Performance Comparison of Voice Packet Sizes in the FIFO Adversarial Queuing and FIFO M/M/1 Model

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Abstract – First-in-First-out (FIFO) is the most widely used scheduling protocol in packet switching network. In fact, it is one of the simplest queuing policies used to provide best effort services in packet-switched network. However, the performance of FIFO is really crucial when it related to stability i.e. question of whether there is a bound on the total size of packets in the network at all times. In this study, our primary objective is to find the optimum packet size of voice packet when using FIFO adversarial network and FIFO M/M/1 network. Our new approach is based on adversarial generation of packets so that positive results are more robust in that they do not depend on particular probabilistic assumptions about the input sequences.

In this paper, we proposed the FIFO scheduling technique that uses adversarial queuing model to find the best packet size of voice packet in FIFO network. Although the simulation results show that the average packet loss is increase when the arrival packet is increased, the average packet delay is improved as compared to FIFO M/M/1 technique, studied by [2]. The proposed algorithm can be applied in packet-switched network, with adaptive routing, in order to transfer voice packet over the Internet.

Keywords: *voice; FIFO queuing model; delay, packet loss; adversarial;M/M/1, optimum packet size*

1. Introduction

First-in-First-out (FIFO) queue is a basic store and forward technique. Packets are queued on a first come first served basis. The packet waiting the longest is transmitted first. When the queues become full, congestion is said to occur and the incoming packets are dropped. FIFO (or FCFS) queuing technique relies on end systems to control congestion using congestion control mechanisms. FIFO queuing works well on uncongested high-capacity links having minimal delay or when differentiation of services for packets traveling through the device is not needed. In this study, we used the two: FIFO adversarial and FIFO M/M/1 networks.

FIFO is one of the simplest queuing policies that are used to provide best effort services in packet-switched networks [4]. One crucial aspect of FIFO's performance is stability, i.e. the question of whether there is a bound on the total size of packets in the network at all times. The stability problem has been investigated under various models of packet routing. One of the models is adversarial queuing model developed by Borodin [1]. This model was developed as a robust model of queuing theory in network traffic, and replaces stochastic by worst-case inputs.

Packet voice technology uses the existing data networks for offering voice. It has a broad appeal in that it is currently unregulated and calls can be placed free of charge to any part of the globe. Since packet networks offer the capability for multiplexing, voice packets travel over the Internet just as data packets do. Thus the operating and maintenance costs may be reduced. The integration of voice traffic with Internet traffic opens up many opportunities. Conventional circuit switched networks dedicated to voice traffic offer very good voice quality because the voice traffic has its own dedicated bandwidth. With circuit switching, data bits go directly to the receiver in an orderly fashion, one after another on a single path. With packet switching routers determine a path for each packet on the fly, directing them over any path available to get to the destination [2]. In IP telephony analog voice signals generated for transmission are first converted into a stream of bits. The digitized voice is then packetized and sent over the network. The process of packetization involves the collection of compressed voice frames into an IP packet. At the receiving end the process is reversed. The voice frame is decompressed. In the case of packet switched networks delay becomes an important issue. The perceived quality of a voice call is delay sensitive. Variation in the delay, jitter, is another problem for interactive voice applications [4]. The time sensitive voice packets and the regular data packets share the same single network. The quality of voice delivered does not always remain the same. The delivered voice quality is an

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important factor in determining the success of VoIP [5]. This research proposed optimum packet size of voice packet using FIFO scheduling network, using leaky bucket algorithm with adversarial generation of voice packet.

The rest of the paper is organized as follows: Section 2 provides a brief description of the related works. Section 3 provides the details of the simulation environment. Section 4 contains a presentation and analysis of the results. Finally, a section 5 lists our conclusions and suggestions for future work.

2. Literature Review

FIFO is by far the most widely used scheduling protocol. In traditional queuing theory, the source which generates network traffic is typically assumed to be stochastic. However, the growing complexity of network traffic makes it increasingly unrealistic to model traffic as, say, a Poisson stream [4]. Adversarial Queuing Theory is a robust and elegant framework developed by Borodin *et al.* to address this problem. In this model, packets are injected into the network by an adversary rather than by a stochastic process. The route of each packet is given along with the packet itself. Each edge in the network can forward at most one packet in one time step. If there are multiple packets waiting to cross the same edge, then we need a contention resolution protocol to decide which packet goes across and which packets wait in the queue.

A packet forwarding protocol is said to be stable against a given adversary and for a given network if the maximum queue size, as well as the maximum delay experienced by a packet, remain bounded. For packet rather than voice packet, it was proved by previous research that FIFO can be unstable at certain load. The original proof of Andrews *et al.* [8] showed instability of FIFO at rate 0.85. Diaz *et al.* [9] improved the threshold of instability for FIFO to 0.83. This was further improved by Koukopoulos *et al.* [10] to 0.749 and recently, Lotker *et al.* [11] further improved it to 0.5.

Phalgun [2] proof that with the FIFO queuing strategy using M/M/1 queuing system model, when the traffic intensity increases the amount of queuing increases. The delay of voice packets being processed increases as the queue size increase. This is due to the fact that as more and more voice packets wait in the queue, the delay for the voice packets increases in time. When using small packet sizes the packetization delay is very small but the queuing delay leads to high end-to-end delay. The queue becomes full and voice packets are dropped.

Motivated by the study of a new approach that was introduced by Borodin [1] to improve packet arrival processes in applications such as heterogeneous ATM networks, we conduct a study to find the optimum packet size of voice data for FIFO scheduling policy using an approach that based on adversarial generation of packets. In this approach the positive results are more robust in that they do not depend on particular probabilistic assumptions about the input sequences.

3. Simulation Methodology and Design

This section discusses the technique undertaken to develop the queuing technique by using leaky bucket algorithm with adversarial packet generation at arrival event. The approach or methodology is a fundamental step in any development process. The performance evaluation technique used to study the method is simulation. Simulation studies are performed, not on the real-world system, but on a model of the system (usually computer-based) created for the purpose of studying certain system dynamics and characteristics.

The C language is used to simulate the implementation of voice packet queuing using adversarial queuing model.

3.1 Simulation Model

The simulation model is based on FIFO scheduling technique, which is used to simulate the performance of voice packet over the network. The simulated model involves one (1) source and one (1) sink, connected together by five (5) fix paths. At the arrival event, the source will inject packets into the network. The packets can travel along five (5) edges connected to the sink.

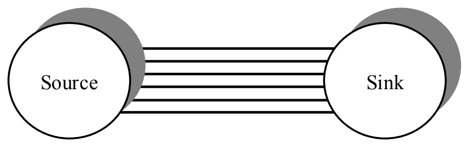

Figure 3.1: Simulation model with 1 source, 1 sink and 5 paths

On this simulation model, some procedures will be executed in order to evaluate the overall performance of the proposed technique:

- Different size of packets will be tested in order to find the optimum size of the packet in FIFO network.
- Maximum delay 150 ms in order to ensure the quality of voice packet is not degraded.
- Maximum packet loss is 1%.

3.2 FIFO using Leaky-bucket Algorithm with Adversarial Queuing Packet Generation

FIFO scheduling technique with adversarial packet generation is used to find out the impact of packet size of voice packet upon FIFO. In adversarial queuing model, time proceeds in discrete steps. A packet is an atomic entity that resides at a node at the end of any step. A packet must travel along a path in the network from its source to its sink, both of which are nodes in the network. When the packet reaches its destination, we say that it is absorbed. During each step, a packet may be sent from its current node along one of the outgoing edges from that node. At most one packet may travel along any edge of the network in a step. Any packets that wish to travel along an edge *e* at a particular time step but are not sent wait in a queue for edge *e*. The delay of a packet is the number of steps which the packet spends waiting in queues [1].

At each step, an adversary generates a set of requests. In this paper, a request is a path specifying the route followed by a packet. We say that the adversary injects a set of packets when it generates a set of requested paths. We restrict ourselves to the case in which the path traversed by each packet is fixed at the time of injection (using leaky-bucket algorithm), so as to be able to focus on the queuing rather than routing aspects of the problem.

The request (number of path) is generated depending on the number of packet that the source created during arrival event. We focus on the non-adaptive model in which the adversary is allowed to split packets and route them using multiple paths; the algorithm, in contrast, is required to route each packet along a single path.

3.3 Performance Metrics

Two performance metrics have been chosen to evaluate the model. These metrics are commonly used in the performance study of VoIP [2,5,6,7]. The simulation was designed to capture these performance metrics.

a) Average delay
Delay is defined as number of steps packet spend waiting in queue. Voice traffic is extremely sensitive to delay [5]. Delay causes two problems, echo and talker overlap. To avoid these problems, delay is allow up to a 150 ms only. If the delay exceeds 151 ms, the packet will be discarded from the queue.

b) Average packet loss
Packet loss results when packets sent are not received at the final destination. For good voice quality in a VoIP network, packet loss of compressed speech should be not more than 1% [5].

For the purpose of the simulation, packet delay is allowed up to 150 ms. If the delay exceeds more than 150 ms, the packet will be discarded from the queue and average packet loss will increases. In order to find optimum packet size, average packet loss is observed where optimum packet size will be number of packets before 1% of packet loss.

3.4 Simulation Tool

The programming language used in the simulation is C language. A program is designed to transfer voice packet from source to sink. All packets can travel through five (5) paths to get to the sink. If all the paths are used, packets will have to wait in the queue. Maximum time for all packets to be in the queue is 150 ms. If delay is greater than 150 ms, the packets will be dropped from the queue.

Before transmitting packets to the sink, the source will generate request. A request is a path specifying the route followed by a packet. We say that the adversary injects a set of packets when it generates a set of requested paths [1]. The path traversed by each packet is fixed at the time of injection.

For the purpose of determining optimum packet size, maximum sink is one, maximum source is one, maximum path is five, and unit of time is measured in millisecond (1000). Based on previous research by Boonchai Ngamwongwattana, the link capacity is assumed to be 256 Kbps and size of the packet is 250 bytes. The voice call traffic is assumed to be a constant bit rate in which techniques, such as voice activation detection, are not applied. The rate of packet generation for the required load condition is computed as $N(r, e) / w \leq \rho$ where $N(r, e)$ is the number of paths injected by the adversary during time interval t that traverse edge *e* and *w* is the window size of the network [1]. The packet size is set at a value of 250 bytes, the average size of a packet in the Internet [2].

The results are stored in output file. Graphs are plotted based on arrival rate, average delay and average packet loss. The assumptions made are that the sink is always available and all the packets are voice packet.

3.5 Simulation Processes

The following are the important processes in the simulation and its brief description.

a) Input Generator
These modules generate input for bandwidth size and packet size in order to generate request.

b) Arrival process

This module specifies the path that the packet travels to get to the sink. Number of paths will depend on request generated at main function. The time for all packets arrive will be taken in order to determine maximum waiting time spend in the queue.

c) Service process

This module calculates the service time for each packet arrives in the system.

d) Departure process

This module calculates total delay for each packet and schedule the next departure.

e) Initialization

This module initializes arrival rate, simulation control, clock, buffer, event list, service rate, buffer utilization and performance parameters.

f) Results

The module calculates the performance metrics and writes it out to the output file. The performance metrics recorded are the arrival rate, average delay and average packet loss.

4. Results and Discussions

In this section we analyze and discuss the results obtained from the simulation. The simulation is based on different packet size and results obtained are compared in order to determine optimum packet size of voice packet in FIFO using adversarial packet generation and leaky bucket algorithm.

4.1 Result when load = 0.5, using FIFO adversarial network

In Figure 4.1 where the arrival rate is 100 packets per time step, an average packet loss still scaled up accordingly. The figure shows that the average delay still remains lower. It is observed that FIFO adversarial network is also not stable at all when load = 0.5.

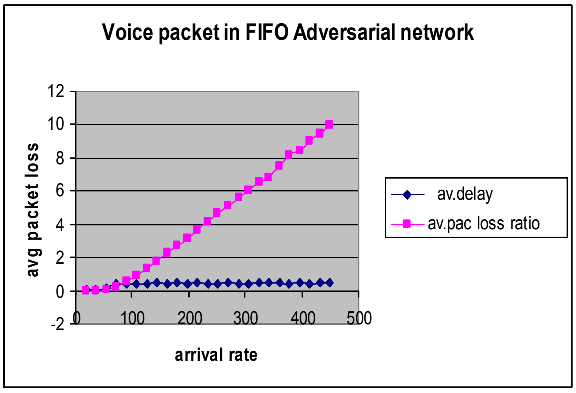

Figure 4.1: The average delay and average packet loss for voice packet in FIFO adversarial network, when packet size is set to 4500 bytes

4.2 Result when load = 0.5, using FIFO M/M/1 network

In Figure 4.2 where the packet size is set to be equal as experiment in Figure 4.1, an average packet loss still scaled up until arrival rate = 3420, where the average packet loss is scaled down. The figure shows that the average delay still remains lower. It is observed that FIFO adversarial network is still not stable when load = 0.5.

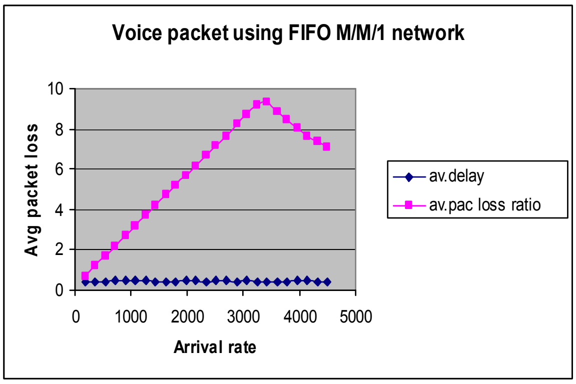

Figure 4.2: The average delay and average packet loss for voice packet in FIFO M/M/1 network, when packet size is set to 4500 bytes

4.3 Result when load = 1, using FIFO adversarial network

In Figure 4.3 where the arrival rate is 200 packets per time step, an average packet loss is scaled up accordingly. The figure shows that average delay remains lower. This is because the characteristic of the voice packet itself, where the packet will be discarded if the delay reach 150 ms. This characteristic caused increment in average packet loss, because the machine is not able to process the packet within the service time allocated for each packet. It is observed that FIFO adversarial network is not stable at all when load = 1 (maximum).

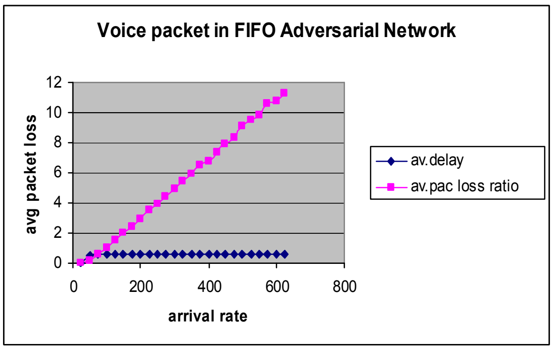

Figure 4.3: The average delay and average packet loss for voice packet in FIFO adversarial network, when load is set to maximum

4.4 Result when load = 1, using FIFO M/M/1 network

In Figure 4.4 where the packet size is set to be equal as experiment in Figure 4.3, an average packet loss is also scaled up to one point (at arrival rate 3750) where the average packet loss is scaled down. The figure shows that the average delay remains lower. This is because the characteristic of the voice packet itself, where the packet will be discarded if the delay reach 150 ms. This characteristic caused increment in average packet loss. It is observed that FIFO M/M/1 network is not stable when load = 1 (maximum).

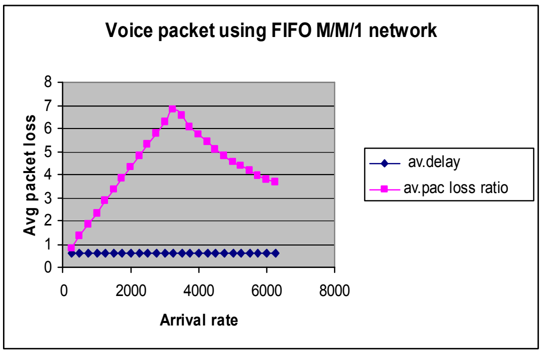

Figure 4.4: The average delay and average packet loss for voice packet in FIFO M/M/1 network, when load is set to maximum.

4.5 Result when packet size is optimum, using FIFO adversarial network and FIFO M/M/1 network

In Figure 4.5 where the arrival rate is 50 packets per time step, an average packet loss is approximately 1. This result fulfills the requirement for the good quality of voice packet. From the result, it is observed that FIFO adversarial network is best performs at arrival rate 176 (packets) or load = 0.88.

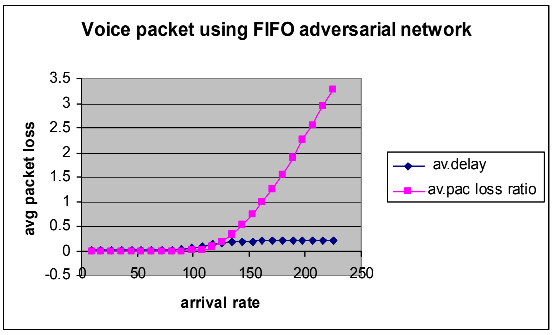

Figure 4.5: The average delay and average packet loss for voice packet in FIFO adversarial network, when packet size is set to 2000 bytes.

On the other hand, Figure 4.6 shows the result of transferring voice packet using FIFO M/M/1 network, to compare the performance with FIFO adversarial network. It can be seen that with same packet size as simulation conducted in section 4.5, the performance is worse. Even the packet will not experience packet losses, but the graph shows that when the intensity increases the amount of queuing increases. The delay of the packet increases as the queue size increase [2]. This performance is not wanted in transferring voice packet because voice packet is delay sensitive.

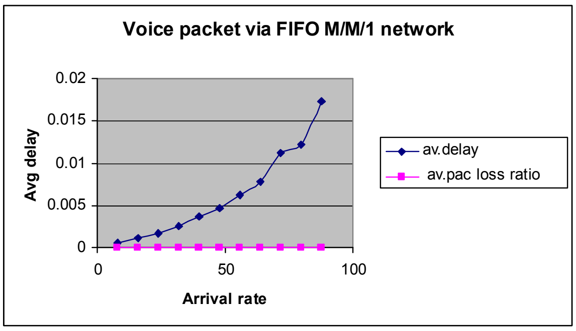

Figure 4.6: The average delay and average packet loss for voice packet in FIFO M/M/1, when packet size is set to 2000 bytes.

5. Summary

The research shows that the Adversarial Queuing Model is a better model (or at least safer) model of arrival processes in applications such as Voice over IP. In the study by Phalgun [2], it was proved that a large delay and packet loss was experienced by voice packet in FIFO M/M/1. The research showed that FIFO M/M/1 only best performs when frame size is compress to 20 bytes [2]. For packet rather than voice packet, it was proved by previous research that FIFO can be unstable at certain load. The original proof of Andrews *et al.* [8] showed instability of FIFO at rate 0.85. Diaz *et al.* [9] improved the threshold of instability for FIFO to 0.83. This was further improved by Koukopoulos *et al.* [10] to 0.749 and recently, Lotker *et al.* [11] further improved it to 0.5.

From the simulation of this research, the results show that the average packet loss is extremely high when the load is high. This result shows that when the load is high, an adversarial packet generation does not improve the performance of FIFO. But Figure 4.3 shows that this technique has better performance compared to FIFO M/M/1 (as showed in Figure 4.5) and can be used as a technique to improve performance of FIFO in transferring voice packet over the Internet.

From the research, we can conclude that optimum packet size of voice packet using FIFO scheduling policy, with adversarial queuing model and leaky-bucket algorithm, is 250 bytes, to be transferred via network with link capacity 256 kbps, but only best perform at load = 0.88.

6. Future Work

FIFO is one of the simplest queuing policies and has been used to provide best effort services in packet-switched networks. Currently, organizations are also pursuing solutions which will enable them to take advantage of excess capacity on broadband networks for voice and data transmission, as well as utilize the Internet and company Intranets as alternatives to costlier mediums. The technique proposed can be applied in packet-switched network, with adaptive routing, in order to transfer voice packet over the Internet. To get better performance, this technique can be applied with dynamic routing where all paths can be fully utilized without depending on the load.

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Bibliography:

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