

AN EFFICIENT FAIR QUEUING MODEL FOR DATA COMMUNICATION NETWORKS

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ABSTRACT

The advent of data communication networks has been one of the greatest discoveries that can ever be witnessed by mankind. Despite the benefits derived from application of communication networks, there are several factors confronting the use of communication networks. One of them is Traffic congestion, which reduces throughput and causes delay of data items.

The aim of the paper is to develop an efficient fair queuing model that is capable of reducing congestion by allocating resources on the network between contending users. The proposed model gives higher priority to real time in order to allow them to have dependable performance. Stimulation of this proposed model is carried out using queuing performance parameters like complexity, throughput and delay time of the information.

Our simulations and analysis demonstrate the effectiveness of our proposed model. It is adequately compared with previous fair queuing schemes.

Keywords: Data, Communication, Networks, Queuing Model, Traffic, Congestion, Throughput, Complexity.

INTRODUCTION

Communication plays a central role in entire world where retrieval and processing of information are important. From any where in the world, one can access the entire wealth of information such as monitor the latest swing on

the stock exchange, read and listen to news, search for academic information and so on.

Despite numerous benefits associated with the advent of data communication network, some complexities are becoming increasingly and rapidly associated with it. This is observed with the

introduction of new functions, services and increase in connectivity. As more people are getting connected to a given network to make use of limited resources on it, there is much increase probability of data traffic congestion on the network.

Fair queuing is a technique that reduces congestion by allowing each flow to pass through network devices to have a fair share of network resources [1]. However, such mechanism usually needs to maintain state, manage buffer or perform packet scheduling on a per flow basis, and this complexity may prevent them from being cost effectively implemented, reduce delay for real-time traffic and widely deploy.

In this paper, a new efficient fair queuing model that significantly reduces this implementation yet still achieves approximately fair bandwidth allocation with minimal delay for real time traffic is implemented.

RELATED WORKS

Communication Networks Classification

The type of data communication facility to be used by any organization depends on the nature of the application, the number of computers involved and their physical separation facilities. Two basic network types are Local Area Networks (LANs) and wide-area (or long-haul) networks (WANs) [2].

Local Area Networks (LANs) connect

computers and peripheral devices in a limited physical area, such as a business office, laboratory, or college campus, by means of permanent links (wires, cables, fiber optics) that transmit data rapidly. A typical LAN consists of two or more personal computers, printers, and high-capacity disk-storage devices called file servers, which enable each computer on the network to access a common set of files. LAN operating system software, which interprets input and instructs networked devices, allows users to communicate with each other; share the printers and storage equipment; and simultaneously access centrally located processors, data, or programs (instruction sets).

Wide-Area Networks (WANs) connect computers and smaller networks to larger networks over greater geographic areas, including different continents. They may link the computers by means of cables, optical fibers, or satellites, but their users commonly access the networks via a modem (a device that allows computers to communicate over telephone lines). The largest wide-area network is the Internet, a collection of networks and gateways linking millions of computer users on every continent [2, 3].

Mode of Data Transfer

Mode of data transfer on the network specifies method by which information or data can be transferred over transmission media on the network [3]. Mode of data transfer can be described in three following ways:

Simplex transmission: This is one-way transmission between a transmitter and a corresponding receiver. The communication is unidirectional as on a one-way road.

Half-duplex transmission: Two-way transmission is

possible, but it cannot take place simultaneously; data must first be transmitted in one direction before transmission in the reverse direction is possible.

Full-duplex transmission: This is simultaneous transmission in both directions. Both stations can simultaneously transmit and receive data from each other [4, 5]. Figure 1 below illustrates three mode of data transfer.

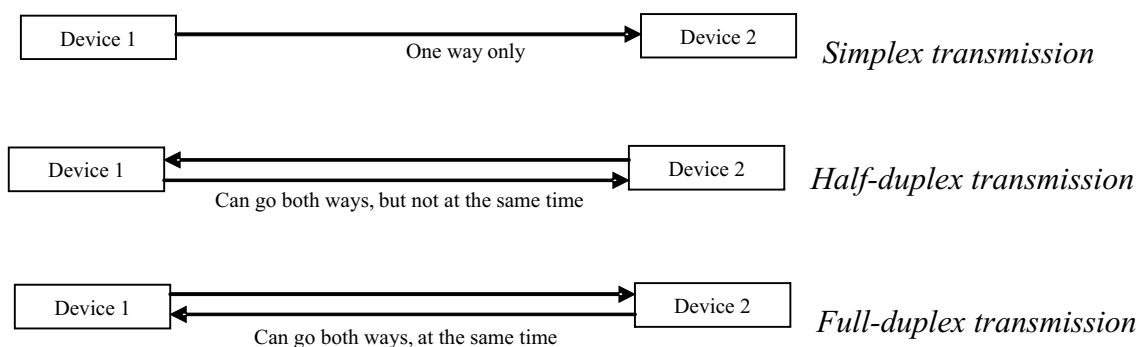


Figure 1. Simplex transmission, Half-duplex transmission and Full-duplex transmission

Congestion Control in Data Communication Network

Traffic congestion is said to occur on the network when there are too much demand for a particular resources beyond what the network can handle. This invariable gives rise to unpleasant delay and through put of data network. Unlike tradition voice communication, where an active call requires constants bit rates from the networks [6]. A typical data session may require very low data rate during periods of inactivity and much higher rate at other times. Consequences, there

may be times when incoming traffic to a network exceeds its capacity and result into low level of data throughput.

Fair Queuing

Fair Queuing is a technique that control traffic congestion on the network by allowing each flow passing through a network device to have a fair share of network resources.[7].

Roles of Fair Queuing in Congestion Control.

Data networks such as the internet,

because of their reliance on statistical multiplexing, must provide some mechanism to control congestion. The current internet which has mostly first-in first-out (FIFO) queuing and drop-tail mechanisms in its routers relies on end-to-end congestion control in which hosts curtail their transmission rates when they detect that the network is congested.

Fair Queuing Performance Metrics

There are varieties of quality services of metrics or measures of a fair queuing performance. The relevance of a particular metric depends upon the type of network (connection oriented or connectionless) [6, 7].

I. Packet Delay: The total time, the network takes to deliver the data packet from the time the first bit of the packet enters the network to the time the first bit of the packet is delivered to the destination.

II. Throughput: The measure of the amount of data delivered per unit time. It is often measured in packet per second.

III. Delay Jitter: An important metric in some virtual circuit packet networks. It is a measure of the degree of variability in the time between successive packets delivered in a virtual circuit.

IV. Blocking probability: It is a fundamental metric of most connection oriented networks, that is, circuit switch and virtual-circuit switch network. In

these networks, an application requests bandwidth in the form of a connection before transmitting data into the network. If insufficient resources are available for the connection (as determined by the type of network, a description of the desired resources and network policy) the request is blocked.

V. Fairness in network use: This is the notion of treating all sessions in the network equally.

VI. Algorithm complexity: This is the measure of efficiency of queuing scheme in respect to time and space utilization.

Definition of Terms

I. REAL-TIME FLOW: Flows that are delay sensitive, it could comprise of audio and video.

II. BEST-EFFORT FLOW: Flows of data that are not sensitive to delay, it is made up of textual data.

III. THROUGHPUT: Measure of amount of packet delivered per time.

IV. PACKET: In data communication, the basic logical unit of information transferred. A packet consists of a certain number of data bytes wrapped or encapsulated in header and trailers that contain information about where the packet came from where it's going and so on.

Fair Queuing Models

There are various queuing models applied to improve the performance of networks and other systems where users statically share

resources. Some of these models exactly predict the performance under some assumed traffic conditions, while others are only approximate [7]. Some are statistical, some are deterministic and some have simple analytical solution, while other requires numerical computation.

First Come First Serve (FCFS)

Most routers use First-come first serve [8] on output links. Here, the order of packet arrival completely determines the allocation of packets to output buffers. The presumption is that congestion control is implemented by the source in such a way that connection are supposed to reduced their sending rate when they sense congestion. However, a rough flow can keep increasing its share of the bandwidth and cause other flows to reduce their share.

Nagles' Fair Queuing

Nagle proposed an approximate solution to the first come first serve (FCFS) by identifying flows using source-destination address and separate output queues for each flow. The queues are serviced in round-robin fashion. This prevents a source from arbitrarily increasing its share of the bandwidth [9]. When a source sends packets too quickly, it merely increases the length of its own queue. Despite its merits, there is a flaw in this scheme it ignores packets lengths. The assumption is that the average packet size over the

duration of a flow is the same for all flows in this case each flow gets an equal share of the output rate.

Bit-By-Bit Round Robin (BR)

In BR scheme, each flow sends one bit at a time in round robin fashion, since it is impossible to be calculated .The packet is then inserted into a queue of packets sorted on departure times. Unfortunately, it is expensive to insert into a sorted queue. The best-known algorithm for inserting into a sorted queue find out requires $O(\log(n))$ times; where (n) is the number of flows. While the BR guarantees fairness, the packet processing cost makes it hard to implement cheaply at high speed.

Self-Clocked Fair Queuing (SCFQ)

The scheme is based on virtual time function that makes computation of the packet departure time from their respective queues to be simpler [10]. Virtual time function, serves as the measure for the work progress in the system to be evaluated for every packet. Moreover, it is shown that the SCFQ scheme is nearly optimal in the sense that the maximum permissible difference among the normalized services offered to the back logged sessions is never more than two times the corresponding figure for any packet based queuing system. Since the virtual function evaluated for every packet in the head of the queuing is simply

extracted from the packet in the head of the queue, its generation involves minimal data processing. However, there are still computational cost associated with the sorting technique used in SCFQ because virtual time computation retains $O(\log(n))$ sorting complexity.

Deficit Round Robin (DRR)

DRR is a scheme that provides solution to the unfairness caused by possible different packet by sizes used by different flows [1]. Flows are assigned to queues such that each queue would be served in round robin arrangement. The only different from the traditional round robin is that if a queue was not able to send a packet in the previous round because its packet size was too large, the remainder from the previous quantum is added to the quantum for the next round. One of the elements of DRR is the possibility that two or more flows will collide, which will equally leads to sharing of bandwidth by the colliding flows.

Priority Queuing

When different traffic types (voice and data) share common network resources, such transmission lines, and router and so on, they may be given (World Wide Web) different service requirements. For example, in a single server system, delay sensitive traffic may be served before delay to tolerant traffic. One possible

scenario is to divide traffic into L priority classes with class “I” heaving priority over class “IH” and maintain a separation queue for each priority class. When a server becomes free, it starts serving a packet from the highest priority queue.

METHODOLOGY

Identification of these difficulties and others make it imperative to propose another queuing model that lays emphasis on delay of real-time flows and fair allocation of resource with reduced Implementation complexity.

Since data communication network consists of both real-time and best effort traffic, scheduling of resources is achieved in a way that incoming flow to the router is identified as real-time flow or best-effort traffic. Each real-time and best-effort flow is temporarily stored in separate buffer before allocation process commences. Higher priority by first providing service to them using ordinary packet by packet round robin while best-effort flows are then served using deficit round robin schemes. The major reason behind serving real-time flows first is to dependence performance with respect to throughput and delay time.

Queuing Model Analysis and Design

Data communication network support different types of services that include real-time, best-effort and many others. These networks

support link sharing, which allow resources sharing among application that require different network services. Different services classes interact with each other at the same output link of a switch. The queuing scheme at the switching node plays a critical role in controlling the interaction among different traffic streams. Application of priority to a queuing system is a simple choice to partition traffic classes for example, a real-time traffic over best-effort traffic. Higher priority for real-time flows, always obtain earlier service to enable them have higher throughput and lower delay.

Design Specification

The specification of the proposed queuing model involves identification of flows, which is a stream of packets that transfers the same router from the source to the destination

and requires service at each router in the path. In addition, every packet can be uniquely assigned to a flow using pre-specified field packet header. Also, a flow could be identified by packet with the same source - destination addressed. We assumed a model of router where packet sent by flow from nodes is attached with special bits on their header, which indicate whether they are real-time flow or best-effort flow. Packet arrive to the buffer that queue a packet to an output link for a router, it is assumed there is a queue process at each output link that is active whenever there is packets queued for the output link. It is also important to note that the queuing system at an output link of transmission speed C with each mode on the network capable of flows emanating from them. More detail of the proposed of the model is illustrated below

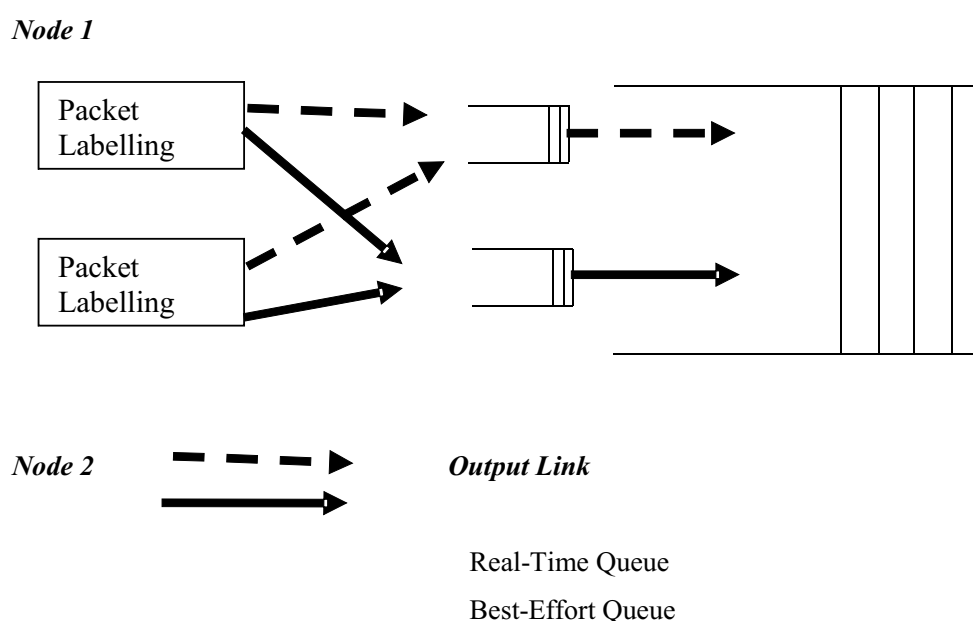


Figure 2. The architecture of the proposed fair queuing model.

ANALYSIS

Identification Stage

Special bits in the packets header identify flows from different nodes on the network. The bits specify whether packet belongs to real-time or best-effort flows. They also give information on the source and destination address of packet. Each node transmits different numbers of real-time flows and best-effort flows in specific time interval . The total amount of flows for the output buffer at time (t) is calculated by addition of real-time and best-effort flows at that time.

λ_R = Number of real-time flows to be enqueued at time (t)

λ_B = Number of best-effort flows to be enqueued at time (t)

λ = Total flows in the output buffer at time (t).

Enqueuing Stage

It is assumed that flows from nodes are combination of real-time and best flows and using separate output queues for each flow, the queues are served according to the status (i.e. based on their header tag). There are two sets of queues. Packets coming from real time flows are stored in one set of queues while packets from best-effort flows are stored in different queues. Therefore, real-time flows (λ_R) are stored in set of queue QR, and best-effort flows (λ_B) in another set of queue QB.

Link Sharing Stage

Let us assume that flow arrival rate for flow 'i' is $r_i(t)$ for time interval 't' and ideal share of the output link between real-time and best-effort flows are $S_R(t)$ and $S_B(t)$. The total arrival rate of all the flows during time 't' is $A(t)$ such that

Similarly, with speed C of output link, there is an influence on the number of packet transmitted through output link by total arrival rate $A(t)$

$$C = [A(t), S_R(t), S_B(t)].$$

For situation where , All the flows will be forwarded or there will be no traffic congestion.

Service Stage of the Model

The two separate sets of queues that consist of real-time and best-effort flows are served based on priority. Flows from real-time queues are given earlier service before those from best-effort queues.

In addition, real-time flows are served using ordinary packet-by packet round robin while those in best-effort are served using deficit round robin queuing scheme.

Queuing Design

The design of agent involves development of unique software process that operates asynchronously with other agents towards achieving fair allocation of resources.

IDENTIFICATION PROCESS

Identification Process involves class of process that identifies flows that arrive to the queue server based on the information on the packet header and decide whether it is real-time flows or best-effort flows. This agent has ability to communicate with other agents and provide useful information about the flow type.

Enqueuing Processes

These processes (real-time and best - effort) are responsible for inserting packets from different

flow into different queue based on their arrival time.

separate queuing schemes applied on both real-time queue and best-effort queue.

Dequeuing Processes

The flows in real-time queue are first served before that of best effort queue. The process is capable of selecting packets to be transferred to the output line from real-time set of queue or best-effort queues. Selection of packets to be transferred from real-time queue and best-effort is based on

Design Structure

The research work is a fair queuing model, in which a number of processes communicate together to achieve the required functionality. Figure 3 illustrates the design structure of the model

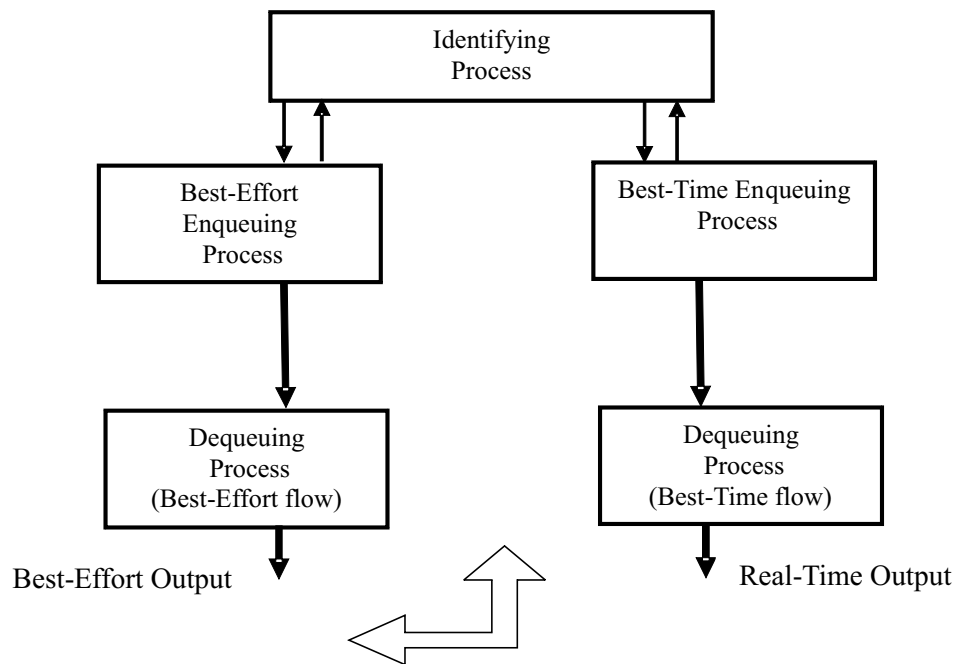


Figure 3: Design Structure

MODEL SIMULATION AND PERFORMANCE ANALYSIS

This paper analyses the behavior of the proposed model under different conditions using graphical simulation and compare the result with performance of the deficit round robin queuing scheme. The various performance metrics used in the simulation are:

- Packets size on bandwidth allocation
- Delay time suffered by real-time packet
- Packet arrival rate on the allocation bandwidth

- Fairness in bandwidth allocation between best-effort and real-time flows

The performance analysis of the proposed model will provide answer to the following questions:

- Is there any appreciable different in delay time suffered by real-time flow in both deficit Round robin scheme?
- Is there any level of fairness in bandwidth allocation to both real-time and best-effort flows?

- Is there any relationship between flow throughput and packet size?
- Is there any level or relationship between packet arrival rate and flow through put

Effect of Packet Arrival Distribution of the Proposed Model

Generally, data network that has no adequate congestion control mechanism is always susceptible to sources that send packets to the central switch in uncontrollable rate and seize a large fraction of the bandwidth [1]. In this proposed model, any source-sending packet at higher rate is order to deprive others of their full portion of the bandwidth will find it impossible to achieve. Table1 shows that flows sending packet at higher rate will achieve nothing but to increase its

own queue before getting to their destination.

To illustrate this graphically, figure 4 shows the reaction of the proposed queuing model to such flow source. Consider a situation where there 29 incoming flows to central switch of a network. The service rate of the switch is 100byte/second. If each of the flows has 28 byte/second packet arrival rate the maximum of 0.821kbyte of packet remained in the queue after time (t). The amount of packet remained in the queue increase from 0.821 to 0.86kbyte as the flow increase in their sending rate. This shows that flows sending at the higher rate can not capture the bandwidth but can only increase the time they spend in queue.

Flows are discouraged from sending at higher rate in order to reduce their delay time.

Table 1: Simulation Result of effect of Arrival rate on queue system.

Arrival Rate (in Kbytes/s)	Fraction of packet in the Queue (in Kbytes/s)
28	0.8120
30	0.8246
32	0.8338
34	0.8452
36	0.8538
38	0.8615

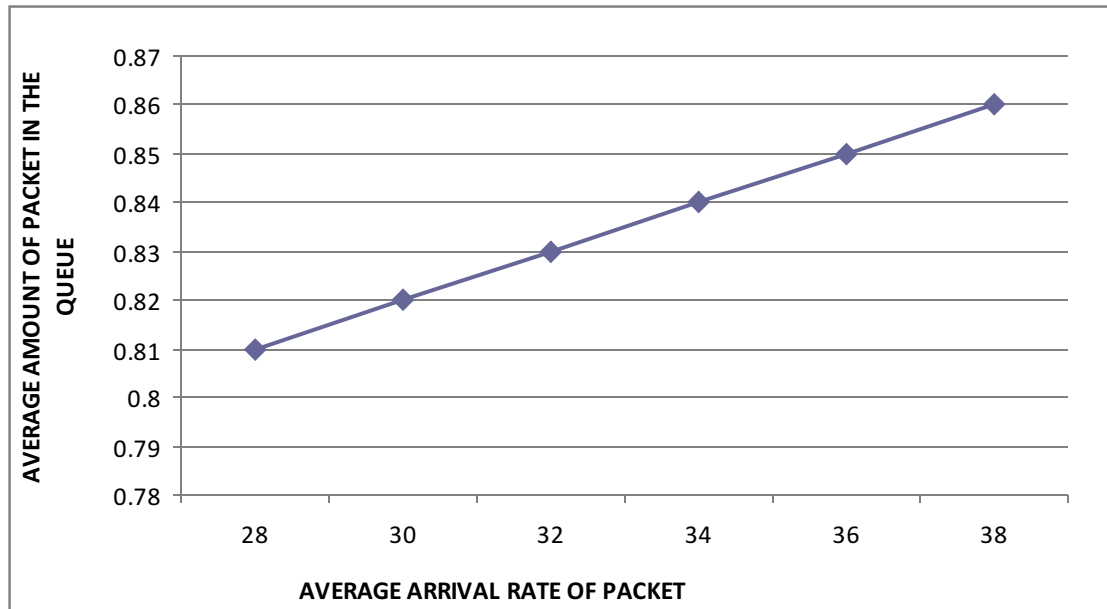


Figure 4 : Graph shows Effect of Packet Arrival Rate on Delay Time

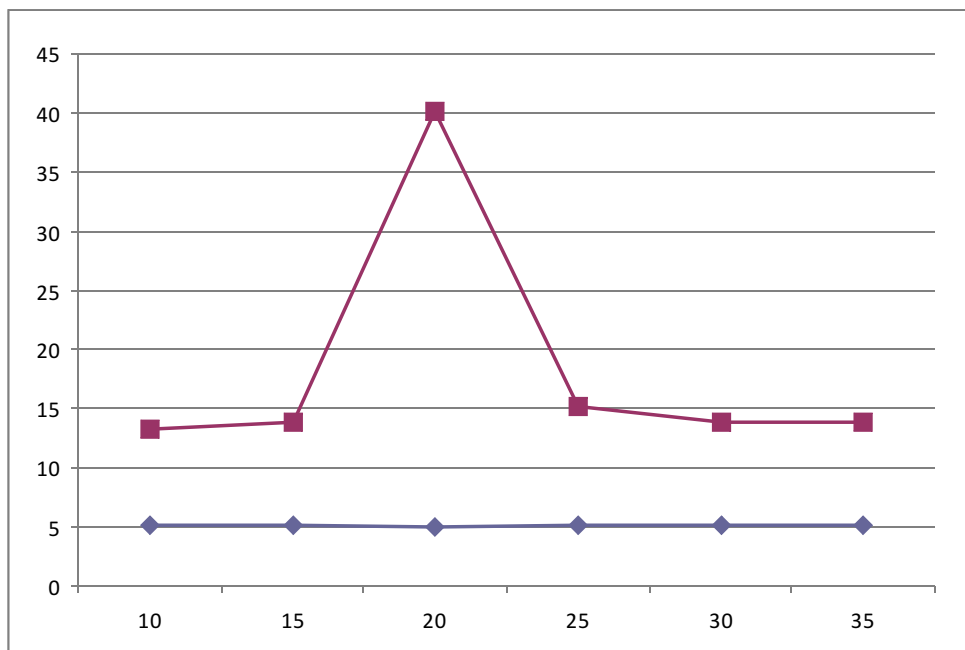
Effect of Arrival Rate on Bandwidth Allocation

The proposed fair queuing has similar features with Deficit Round robin scheme because allocation of bandwidth to flows is not based on the arrival rate of packet. Arrival rates of packets are not used to schedule flows in the proposed

flow. Looking at table 2 it's observed that proposed queuing model is fair in allocation bandwidth to flows on the network. All the flows are given equivalent irrespective of the arrival rate of each flow. The curve for the proposed model remains uniform through the course of simulation.

Flow	Arrival Rate	Packets size Kbytes	Bandwidth (FCFS) Kbytes	Bandwidth of Proposed Model
10	30	147	13.2929	5.1250
15	31	150	13.8393	5.1724
20	90	145	40.1786	5.0291
25	34	150	15.1786	5.1724
30	30	146	13.8929	5.1652
35	31	149	13.8393	5.1652
40	30	144	13.8929	5.0723
45	30	148	13.8929	5.1391
50	30	150	13.8929	5.1724

Table 2: Arrival Rate on bandwidth Allocation



Bandwidth is (FCFS) Kbytes/s Flow Identification
Bandwidth of Proposed Model
Figure 5: Graphical effect of Arrival rate on bandwidth

Effect of Packet Size Distribution on Bandwidth

The packet size of the flows in the system does not have any significant effect on the throughput. Some flows sources that are capable of generating bigger size packet do not enjoy any special treatment in respect to bandwidth allocation in the proposed model. Table 3 shows fairness ability of the proposed model with respect of packet size distribution. Consider two separate flows 10 and 25 of packet size of 145 and 150 Kbytes respectively. Both flows have nearly equivalent output through put. In best effort session of the proposed model, where deficit round robin scheme is employed flow with packet that can not be served in service round will be compensated in the next round of service. This

Flow	Packets	Bandwidth (FCFS)	Bandwidth of Proposed Model
10	145	21.0756	7.7700
15	147	21.3663	7.8199
20	148	21.5116	7.8449
25	150	21.8023	7.8947
30	147	21.3663	7.8199
35	148	21.5116	7.8449
40	147	21.3663	7.8199

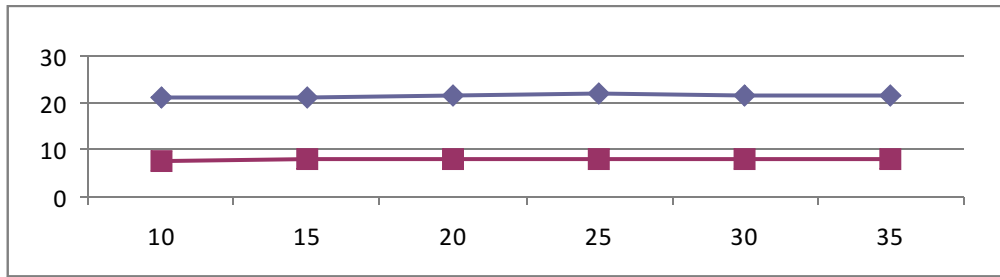
Table 3: Packet Size Distribution on Bandwidth

shows that flows sending bigger packet will have to wait till there is sufficient service to service to satisfy them. Smaller packets only wait for short time to be served while large packets wait for considerable amount of time to be served. At the long run, the through of the bigger packet and the smaller one is approximately equivalent. Therefore, the proposed model share similar property with deficit round robin in through put of the network. In addition, from table 3, flow 25, which has considerable large packet size, has larger fraction of the output bandwidth allocation to it in FCFS queuing model. Figure 6 shows that there is no fairness in bandwidth allocation in FCFS model unlike proposed model where allocation is fair irrespective of the packet size.

Delay Effect of Real Time Flows

Table 4 shows that there is considerable different in delay suffered by real-time packet in both deficit round robin and proposed model. The delay suffered by the real-time packet in deficit round robin model is much larger than the one experience in proposed queuing model. Figure 7 is graphical representation of the simulation

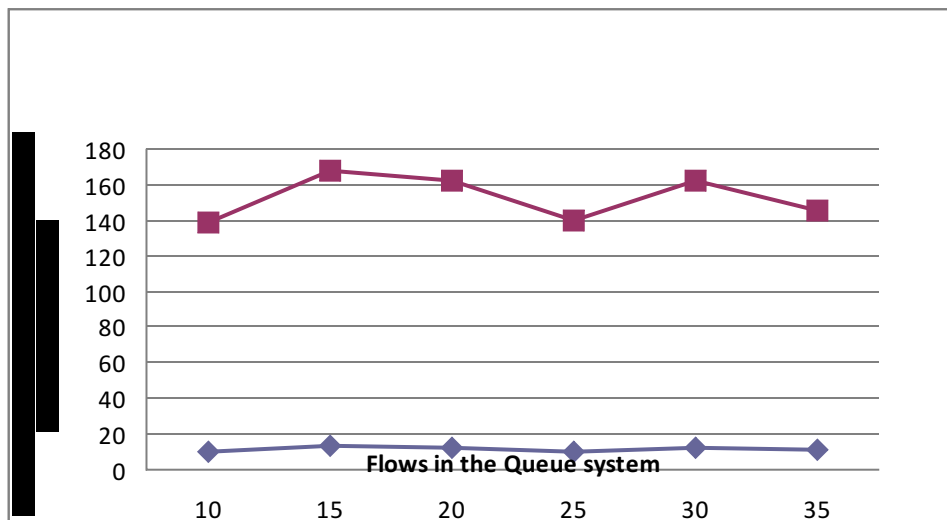
Delay Effect of Proposed Queuing Model



FLOW IDENTIFICATION
 Bandwidth (FCFS) Kbytes/s
 Bandwidth of proposed model

Figure 6: Graphical Effect of Packet Size on Bandwidth

Flow	Arrival rate Kbytes	Proposed model	Deficit round robin
10	30	10.000	139.000
15	35	13.333	167.233
20	34	12.667	161.627
25	30	10.000	139.200
30	34	12.667	161.627
35	31	10.667	144.809



-Delay ration Proposed Model
 -Delay Ration deficit Round Robin

Figure 7: Graph of Delay Time of Real-Time Packet in D-Robin/Proposed Model

DISCUSSION

This research work confirms that communication networks today play a central role in our lives by enabling communication for the exchange of information between various computers. There is increased probability of data traffic congestion on the network with the introduction of new functions services, increase in connectivity and as more people are getting connected to given network to make use of limited resources on it. Fair queuing is a technique that reduces congestion by allowing each flow passing through a network resource.

Previous fair queuing models have many desirable properties for congestion control on data communication network. However, such mechanism usually need to maintain state, manage buffer or perform packet scheduling on a per flow basis and this complexity may prevent them from being cost effectively implemented, reduced delay for real-time traffic and widely deployed.

This paper provides means of achieving approximately fair bandwidth allocation with substantial simplicity and ease of implementation in high-speed networks. Also to keep the traffic level in the network low enough to prevent buffer forever flowing and maintain relatively low end-to-end delay.

In addition to the obvious objectives of limiting delay and buffer overflow. The proposed fair

queuing will treat a session fairly.

Final simulation of the proposed queuing model confirmed that fair queuing for data communication network could be achieved with simplicity and minimal delay for real-time packet.

R E C O M M E N D A T I O N

The proposed model can be properly implemented in packet switched networks for internet or any form of Ethernet network. The proposed model could be applied in multimedia networks where higher premium is given to real-time flows e.g. voice and video.

By the goal of this study, we have tried to develop a new fair queuing model that provides near perfect isolation at very low implementation. The final simulation and performance analysis of model confirmed that is possible to allocate fairly resources on the network between real-time flows and non-real-time flows with low implementation and minimal delay time for real-time flows. The major motivation behind giving higher priority to real-time flows is to allow them to have predictable and dependable performance with respect to delay and bandwidth.

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