# BANDWIDTH UTILIZATION AND NETWORK PERFORMANCE

**by**

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## DECLARATION

I Kato Charles do hereby declare that this Project Proposal is original and has not been published and/or submitted for any other degree award to any other University before.

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# 1 INTRODUCTION

This study examines how bandwidth utilization affects network performance. Bandwidth describes the level of traffic and data allowed to travel over a network or Internet connection. It is a gross measurement, taking the total amount of data transferred in a given period of time at a particular rate, without taking into consideration the quality of the signal itself as stated by [1]. Bandwidth may be characterized as network bandwidth, data bandwidth or digital bandwidth. Data flows quickly and smoothly when the amount of traffic on the network is small relative to its capacity. When the amount of traffic nears the capacity of the network, the speed at which data travels begins to drop. Since several users must share the common bandwidth capacity on the Internet, there will be locations in the network where the demand is higher than the capacity. This can cause network congestion and has negative impact, from the user perspective, on the data transmission rate and quality.

Hence bandwidth utilization in this study is conceived as the independent variable while network performance is the dependent variable. Both bandwidth utilization and performance will be measured in form of experiments. In addition to the introduction, this chapter will also deal with the background to the study, the statement of the problem, the general objectives, the specific objectives of the study, the scope of the study, the significance, Justification and operational definition of terms and concepts.

Bandwidth is like a pipe and if the flow of the material inside the pipe is not monitored and managed properly then it will clog up with unwanted traffic. Similar is the case for computer network bandwidth where it can be hijacked by viruses, spam, peer-to-peer file-sharing traffic, etc. Furthermore [2] states that, the useful resource of any organisation will be eaten by unproductive applications and may be difficult to avail useful services by the needy ones. [3]argue that the goal of managing network capacity is to have the right amount of bandwidth in the right place at the right time for the right set of users and applications. Therefore to resolve the problem, it is important to monitor web browsing activities, identify types of applications used including the main consumer(s) of the bandwidth.

Managing bandwidth allocation for todays traffic diversity is a definite challenge. Network traffic and applications do not share the same characteristics or requirements hence different performance expectations for all traffic. Traffic can be managed under two broad classifications i.e. The first classification is bandwidth management mechanisms which are mechanisms that manage the network resources by coordinating and configuring network devices. The main mechanisms here are resource reservation signaling and admission control. The second classification is traffic handling mechanisms which are mechanisms that classify, handle, police, and monitor the traffic across the network. The main mechanisms here are classification, channel access, queuing mechanism, and traffic policing. All these components aim to improve performance of a network but in this research we shall major on queuing mechanisms.

## 1.1 BACKGROUND

Queuing mechanisms are used primarily to manage the allocations of bandwidth among various flows. The five major router-based queuing mechanisms include Priority Queuing (PQ), Custom Queuing (CQ), Weighted Fair Queuing (WFQ), Class-Based Weighted Fair Queuing (CBWFQ) and Low-Latency Queuing (LLQ) according to [4]. Although these mechanisms attempt to prioritize and distribute bandwidth to individual data flows so that low volume applications dont get overtaken by large transfers there is no unique queuing mechanism to handle most applications requirement. Different queuing mechanisms have different advantages thus combining different queuing mechanisms into a new hybrid queuing mechanism with the most possible positive properties of an individual mechanism can improve time sensitive application requirements. Time-sensitive application traffic must travel undisturbed through the network. When traffic goes through a network device it also goes through queuing mechanisms. For time-sensitive application traffic, a combination of PQ and CBWFQ mechanisms are especially appropriate according to [5].

PQ-CBWFQ mechanism permit delay sensitive voice and video traffic to be scheduled first before the packets in the other queues. The key difference between the PQ-CBWFQ and the PQ is that the PQ-CBWFQ's priority queue will not starve the low priority queues. It is controlled by the bandwidth policer, either by the bandwidth or a percentage of the bandwidth [6]. A PQ drains all frames queued in the highest-priority queue before continuing on to service lower-priority traffic classes. PQ assumes that the different types of traffic can be differentiated and treated preferentially. Separate FIFO queues are created for each defined priority level and the arriving traffic is sorted into its proper queue as it arrives. Typically between two and five levels of priority are defined (e.g high, medium, normal and low), although there is no theoretical limit to how many levels can be defined. More queues, however, means more complexity in running the algorithm.

At the service side of the queue, the processing rule is simple: higher priority FIFO queues are always processed to completion before lower priority queues are processed. Traffic randomly enters a router and leaves the router according to its sorted priority. As shown in the figure below.

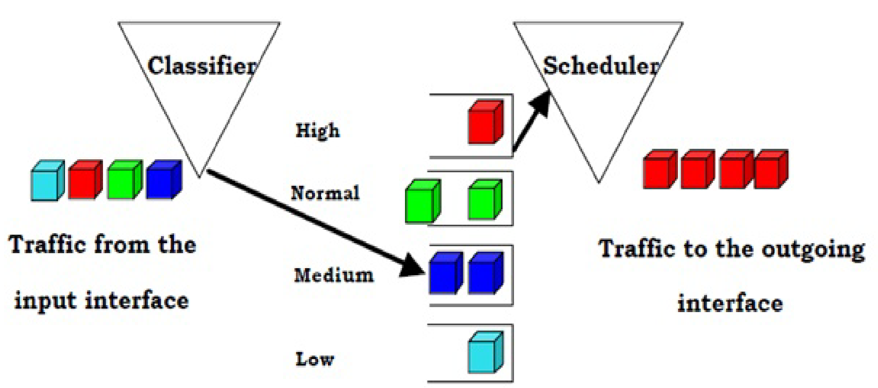


Figure 1: Priority Queuing (PQ)

In a three-queue system, if the two highest priority queues had no packets buffered, then the lowest priority queue would be serviced. The moment a higher priority packet arrived in its FIFO queue, however, servicing of the lower priority packets would be interrupted in favor of the higher priority queue. The existing PQ-CBWFQ algorithm gives strict priority mostly to voice applications only. Unfortunately the increase in demand for higher quality video services due to applications such as video conferencing, interactive video applications, which deliver both video and voice and E-Learning applications has reduced the expected QoS levels which cannot be guaranteed especially if sensitive audio and video packets are processed in the single priority queue due to resource sharing between many applications. To avoid jitter, voice traffic requires a non-variable delay, which is the most important for voice applications. However, video traffic could introduce a variation in the delay, thereby spoiling the steadiness of the delay required for successful voice traffic transmission. There is no distinctive consideration for video applications, which also require more throughputs and less delay.

## 1.2 STATEMENT OF THE PROBLEM

Video application streaming in PQ-CBWFQ algorithms can withstand a little bit delay and buffer tolerant compared to the interactive video applications. In addition to the voice traffic, strict priority status can be given to the video traffic to satisfy the QoS requirements in interactive video applications and to provide the service guarantee in streaming video applications. Unfortunately, the expected QoS level cannot be guaranteed, if sensitive audio and video packets are processed in the single PQ. The reason is that the behavior of the voice traffic is controllable whereas the video traffic is uncontrollable. Video traffic could introduce a variation in the delay, thereby spoiling the steadiness of the delay required for successful voice traffic transmission.

## 1.3 GENERAL OBJECTIVE

The main objective of the study is to improve the network performance through the management of bandwidth utilization by dynamically determining the allocation and division of bandwidth for voice and video traffic in the Priority Queue when using the PQ-CBWFQ queuing discipline.

### 1.3.1 SPECIFIC OBJECTIVES

The study will be guided by the following objectives:-

1. To study the issues related to PQ-CBWFQ queuing discipline in relation to network performance.

2. To develop an algorithm for a dynamic PQ-CBWFQ queuing discipline that meets the requirements of handling multiple time sensitive applications.

3. To implement, test and validate the designed algorithm..

## 1.4 SIGNIFICANCE OF THE STUDY

Research solutions proposed and validated in this research will make the following contributions:

1. The assessments that will be done on bandwidth utilization will determine whether the usage of bandwidth provides for each application hence satisfying the users needs and requirements.
2. The analysis of network performance will allow the network administrator to enhance their network performance more efficiently and much faster.
3. The network will be more manageable and the traffic condition will be much better and thus it will enhance the network capability.

## 1.5 SCOPE

The study is aimed at implementing an algorithm for dynamically determine the allocation and division of bandwidth in the Priority Queue when using the PQ-CBWFQ queuing. It is limited to the PQ-CBWFQ hybrid queuing mechanism due to limitations of time, cost and complexity.

## 1.6 OPERATIONAL DEFINITIONS

The key concepts to be used in the study include:-

Latency - This refers to any delay or waiting that increases real or perceived response time beyond the response time desired.

Utilization - Measure of the usage of a link, port, or network resources.

Availability - Measure of what percentage of the time a network resource is available for use. Clearly, high availability is better because down time is not welcome.

Delay - The amount of time for a packet to traverse, either one-way or round trip, a network, network segment or network device.

Queuing delay - The time spent by a packet in the queues at the input and output ports before it is processed. It is mainly due to congestion in the network.

Packetization/depacketisation delay - The time taken to assemble packet at the senders end and time taken to strip the headers at the receiver end.

Throughput - A measure of how much data can be sent on or through a network resource in a given time period. Also referred to as available bandwidth. Higher throughput is certainly better from users' and administrators' points of view

Bandwidth Management- The control of data flow in a network to provide consistent, reliable, predictable, and manageable flows of traffic.

# 2 REVIEW OF THE RELATED LITERATURE

This chapter provides a brief review of the extensive literature that exists in the area of bandwidth utilization in relation to network performance, as relevant to our study. Network performance management is one of the important tasks required by the network administrator to manage, control and optimize the operations of the network. This is to ensure that the network is operating at an acceptable performance level as stated by [7]. In addition to the introduction, this chapter will also deal with thematic review, the empirical studies, and synthesis of the literature review.

## 2.1 Issues related to PQ-CBWFQ queuing discipline

The increase in the number of system communication users has led to the increase in video streaming services and real-time applications such as Web Server, VoIP, E-mail, FTP and Voice Server. The traffic characteristics of these applications require a certain Quality of Service (QoS) from the network in terms of bandwidth and delay requirements. QoS is considered as potential of the network to produce consistently high quality voice transmissions. In real time applications different parameters (Delay, Jitter and Packet Loss) can be used to measure the QoS of VoIP [8]. Voice applications tolerate little variation in the amount of delay. This delay variation affects delivery of voice packets. Packet loss and jitter degrade the quality of the voice transmission that is delivered to the recipient.

### 2.1.1 Jitter

Jitter is defined as a variation in the delay of received packets. Most of the time jitter is caused due to low bandwidth as stated by [9]. At the sending side, packets are sent in a continuous stream with the packets spaced evenly apart. Due to network congestion, improper queuing, or configuration errors, this steady stream can become lumpy, or the delay between each packet can vary instead of remaining constant. When jitter is high, packets reach destination rapidly. One approach to avoid jitter is to use buffer at end points, but these buffer has to release packets at every 150 ms or even sooner because of transport delay. According to [10].

### 2.1.2 End-to-End Delay

Time elapsed between sending and receiving a packet between two devices is called end-to-end delay. A One-way directinal end-to-end delay should be less than 150 ms for most of the applications. According to [11] delays of 150-400 ms are acceptable if administrators are aware of the time impact on the transmission quality to the user. Performance is considered good if the value of the resulting delay getting smaller.

### 2.1.3 Packet Loss

Packet loss is the term used to describe the packets that do not arrive at the intended destination. It is the measure of the number of packets that were not received compared to the total number of packets transmitted [12] This happens when a device (router, switch, and link) is overloaded and cannot accept any incoming data at a given moment according to [13]. In VoIP networks, packet loss result in short periods of silence and voice distortion. The table below shows the traffic characteristics for video and voice applications according to [14].

| Application | Bandwidth | Delay | Jitter | Loss |
|---|---|---|---|---|
| Voice | Low | Low | Low | Low |
| Interactive video | High | Low | Low | Low |
| Video | High | Medium/High | Low | Low |

Table 1: Traffic characteristics for different applications

In a network, packets are accumulated and queued into memory buffers of routers and switches. A queue is used to store traffic until it can be processed or serialized. Both switch and router interfaces have ingress (inbound) queues and egress (outbound) queues. An ingress queue stores packets until the switch or router CPU can forward the data to the appropriate interface. An egress queue stores packets until the switch or router can serialize the data onto the physical wire. Switch ports and router interfaces contain both hardware and software queues. Avoiding and managing network congestion and shaping network traffic are provided by using different queuing disciplines, which are a basic part of QoS assurance.

Switch (and router) queues are susceptible to congestion. Congestion occurs when the rate of Ingress traffic is greater than can be successfully processed and serialized on an egress interface. By default, if an interfaces queue buffer fills to capacity, new packets will be dropped. [15] discussed that various queuing disciplines can be used to control which packets get transmitted (bandwidth allocation) and which packets get dropped (buffer space). The queuing discipline affects the latency experienced by a packet, by determining how much time a packet waits to be transmitted. During periods of congestion, QoS provides switches and routers with mechanisms to queue and service higher priority traffic before lower priority traffic and also to drop lower priority traffic before higher priority traffic. In order to provide a preferred level of service for high-priority traffic, some form of software queuing must be used.

## 2.2 Empirical studies

Queues are managed in a way to ensure each queue gets the level of services required for its class. Software queuing techniques include First-In First-Out (FIFO) (default), Priority Queuing (PQ), Custom Queuing (CQ) , Weighted Fair Queuing (WFQ), Class-Based Weighted Fair Queuing (CBWFQ) and Low-Latency Queuing (LLQ). Each queuing mechanism has their own feature and different queuing mechanisms have different advantages, combine different queuing mechanisms into new hybrid queuing methods will reduce the end-to-end delay, Ethernet delay, and jitter as stated by [16] Many different hybrid queuing methods are possible but LLQ which is the combination of PQ and CBWFQ is a queuing scheme that adds strict priority queuing to CBWFQ. Strict-priority queuing allows for delay-sensitive data, such as voice, to be sent first, before packets in other queues are de-queued. When voice packets enter the LLQ system a fixed bandwidth is allocated, data packets enter CBWFQ system where they are treated according to CBWFQ assigned weights as stated by [5].

### 2.2.1 CBWFQ

Class-based weighted fair queuing (CBWFQ) is a scheduling mechanism intended for handling congestions while providing greater flexibility i.e When one class of traffic is not using its allocated bandwidth, it allows the other class to utilise its bandwidth and allow for overflowing as stated by [20]. It is usable in situations where we want to provide a proper amount of the bandwidth to a specific application. In these cases, the network administrator must provide classes with defined bandwidth amounts, where one of the classes is, for example, intended for a videoconferencing application, another for VoIP application, and so on. Instead of waiting-queue assurance for each individual traffic flow, CBWFQ determines different traffic flows. A minimal bandwidth is assured for each of such classes. The network administrator just defines, for example, the video-conferencing class and installs a video session into that class. The same principle can be used for all other applications which need specific amounts of the bandwidth. Such classes are served by a flow-based WFQ algorithm which allocates the remaining bandwidth to other active applications within the network.

The strength of this scheduling mechanism is that it allows coping among flows with considerably different bandwidth requirements. This is done by assigning a specific percentage of the link bandwidth to each queue. CBWFQ also avoids the bandwidth starvation problem of the PQ scheduling mechanism as at least one packet is served from each queue during each service round. CBWFQ allows the user to reserve a minimum bandwidth for a class during congestion. However, this scheme does not work well for voice traffic, which is intolerant of delay. Delay in voice traffic results in irregular transmission causing jitter in the heard conversation.

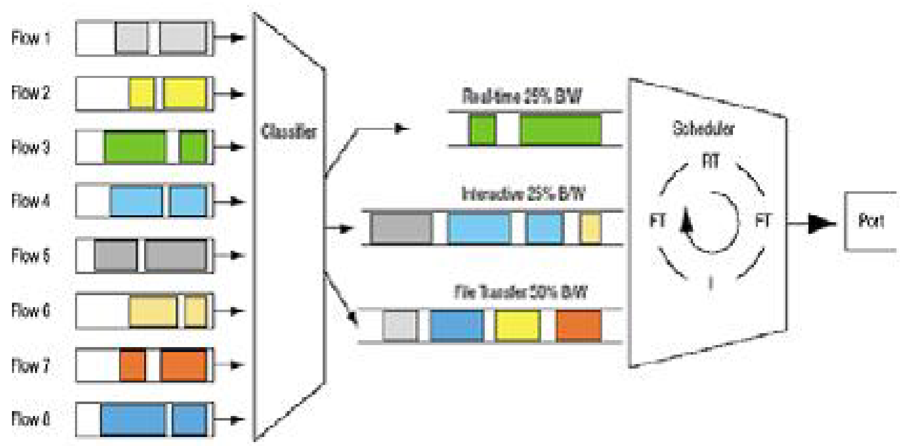


Figure 2: Class-based weighted fair queuing (CBWFQ). [19]

As shown in the figure 2 above in this mechanism packets are classified into a service class according certain criteria (addresses, protocol, users, etc...) and stored in the corresponding queue. Each queue is served in round robin order. A different amount of bandwidth can be assigned for each queue in two different ways: allowing a queue to send more than one packet for each service round or allowing a queue to send only one packet for each service round but the same queue is visited more than one time during the same service round.

### 2.2.2 Priority queuing

As defined by [17] Priority queuing (PQ) is the basis for a class of queue scheduling algorithms that are designed to provide a relatively simple method of supporting differentiated service classes. This technique uses multiple queues to a network interface with each queue being given a priority level. The queues with higher priority are serviced. Priority Queuing has four preconfigured queues, high medium, normal and low priority queue. By default each of these queues has 20, 40, 60 and 80 packets capacity as stated by [18]. Packets are placed in one of the queues according to their classification and scheduled from a certain queue only if the higher priority queues are empty. Packets from the highest priority queue are scheduled and served in FIFO order within each priority queue.

As shown in the figure 1 below.

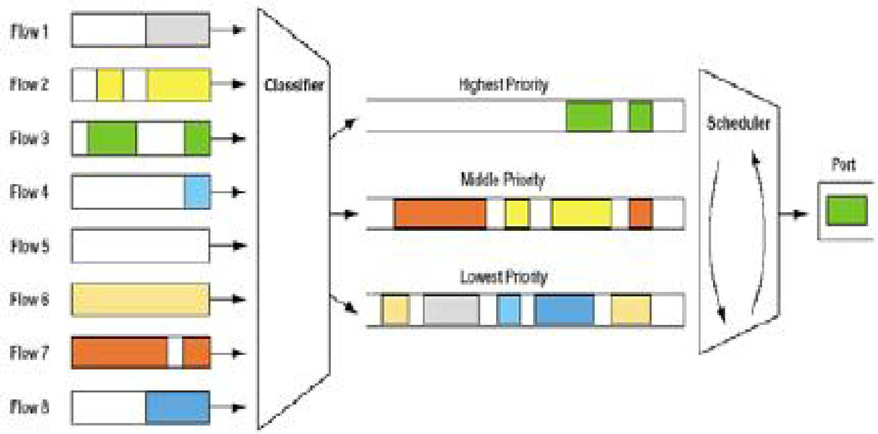


Figure 3: Priority Queuing (PQ) [19]

When the highest priority queue is empty, the remaining queues will be served using the same principle. The main benefit of PQ is that packets of different classes can be managed using different queues and consequently, a certain class of traffic can be handled in different way than another one. Another advantage is that the PQ technique is relatively simple when compared with other more elaborated queue disciplines. The maijor drawback of PQ is that if the amount of highest priority traffic is excessive then the lower priority queue may not get any service until the highest priority traffic is served completely. During this while, the queues allocated to lower priority traffic may overflow. As a result, the lower priority traffic may experience a large delay or, in the worst case, a complete resource starvation. A solution to this important problem consists of limiting the maximum amount of highest priority traffic (in general, limiting all the priority queues) that can be accepted using appropriate admission control policies.

### 2.2.3 Hybrid Queuing PQ - CBWFQ

(PQ-CBWFQ) is a queuing scheme that adds strict priority queuing to CBWFQ. Strict-priority queuing allows for delay-sensitive data, such as voice, to be sent first, before packets in other queues are de-queued. Unlike the plain old PQ, whereby the higher-priority queues might not give a chance to the lower-priority queues and effectively starve them. PQ-CBWFQ strict-priority queue is policed. This means that the PQ-CBWFQ strict-priority queue is a priority queue with a minimum bandwidth guarantee, but at the time of congestion, it cannot transmit more data than its bandwidth permits. If more traffic arrives than the strict-priority queue can transmit, it is dropped. Hence, at times of congestion, other queues do not starve, and get their share of the interface bandwidth to transmit their traffic according to [5]

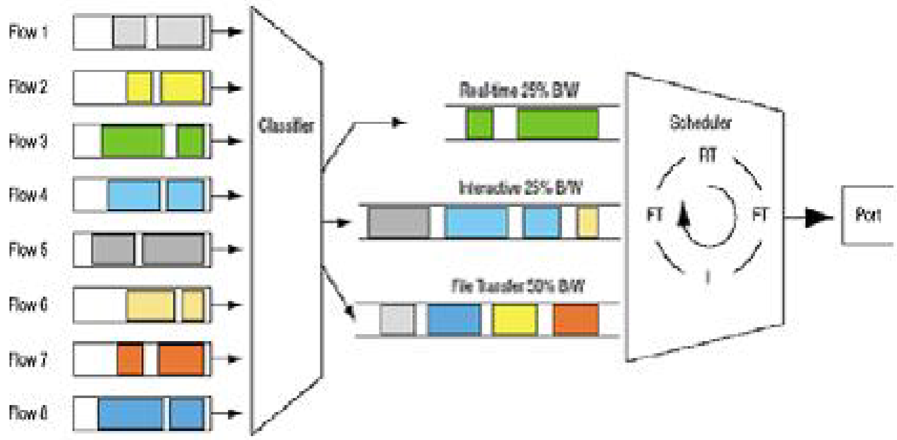


Figure 4: Priority - Class-based weighted fair queuing (PQ-CBWFQ). [21]

As shown in the figure 3 above the incoming traffic is arranged into waiting queues according to the priorities in the individual packets and also by their importance to four different internal priority queues. The output interface algorithm first serves the highest-priority data stream (packets that are in the queue with the highest importance) and then all other lower ranking queues. Once the packets appear at the outgoing interface of the priority queuing mechanism, they are again scheduled into admin-defined classes of CBWFQ mechanism. Such defined classes already have the needed bandwidth pre-reserved as set by the network administrator. This way the packets at the CBWFQ mechanism output interface do not need to fight for bandwidth, as it is guaranteed in advance. This accelerates the transfer of high priority flows, and such flows become independent of all other lower-priority flows.

It is possible to have more than one strict priority queue. One queue for voice and the other for video traffic which can be separately policed. However, after policing is applied, the traffic from the two classes is not separated. It is sent to the hardware queue based on its arrival order (FIFO). As long as the traffic that is assigned to the strict priority class does not exceed its bandwidth limit and is not policed and dropped, it gets through the PQ-CBWFQ with minimal delay.

## 2.3 Existing Algorithms Implementing PQ-CBWFQ Mechanism

Time-sensitive applications require different Quality of Service (QoS) in terms of delay and throughput in a resource constrained wireless network. PQ-CBWFQ places delay sensitive applications such as voice and video in the high priority queues and treats them preferentially over other traffic by allowing the applications to be processed and sent first from these queues.

A model has been developed to give a higher priority to voice and video traffic which is the most sensitive as stated in [22]. The model monitors all incoming traffic and categorizes it based on the level of their sensitivity. Then, it assigns the highest priority to voice and video traffic and a lower priority to other traffic which are delay tolerant. This sequence occurs so that high priority traffic can be delivered to the destination directly without considering a congestion avoidance technique. Another novel Low Latency and Efficient Packet Scheduling (LLEPS) algorithm is developed to ensure low latency for real time audio and video streaming applications [23]. The behavior of queues and their traffic is monitored to address the buffer under-run problem.

PQ-CBWFQ related simulations have been done using OPNET Modeler. It is an extensive and powerful simulation software tool with wide variety of capabilities. It enables the possibility to model a network structure with various protocols. The main goal in these simulations is oriented towards improving the network performances with regard to the VoIP end-to-end delay, Ethernet delay, and jitter. Web and FTP applications are applied only for creating low-priority traffic flows.

Shaimaa and others in [24] have demonstrated that the combination of Class-Based Weighted Fair Queuing and Low Latency Queuing (CB-WFQ-LLQ) improved the performance of multimedia applications. The only class able to provide low latency in CB-WFQ-LLQ is a single priority queue [25]. If the Video conference and the Voice traffic are merged together into one class, the bursty, large-packet Video stream would severely punish the small-packet Voice traffic.If a bursty video packet comes, the voice traffic also may not be delivered successfully [26]. The reason is that the behavior of the voice traffic is controllable whereas the video traffic is uncontrollable.

### 2.3.1 Synthesis of the literature review

The increase in use of wireless communication technologies has increased the demand for the real time audio and video applications in wireless networks. In the resource constrained wireless networks these applications require different Quality of Service (QoS) in terms of delay and throughput and thus packet scheduling algorithms such as PQ-CBWFQ have been developed. PQ-CBWFQ places delay sensitive applications such as voice and video in the PQ and treats them preferentially over other traffic by allowing the application to be processed and sent first from the PQ. Although it is possible to enqueue various types of real-time traffic in PQ-CBWFQ, its been noted that having multiple real time applications such as video and voice could introduce variation in delay, thereby thwarting the steadiness of delay required for successful voice traffic transmission in the priority queue. In this research we propose an algorithm that will make use of two separate dedicated priority queues. One for scheduling the video applications and another for the voice applications. This algorithm will be able to decide the selection of the strict priority queues for scheduling the packets dynamically.

## 2.4 Conclusion

In Section 2.1 of this Chapter we discussed the increase in the number of system communication users which has led to the increase in video streaming services and real-time applications such as Web Server, VoIP, E-mail, FTP and Voice Server. QoS is considered as potential of the network to produce consistently high quality voice transmissions. In real time applications different parameters (Delay, Jitter and Packet Loss) can be used to measure the QoS of VoIP. Avoiding and managing network congestion and shaping network traffic are provided by using different queuing disciplines, which are a basic part of QoS assurance.

In order to provide a preferred level of service for high-priority traffic, some form of software queuing must be used as discussed in Section 2.2 of this Chapter. Many different hybrid queuing methods are possible but LLQ which is the combination of PQ and CBWFQ is a queuing scheme that adds strict priority queuing to CBWFQ. Strict-priority queuing allows for delay-sensitive data, such as voice, to be sent first, before packets in other queues are de-queued. In Section 2.2.1, 2.2.2 and 2.2.3 of this Chapter we discuss the merits and demerits of the individual queuing methods PQ and CBWFQ and the merits and demerits of the hybrid queuing method PQ-CBWFQ.

In section 2.3 of this Chapter we are analysed and discussed the different existing models such as the one in [22]. This model monitors all incoming traffic and categorizes it based on the level of their sensitivity. Then, it assigns the highest priority to voice and video traffic and a lower priority to other traffic which are delay tolerant. Another model the novel Low Latency and Efficient Packet Scheduling (LLEPS) algorithm is developed to ensure low latency for real time audio and video streaming applications as stated in [23]. Unfortunately if the Video conference and the Voice traffic are merged together into one class, the bursty, large-packet Video stream would severely punish the small-packet Voice traffic. Therefore in Section 2.3.1 of this Chapter we propose an algorithm that will dynamically make use of two separate dedicated priority queues. One for scheduling the video applications and another for the voice applications.

# 3 METHODOLOGY

This chapter presents the methodology of the study. It deals with many different methods that will be used in carrying out the the specific objectives of the study in general terms. This chapter discusses the main methodological aspects that includes data collection methods, data interpretation and analysis, algorithm design and evaluation.

## 3.1 Literature Survey

In this study various documented information and other case studies in the current or in the developing areas of queueing mechanisms will be sourced from scientific journals, research published papers, textbooks, thesis/dissertations, papers on international proceedings, and the Internet in order to identify different types of time sensitive applications and how they are considered to be critical to the network performance. Analysis of the existing simulations will be done using existing simulation results.

## 3.2 Algorithm Design

A new algorithm to address the problems resulting from the allocation and division of bandwidth to the Priority Queue when using the PQ-CBWFQ queuing discipline on other time sensitive applications together with VOIP is designed using flow charts.

## 3.3 Implementation of the proposed algorithm

Simulation is a valuable tool to verify and evaluate the performance of networks. The most commonly used tools include OPNET, NS-2, NS-3, OMNeT++ and QualNet. Currently, QualNet [28] mainly for wireless networks and OPNET Modeler [29] are known as a commercial simulation but in this research we will use the academic OPNET version. OPNET simulator is a tool to simulate the behavior and performance of any type of network. The main difference with other simulators lies in its power and versatility. It is very useful when working with complex networks with a big number of devices and traffic flows, or in networks where a little change could be critical. Before implementing any change, it is possible to predict the behavior and to verify the configurations of the devices. OPNET has different tools that allow administrators to analyze their networks and the future implementations they want to do.

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# 4 APPENDICES

## 4.1 Appendix 1: Work plan and Timeframe

This is the schedule or timetable of activities and the period in which the research is to be conducted with due regard to budgetary limitation.

| Activity | Duration | Dates |
|---|---|---|
| Submission & approval of the proposal | 5 – 10 days | August |
| Design of a research plan | 5 – 10 days | August |
| Literature review | 10 days | August |
| Algorithm design & implementation | 10 – 15 days | August |
| Analysis of quanitative data | 10 days | August |
| Report up of findings | 5 – 10 days | August |
| Presentation of final research product(s) | 5 – 10 days | September |

Table 2: Work plan and Timeframe