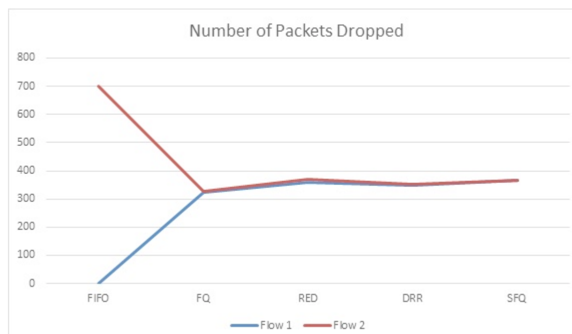

Fig. 5. Graph shows the number of packets dropped from each flow.

D. End-To-End Delay

Flow Number	**Protocol Used**				
	FIFO	***FQ***	***RED***	***DRR***	***SFQ***
one	0.248386	0.419996	0.153469	0.181862	0.137542
two	0.236475	0.420711	0.161158	0.181946	0.137693
Both flows	0.245646	0.420353	0.15728	0.181904	0.137617

TABLE 2.

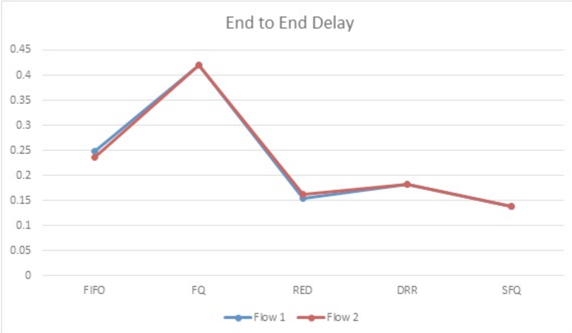

Fig. 6. Graph shows the end-to-end delay of each flow.

E. *Delay Jitter*

Flow Numbe r	Protocol Used				
	FIFO	***FQ***	***RED***	***DRR***	***SFQ***
one	0.195	0.4	0.198	0.2	0.08
two	0.192	0.396	0.195	0.192	0.076
Both flows	0.19431	0.398	0.1965	0.196	0.137617

TABLE 3.

Fig. 7. Graph shows delay jitter of each flow.

F. *Analysis*

FQ was giving the best results because it had fewer lost packets and the dropping fairness between the flows was almost the same. The main reason for this success is that FQ uses a round-robin algorithm that serves packets from each flow in turn. Whereas FIFO and DRR were performing very similarly in the case of packet loss ratio, DRR was providing much better performance in the case of dropping fairness because FIFO is serving the first flow of packets comes to the buffer result in starving out other flows. However, RED and SFQ are providing better results than FIFO and DRR in the case of dropping fairness, but in the case of packet loss ratio, they are the worst.

In the case of end-to-end delay, SFQ and RED are the best with less average delay for the first flow when RED was taking place, whereas SFQ provided the same delay for both flows. However, FQ was the worst because it uses a round-robin algorithm, which takes time to switch between flows. Finally, DRR and FIFO performed better than FQ but worse than SFQ and RED, with a slightly better performance for DRR that had a similar delay for each flow.

FQ had high delay jitter; on the other hand, SFQ had small delay jitter because a fixed number of flows were used. In contrast, FIFO performed better than FQ, whereas DRR and RED had very similar average delay jitter. Also, in general, the second flow had less delay than the first flow for all queuing protocols.

III. CONCLUSION AND FUTURE WORK

The proposed model provides how results are calculated and collected, including the network simulator, NS-2, and are then used to achieve the purpose of the study. Additionally, it provides how to choose the suitable protocol for the application requirements to determine whether the network is able to meet the application requirements. For example, mail servers can use FIFO since it does not care much about delay, whereas a real-time application needs a guarantee of delay and data arrival. Therefore, choosing the right protocol with a balance between delays and dropping fairness is very important.

Unicast network performance has been evaluated under different queuing protocols using multimedia data that has been represented by the UDP agent over ftp. In order to evaluate the performance of queuing protocols under metrics, such as packet loss ratio, dropping fairness, end-to-end delay and delay jitter, the proposed model was used. The scenario consists of making two nodes send traffic to a destination and then collecting results under different metrics. FQ was providing the best results in the case of packet loss ratio and dropping fairness, whereas SFQ and DRR were better in the case of end-to-end delay. Finally, SFQ had the best performance in the case of delay jitter, however FIFO is the most used protocol nowadays due to its simplicity.

The current research can be extended in different ways, such as changing the bandwidth, topology, delay time and buffer space. Moreover, using TCP over ftp instead of using UDP over CBR is another way to extend the work. Further, adding more factors and metrics, such as overhead state, could be an extension for this work.

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