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UNIVERSITI TEKNOLOGI PETRONAS

A VOICE PRIORITY QUEUE (VPQ) SCHEDULER FOR VOIP OVER WLANs

by

KASHIF NISAR

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A VOICE PRIORITY QUEUE (VPQ) SCHEDULER FOR VOIP OVER WLANs

by

KASHIF NISAR

A Thesis

Submitted to the Postgraduate Studies Programme

as a Requirement for the Degree of

DOCTOR OF PHILOSOPHY

COMPUTER AND INFORMATION SCIENCES DEPARTMENT

UNIVERSITI TEKNOLOGI PETRONAS

BANDAR SERI ISKANDAR,

PERAK

APRIL 2011

DECLARATION OF THESIS

Title of thesis	A VOICE PRIORITY QUEUE (VPQ) SCHEDULER FOR VOIP OVER WLANs

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Hereby declare that the thesis is based on my original work except for quotations and citations which have been duly acknowledged. I also declare that it has not been previously or concurrently submitted for any other degree at UTP or other institutions.

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In the name of the Allah, the most gracious and the most merciful

I would like to dedicate my thesis to my beloved mother and my father.

Without their support, my vision to complete this research could hardly be realize

ACKNOWLEDGEMENTS

Firstly, I would like to thanks ALLAH the almighty, who blessed me with the power and courage to endlessly pursue what is so called knowledge. Secondly, I would like to express my utmost gratitude to Universiti Teknologi PETRONAS for supporting this research work.

Thirdly, I would like to thank my supervisors, Associate Professor Dr. Abas Bin Md Said and Dr. Halabi Bin Hasbullah for being generous and understanding persons. Without their help, the achievements in my research would never have been possible.

During this research work, the author has received support from the knowledge and advice of several individuals. I sincerely appreciate the technical help from Associate Professor Dr. Wan Rozaini Binti Sheik Osman, Director of the International Telecommunication Union, Universiti Utara Malaysia (ITU-UUM), Asia Pacific-Centre of Excellence (ASP-CoE) and Associate Professor Abdul Ghani Bin Golamdin, Managing Director/CEO, R&D SerindIT Com Sdn Bhd. Next, I would like to express my thanks to my friends who have truly proven their friendship, encouragement and support throughout my research work.

Last but not least, I would like to express my gratitude to my dear parents and my supporting family, who have helped me to grow as an independent person and guided me throughout my life.

ABSTRACT

The Voice over Internet Protocol (VoIP) application has observed the fastest growth in the world of telecommunication. The Wireless Local Area Network (WLAN) is the most assuring of technologies among the wireless networks, which has facilitated high-rate voice services at low cost and good flexibility. In a voice conversation, each client works as a sender and as a receiver depending on the direction of traffic flow over the network.

A VoIP application requires a higher throughput, less packet loss and a higher fairness index over the network. The packets of VoIP streaming may experience drops because of the competition among the different kinds of traffic flow over the network. A VoIP application is also sensitive to delay and requires the voice packets to arrive on time from the sender to the receiver side without any delay over WLANs.

The scheduling system model for VoIP traffic is still an unresolved problem. A new traffic scheduler is necessary to offer higher throughput and a higher fairness index for a VoIP application. The objectives of this thesis are to propose a new scheduler and algorithms that support the VoIP application and to evaluate, validate and verify the newly proposed scheduler and algorithms with the existing scheduling algorithms over WLANs through simulation and experimental environment.

We proposed a new Voice Priority Queue (VPQ) scheduling system model and algorithms to solve scheduling issues. VPQ system model is implemented in three stages. The first stage of the model is to ensure efficiency by producing a higher throughput and fairness for VoIP packets. The second stage will be designed for bursty Virtual-VoIP Flow (Virtual-VF) while the third stage is a Switch Movement (SM) technique. Furthermore, we compared the VPQ scheduler with other well known schedulers and algorithms. We observed in our simulation and experimental environment that the VPQ provides better results for the VoIP over WLANs.

ABSTRAK

Aplikasi Voice over Internet Protocol (VoIP) dilihat telah mengalami pertumbuhan paling pantas dalam dunia telekomunikasi. Wireless Local Area Network (WLAN) adalah teknologi yang paling meyakinkan di antara rangkaian tanpa wayar yang telah digunakan dalam perkhidmatan suara berkualiti tinggi dengan kos yang rendah dan fleksibiliti yang baik. Dalam perbualan suara, setiap pelanggan bekerja sebagai penghantar dan sebagai penerima bergantung pada arah arus lalu lintas melalui rangkaian.

Sebuah aplikasi VoIP memerlukan throughput yang lebih tinggi, kurang kehilangan paket dan indeks fairness yang lebih tinggi melalui rangkaian. Paket VoIP streaming mungkin mengalami kehilangan kerana persaingan antara pelbagai jenis arus lalu lintas melalui rangkaian. Sebuah aplikasi VoIP juga sensitif terhadap delay dan dalam rangkaian WLAN perlu dipastikan pakej suara dapat dihantar dari penghantar ke penerima tepat pada masanya tanpa sebarang gangguan delay.

Sistem model penjadualan untuk lalu lintas VoIP masih merupakan masalah yang belum dapat diselesaikan. Sebuah penjadualan lalu lintas yang baru diperlukan untuk menawarkan throughput dan indeks fairness yang lebih tinggi untuk aplikasi VoIP. Tujuan tesis ini adalah untuk menawarkan penjadualan dan algoritma yang baru untuk menyokong aplikasi VoIP disamping untuk menilai, mengesahkan dan menyemak penjadualan dan algoritma baru yang dicadangkan berbanding dengan algoritma penjadualan yang sedia ada dalam rangkaian WLAN melalui aplikasi simulasi dan eksperimen.

Kami mencadangkan model sistem penjadualan dan algoritma Voice Priority Queue (VPQ) yang baru untuk menyelesaikan masalah penjadualan. Model sistem VPQ ini dilaksanakan dalam tiga tahap.

Tahap pertama dari model ini adalah untuk memastikan kecekapan dengan menghasilkan throughput fairness yang lebih tinggi dan untuk paket VoIP. Tahap kedua akan direka untuk Arus Virtual-VoIP bursty (Virtual-VF) sedangkan tahap ketiga adalah teknik Switch Movement (SM). Selanjutnya, kami membandingkan penjadualan VPQ dengan penjadualan dan algoritma lain yang telah dikenali. Hasil dari proses simulasi dan eksprerimen kami mendapati bahawa VPQ memberikan hasil yang lebih baik untuk VoIP melalui WLAN.

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LIST OF ABBREVIATIONS

ACK	Acknowledgement
ADRR	Airtime Deficit Round Robin
AP	Access Point
BSS	Basic Service Set
CAPS	Controlled Access Phase Scheduling
CBQ	Class-Based Queueing
CBR	Constant Bit Rate
CBU	Class-Based Queuing
CB-SCFQ	Credit Based-SCFQ
CIS	Computer and Information Sciences
CSMA/CA	Carrier Sense Multiple Access with/Collision Avoidance
CTS	Clear To Send
CATS	Contention-Aware Temporally fair Scheduling
CW	Contention Window
D-CATS	Decentralized-CATS
DCF	Distributed Coordination Function
DIFS	Distributed Interframe Space
DiffServ	Differentiated Services
DO-WF2Q	Worst-case Fair Weighted Fair Queuing
DRR	Deficit Round Robin
DDRR	Dynamic Deficit Round Robin
DWFQ	Dynamic Weighted Fair Queuing
DWFS	Dynamic Weighted Fair Scheduling
DWFSS	Dynamic Weighted Fair Scheduling Scheme
DTT	Deficit Transmission Time
DTF	Dequeue Traffic Flows
EFS	Efficient Fair Scheduling

EDCA	Enhanced Distributed Channel Access
EIFS	Extended Interframe Space
ESS	Extended Service Set
FAHPS	Fair HCCA Priority Scheduling
FAS	Feedback-Assisted Scheduling
FCFS	First Come First Serve
FIFO	First-In-First-Out
FQ	Fair Queueing
FFQ	Frame-Based Fair Queueing
FTP	File Transfer Protocol
GPS	General Processor Sharing
HCCA	HCF Controlled Channel Access
HCF	Hybrid Coordination Function
HDTV	High definition TV
HP-VC	High Performance-Video Conferencing
ICT	Information and Communication Technology
IEEE	Institute of Electrical and Electronics Engineers
IFS	Interframe Space
iLBC	Internet Low Bit Rate Codec
InterServ	Integrated Services
IP	Internet Protocol
IPTV	Internet Protocol TV
ISDN	Integrated Services Digital Network
ITU	International Telecommunication Union
LL	Link Layer
LLEPS	Low Latency Efficient Packet Scheduling
LLC	Logical Link Control
LLQ	Low Latency Queueing
MAC	Medium Access Control
MAHS	Multiple Access Hybrid Scheduling
MH	Mobile Nodes
MOS	Mean Opinion Score
MSI	Maximum Service Intervals

NAM	Network Animator
NAV	Network Allocation Vector
NB	Non-Bursty
NGN	Next Generation Network
NIC	Network Interface Card
NS-2	Network Simulator-2
Non-VFB	Non VoIP Flow Buffer
NVF	Non VoIP Flow
ORR	Ordered Round-Robin
OSI	Open Systems Interconnection
PCF	Point Coordination Function
PCM	Pulse Code Modulation
PHY	Physical Layer
PIFS	PCF-Interframe Space
PQ	Priority Queue
PSTN	Public Switched Telephone Network
QoS	Quality of Service
QP-CAT	QP-Computation of Additional Transmission
QSTAs	Quality of Service Stations
RED	Random Early Detection
RR	Round Robin
RSVP	Resource Reservation Protocol
RTCP	Real Time Transport Control Protocol (IETF)
RTP	Real Time Transport Protocol (IETF)
RT	Real-Time Traffic
RTS	Request To Send
SCFQ	Self-Clocked Fair Queueing
SFQ	Start-Time Fair Queueing
SM	Switch Movement
SNR	Signal to Noise Ratio
SIFS	Short Interframe Space
SIP	Session Initiation Protocol (IETF)

CHAPTER 1

INTRODUCTION

1.1 VoIP Background

Transmission of Voice over Internet Protocol (VoIP) on packet switching networks is one of the rapidly emerging real-time applications. The VoIP is a form of audio and voice communication. It receives voice signal activities which are then encoded in digital form and divided into small parts of information in the form of voice data network packets. These data network packets are decoded and transmitted as voice signals then the sender and receiver have a voice conversation [1], [2]. In a voice conversation, the clients send and receive packets in a bidirectional method. Each client works as a sender and as a receiver depending on the direction of traffic flow over the network [3].

The VoIP is making very cheap Cell-Phone or IP-Phone calls locally and internationally. The VoIP is gaining attractiveness as a technique to apply business communication anywhere and anytime. The VoIP is deployed on a Wireless Local Area Network (WLAN), based on IEEE 802.11 standards. Combined, these two applications have been growing as an infrastructure to provide the high quality speech for real-time voice applications [4].

Right now, there are approximately one billion fixed telephony lines and two billion mobile-phones in the world [5]. These connections are moving to IP-based networks such as VoIP applications. The VoIP is an essential part of research in the world of telecommunication. The International Telecommunication Union (ITU) describes VoIP as the transmission of voice, audio and associated technologies over packet switched networks [6]. The high profit made by the telecommunication business is a motivation to increase solutions for transmitting voice traffic over other applications rather than the traditional, circuit switching network [7].

1.2 VoIP Network Systems

Figure 1.1 describes the processing component involved in transmitting voice traffic over IP-based packet networks from the sender to the receiver. The VoIP system structure from the sender side is based on analogue voice signals with bandwidth 4 KHz. The ITU has standardized many encoding schemes.

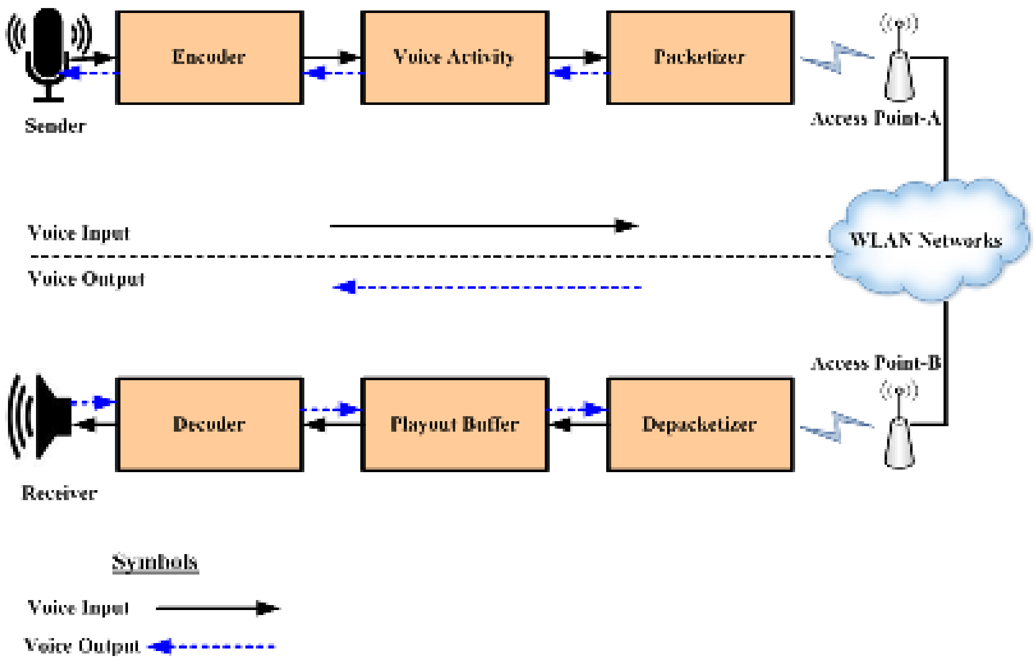


Figure 1.1 VoIP Network Systems

The most utilized codec is G.711 which is based on the compression method of Pulse Code Modulation (PCM). It generates a digitalized signal with the following characteristics of G.711: bit-rate of 64 kb/s, frame of G.711 is 0.125 ms, frame size of 8 bits per frame and Mean Opinion Score (MOS) of 4.1 [8], [9] and [10]. The encoded voice activity is then packetized into small parts of packets. The VoIP system structure from an IP-based network includes the internet backbone transmission, WLAN IEEE 802.11a/b/g and Access Point (AP) with an omni-direction antenna. The VoIP system structure from the receiver side has a depacketized and playout buffer to provide a control for decompression. The content of the received voice packets is sent to the decoder for packet loss concealment and again analogue voice signals for audio or voice conversion [11], [12].

1.3 VoIP Protocol Architecture

Figure 1.2 shows the fundamentals of VoIP protocol stack architecture to implement a VoIP network system over WLANs [13], [14]. Voice packets are transmitted over IP-based networks. VoIP is a real-time application and transmits the voice on a Real-Time Transport Protocol (RTP), User Datagram Protocol (UDP) and Internet Protocol (IP) over networks [15]. Each voice packet is small in size and the voice packet has the headers: RTP (12 bytes), UDP (8 bytes), and IP (20 bytes) headers. The data-link layer Medium Access Control (MAC) has a (34 bytes) header. All these headers sum up to 74 bytes of overhead in the VoIP application.

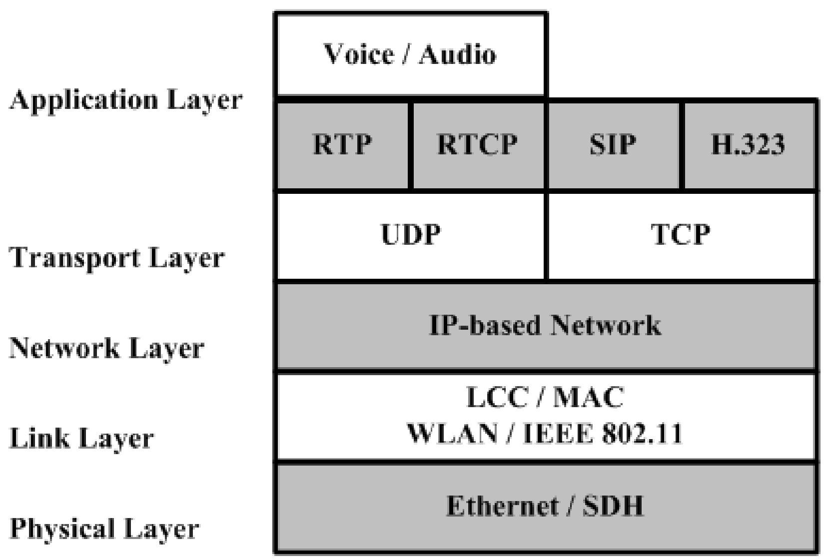


Figure 1.2 VoIP Protocol Architecture

The Session Initiation Protocol (SIP) was considered to handle a multimedia call setup and H.323 is considered by ITU to allocate IP-based phones on the public telephone network to talk to a PC-based phone over IP-based networks [16]. It is a standard that specifies the components, protocols and procedures for multimedia communication services such as real-time audio, video and data communications over IP-based packet networks [17], [18], [19] and [20].

1.4 VoIP over WLANs

The VoIP over WLANs have observed the fastest growth in the world of communication. The WLAN is the most assuring of technologies among the wireless networks, which has facilitated high-rate voice services at low cost and good flexibility over IP-based networks [21].

The main reason for such adaptation is that a VoIP real-time application is more flexible than traditional public switched telephone network systems (PSTN) [22]. Moreover, the VoIP can support multiple infrastructure environment IP-based-Phones (IP-Phone, PC-based soft-Phone, IP-based Packet-Phone), Soft-Phones (PC-to-PC Phones) as well as Traditional Phones and Mobile Phones (Telephone, Cell-Phone). Details are as shown in Table 1.1.

Table 1.1 Classifications of Phone Systems

Traditional Phones	IP-based Phones	Soft and Hard Phones
Dialup Phone	IP-Phone	PC-to-PC Phone
Telephone	PC-based Phone	PC-to-Phone
Cell-Phone	PC-based Soft Phone	Phone-to-PC

The VoIP provides mixed-mode communication with PC-to-PC, PC-to-IP-Phone and PC-to-Cell-Phone over WLANs. WLANs are implemented in campuses, hotels, educational institutions, airports, health care facilities, commercial areas and industries to provide voice traffic. WLANs also provide audio, voice and video conferencing over IP-based networks [23] and [24].

The VoIP over a WLAN environment assigns to the user IP-based calls over a WLAN to the global networks. In IP-based networks, analogue voice signals are digitized and moved on a real-time transmission over the network. They find the most efficient path to reach the proposed destination. Normally, they are not in the original order. The receiver side packets are rearranged in the proper order before being converted into analogue voice signals.

In WLANs, there are two essential kinds of the service architectures: ad-hoc architecture and infrastructure architecture. In ad-hoc architecture, a station (STA, a mobile node) can be able to connect with an IP-based network without the connectivity to any wired backbone network and without the need of an Access Point (AP) [25].

In infrastructure architecture, the STA can be able to connect with an IP-based network with the connectivity to any wired backbone network and with the need of an AP. In this thesis, it will focus on an infrastructure architecture network where VoIP traffic is transmitted as signals via an AP. WLANs provide a number of industry standards of AP. Each AP can maintain a restricted number of parallel voice nodes [26].

Figure 1.3 shows a VoIP over WLAN architecture with two voice sources from the VoIP traffic server. WLANs support both wired and wireless applications. Voice sources are given two traffic paths; one is Access Point-A (AP-A) and the other is Access Point-B (AP-B) with a Basic Service Set (BSS) and an Extended Service Set (ESS). VoIP gives a number of real-time VoIP sessions in the WLANs [27].

An AP can support (10) to (16) Mobile Nodes (MN) over 802.11b on the G.711 codec technique over an infrastructure architecture network [28]. Normally, an AP is positioned as a central direction with communication for MN over WLANs. The bidirectional communication describes the uplink voice flow transmitted by the VoIP client and the downlink voice flow transmitted by the AP. The AP is usually present as the gateway between the wired node and the wireless node VoIP clients.

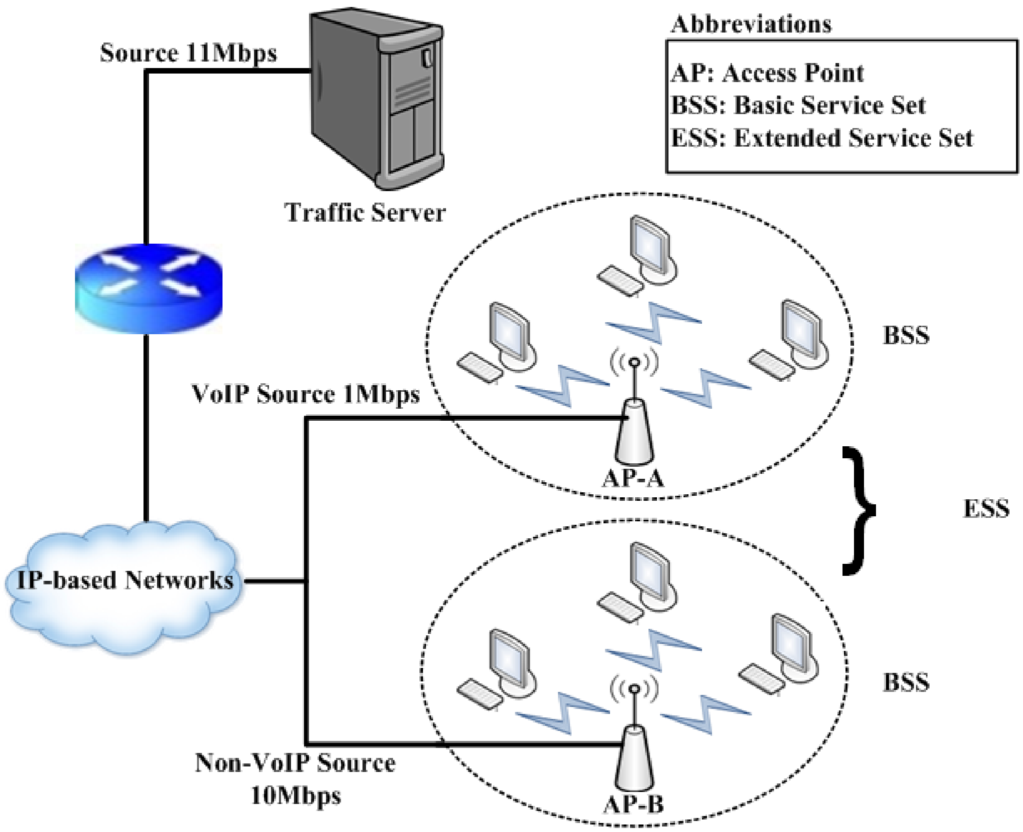


Figure 1.3 VoIP over a WLAN Network

1.5 VoIP over WLANs using IEEE 802.11 Standards and MAC

The IEEE 802.11 WLAN is a wireless Ethernet, playing an important function in the Next Generation Networks (NGNs). The WLAN is based on Link Layer (LL). LL is divided into Logical Link Control (LLC) and Medium Access Control (MAC) sub-layer categories with two functions, the Distributed Coordination Function (DCF) and the Point Coordination Function (PCF) [29], [30] and [31]. The IEEE 802.11 WLANs support both contention-based DCF and contention-free PCF functions. DCF uses Carrier Sensing Multiple Access/Collision Avoidance (CSMA/CA) as the access method [32] and [33]. IEEE 802.11 standards 802.11a support 54Mbps data rate and 5GHz frequency, 802.11b support 11Mbps data rate and 2.4GHz frequency, 802.11g support data rate 54Mbps and 2.4GHz frequency. Details are as shown in Table 1.2 [34] and [35].

Table 1.2 WLAN using IEEE 802.11 Standards

	IEEE 802.11a	IEEE 802.11b	IEEE 802.11g
Data Rates	54Mbps	11Mbps	54Mbps
Frequency	5GHz	2.4GHz	2.4GHz

1.6 Queue Schedulers and Algorithms

A variety of new multimedia applications such as the VoIP, video on demand (VoD), Internet Protocol TV (IPTV) and teleconferencing are based on network traffic scheduling algorithms. The VoIP utilizes IP-based WLAN using IEEE 802.11 standards.

A combination of these two technologies and collaboration with NGNs might be a leading application in the world of communication [36]. A number of research solutions have been proposed to satisfy different Quality of Service (QoS) requirements.

For audio speech quality in packet switch network applications, the main concern is end-to-end delay, fairness, throughput and packet loss. A number of packet scheduling algorithms have been introduced. Weighted Fair Queueing (WFQ) and Start-Time Fair Queueing (SFQ) were mainly designed to provide the bandwidth reservation. The Low Latency Queueing (LLQ) addresses this problem [37].

The Strict Priority (SP) is low-cost to maintain the delay sensitive voice traffic. Moreover, a number of research scheduling solutions have been proposed for the IP-based network [38] such as Class Based Queueing (CBQ), General Processor Sharing (GPS), Fair Queueing (FQ), Deficit Round Robin (DRR), Nested Deficit Round Robin (NDRR), Dynamic Deficit Round Robin (DDRR) and Low Latency Efficient Packet Scheduling (LLEPS).

1.7 Problem Statement

Quality of Services (QoS) is considered as the main issue in VoIP systems [39]. A VoIP application requires a higher throughput, less packet loss, and a higher fairness index over the network. The packets of VoIP streaming may experience drops because of the competition among the different kinds of traffic flow over the network. Therefore, the quality of streaming applications cannot be guaranteed. A VoIP application is also sensitive to delay and requires the voice packets to arrive on time from the sender to receiver side without any delay over WLANs.

IP-based networks manage voice, data, web browsing, email and video applications on the same network flow over WLANs. However, they are not specifically designed to transmit real-time applications over WLANs and that may cause a bottleneck problem. Figure 1.4 shows a bottleneck topology of mixed-mode traffic over a WLAN.

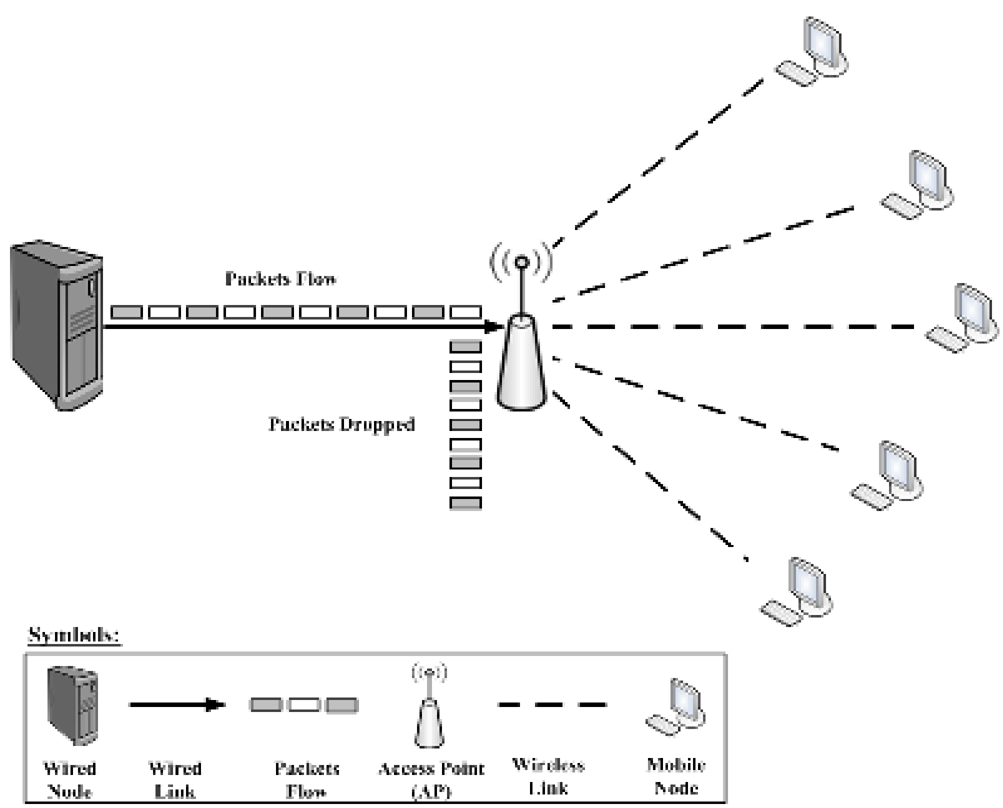


Figure 1.4 Bottleneck Topology of mixed-mode Traffic over WLANs

The above problems degrade the QoS of the VoIP over IP-based networks. Figure 1.4 shows the details of bottleneck of mixed-mode traffic over WLANs. It is based on backend nodes that are connected with a wired network and front-end nodes that are connected with one Access Point (AP) over WLANs. AP node is similar to gateway between wired and wirless nodes and allows packets to exchange between two types of nodes.

A new traffic scheduling system model is necessary to offer QoS for a VoIP application over WLANs using IEEE 802.11 standards. Due to this, research focuses on addressing the VoIP scheduling algorithm issues. The new method should be fair, provide a higher throughput and a bandwidth guarantee that will enhance performance of the VoIP over WLANs. This research plans to compare some well-know scheduling algorithms over WLANs.

Through the past decades many schedulers were introduced to solve real-time traffic application issues. These schedulers can be divided into three groups: namely, packet-based schedulers, frame-based packet schedulers and regulative packet schedulers. It will discuss these schedulers in more detail in chapter two. The proposed method tries to achieve better acceptable results for the VoIP high-speed real-time application. The VoIP is a real-time application that needs timely techniques to enhance traffic over networks. This is a challenging task for the VoIP over WLANs.

1.8 Thesis Aim and Objectives

The aim of this thesis is to propose a new scheduling system model and algorithms that support the VoIP application over WLANs. It will study essential related work to examine the available scheduling algorithms outcome and drawbacks. It will propose new scheduling system model and algorithms to enhance the performance of the VoIP over WLANs using IEEE 802.11 standards. It will evaluate, examine and simulate techniques with related scheduling algorithms for real-time applications. To improve the real-time traffic scheduling algorithm it must be possible to resolve many of these problems.

In this research the specific objectives are as follows:

- To propose a new scheduling system model and algorithms for VoIP traffic that are able to fulfill the scheduling requirements over WLANs.

- To classify VoIP Flow (VF) traffic and Non-VoIP Flow (NVF) traffic over a WLAN using IEEE 802.11 standards.

- To evaluate, validate and verify newly proposed scheduler and algorithms with the existing algorithms over WLANs through simulation

- To validate and verify the scalability of VPQ for VF and NVF traffic over a WLAN, using a test-bed for a VoIP application over WLANs.

1.9 Contributions of This Thesis

The main contribution of this work is a new scheduling algorithm named Voice Priority Queue (VPQ) scheduler and related algorithms for real-time applications like the VoIP application. It describes several methods to enable solutions for maintaining consistent voice quality over WLANs. It is identified that the voice scheduler is the most appropriate method for monitoring the VoIP quality in IP-based networks. A detail research and literature review has been carried out to understand the fundamentals of traffic schedulers and algorithms. The significant discovery of research work is that found it performs better than other related schedulers and algorithms. The contributions of this thesis are as follows:

- The contribution of research work is a new Voice Priority Queue (VPQ) scheduling system models for the VoIP over a WLAN using IEEE 802.11 standards.

- VPQ scheduling system model that provides classification of the VoIP Flow (VF) and the Non-VoIP Flow (NVF) over a WLAN using IEEE 802.11 standards. VPQ scheduling system model is implemented in three stages. The first stage of the scheduler is to ensure efficiency by producing a higher throughput and fairness for VoIP packets. The second stage will be designed for bursty Virtual-VoIP Flow (Virtual-VF) while the third stage is a Switch Movement (SM) technique.

- Another contribution of this thesis is that proposed three stages of VPQ scheduling algorithms for verification and validation of VF over WLANs.

- VPQ scheduler performances are proven to be better than the related schedulers such as T-WFQ, CATS, D-CATS, D-CATS and CAPS.

Thus it is expected from the proposed scheduling model that VPQ will be more accurate in the VF scheduling applications over WLANs.

1.10 Structure of This Thesis

In Chapter 2, it will review related schedulers and algorithms. These include the following; Class Based Queue (CBQ), Weight Faire Queue (WFQ), Generalized Processor Sharing (GPS), Worst-case Fair Weighted Fair Queueing (WF^2Q), Deficit Round Robin (DRR), Deficit Transmission Time (DTT), Low Latency and Efficient Packet Scheduling (LLEPS), Temporally-Weight Faire Queue (T-WFQ), Controlled Access Phase Scheduling (CAPS), Contention-Aware Temporally fair Scheduling (CATS) and Decentralized-CATS (D-CATS).

In Chapter 3, it will focus on a new Voice Priority Queue (VPQ) scheduling system model and algorithms. It will explain three stages of VPQ scheduling system model for the VoIP over WLANs. VPQ classifies the traffic flow into VoIP Flow (VF) and Non-VoIP Flow (NVF), sharing the same transmission media. The VPQ provides a traffic scheduler performance technique to allocate the Priority Queue (PQ) for VF and NVF traffic over IP-based networks. VPQ-Stages and classifications of the traffic flows are the main contributions in chapter 3. This chapter further explains the basic components of VPQ which are VF and NVF Enqueue, Traffic Shaping (TS), Token Bucket (TB), VPQ-Component, VoIP Flow-Buffer (VFB) and Non-VoIP Flow Buffer (Non-VFB), VF-Switch, and VF and NVF Dequeue. In addition, it introduces VPQ algorithms over WLANs.

In Chapter 4, it will perform the simulation on topologies, design and how the experiments are carried out as well as how the tests are selected. The (Network Simulator-2) NS-2 will be used for analyzing and evaluating the results. It will present a number of different types of topologies to validate and verify the Voice Priority Queue (VPQ) scheduling system model for the VoIP over WLANs. An NS-2 environment fully supports the VoIP over WLANs. These simulations are based on all three stages of VPQ scheduling system model. It will also explain the VF and NVF traffic flows. It will define throughput, fairness, bandwidth and end-to-end delay over IP-based networks. In addition, it will arrange the experimental environment over the VoIP test-bed.

In Chapter 5, it will present the results and discuss in the findings. It compare the research work with the most related and prominent scheduling techniques over WLANs. Results show better performance than related techniques and are quite remarkable even under the bursty traffic over WLANs. Furthermore, it extends the simulation results and provides the test-bed results. It found through results that, VPQ has unique abilities for classifying VoIP Flow (VF) and Non-VoIP Flow (NVF) traffic flows. VPQ enhanced the performance of the VoIP over WLANs using IEEE 802.11 standards.

In Chapter 6, it will present the conclusion of the thesis as well as describe the achieved objectives, the limitation of the current research work, achieved contributions and the suggestions for future works. The new VPQ scheduling system model provides possible solutions for VoIP traffic that is not possible with traditional techniques over IP-based networks. The thesis outline is illustrated in Figure 1.5.

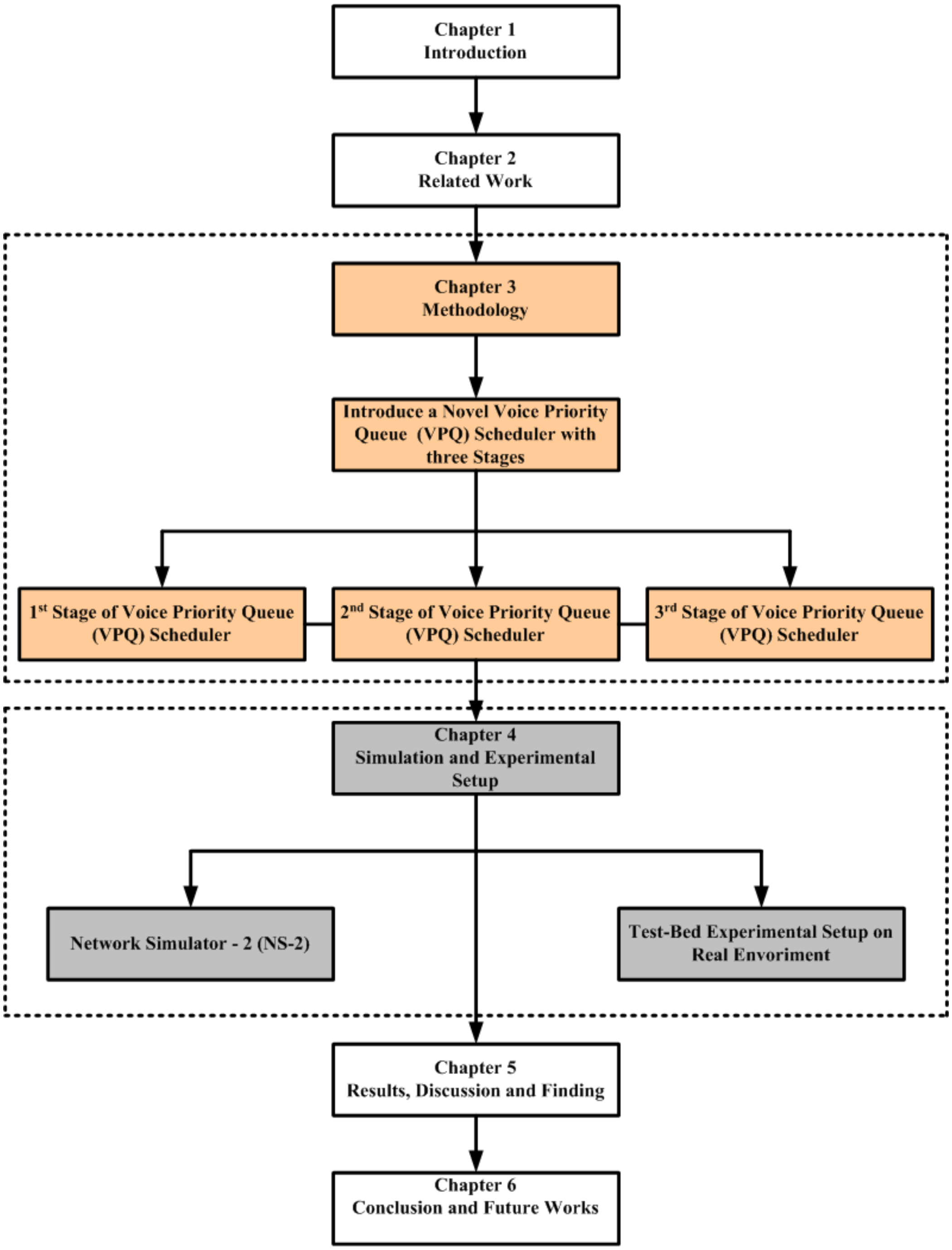


Figure 1.5 Thesis Reading Outline

CHAPTER 2

RELATED WORK

2.1 Introduction

The Voice over IP (VoIP) application is one of the most rapidly emerging technologies to utilize high-speed packet-switched networks. The IEEE 802.11 standard has increased in importance because of research in the last decade. The high-speed packet-switched networks are essential research areas in the world of telecommunication [40].

The VoIP is a delay sensitive application over packet-switched networks. A VoIP application would expect the network to ensure that each traffic flow is able to provide an efficient performance guarantee, real-time voice flow, better throughput and a fair share of the bandwidth. Packet scheduling algorithm is an important method to enhance the performance of the VoIP over WLANs. Queue management scheduling is a dynamic area of research over a WLAN which is based on the IEEE 802.11 standard.

This chapter will discuss the fundamental background of the related schedulers and algorithms. This chapter also identifies the importance of the scheduling techniques over WLANs. This chapter will also study a general discussion of related research work for real-time and specifically VoIP applications. There are a number of traffic schedulers and algorithms introduced to meet the above requirements [41], [42], [43], [44], [45], [46], [47] and [48]. In related work describe the traffic scheduling algorithms, advantages, disadvantages, and challenges.

There are some scheduling algorithms to support packet scheduling over networks. Some of them are Class Based Queue (CBQ), Faire Queue (FQ), Weight Faire Queue (WFQ), Generalized Processor Sharing (GPS), Worst-case Fair Weighted Fair Queueing (WF^{2}Q), Deficit Round Robin (DRR), Deficit Transmission Time (DTT), Low Latency and Efficient Packet Scheduling (LLEPS), Credit Based-SCFQ (CB-SCFQ), Controlled Access Phase Scheduling (CAPS), Queue size Prediction-

Computation of Additional Transmission (QP-CAT), Temporally-Weight Fair Queue (T-WFQ), Contention-Aware Temporally fair Scheduling (CATS), and Decentralized-CATS (D-CATS) [49]. These scheduling algorithms will be studied.

Most of the available traffic scheduling architecture and algorithms are for wired networks. The traditional scheduling techniques are applied without any decision on how to arrange traffic flow with real-time applications such as a VoIP application. This chapter will study the most related scheduling algorithms to ensure appropriate QoS particularly for a VoIP application.

The high-speed packet-switched networks from various flows enter at a switch or router for controlling the voice traffic. The purpose of the scheduling algorithm is to choose each voice traffic flow from the switch or router and transmit it in the output node. A better scheduling algorithm manages the various flows in an efficient and suitable manner [50], [51], [52] and [53].

2.2 Classifications of Scheduling Algorithms

The VoIP is an end-to-end delay sensitive application and requires a proper traffic scheduler algorithm over the network. In addition, it can classify scheduling system model and algorithms due to their nature of behavior over IP-based networks [54], [55] and [56]. It can classify schedulers as a packet-based scheduler, frame based-packet scheduler, bit-by-bit scheduler, priority packet-based and regulative packet scheduler. Details are as shown in Table 2.1.

The Generalized Processor Sharing (GPS) is a concept of how multiple tasks share a single processor. The process of GPS is bit-by-bit over the network and the bits need to be allocated per link under the GPS. The Class Based Queue (CBQ) exploits a bandwidth sharing mechanism for a bandwidth guarantee.

Table 2.1 Classification of Scheduling Algorithms

Scheduler Name	Authors	Classification
CBQ	S. Floyd et al. 1998 [65]	Priority Packet-based
RCSP	H. Zhang 1993 [81]	Regulative Packet Scheduler
GPS	A. Parekh 1994 [85]	Bit-by-Bit Scheduler
DRR	M. Shreedhar 1995 [87]	Frame based-packet Scheduler
DTT	R. Garroppo et al. 2007 [98]	Frame based-packet Scheduler
NDRR	S. Kanhere 2001 [93]	Frame based-packet Scheduler
DDRR	K. Yamakoshi 2002 [95]	Frame based-packet scheduler
WFQ	A. Demers et al.1989 [74]	Packet-based Scheduler
WF^2Q	J. Bennett 1996 [81]	Packet-based Scheduler
DO-WF^2Q	X. Fei 2002 [86]	Packet-based Scheduler
CB-SCFQ	E. Palacios et al. 2004 [105]	Packet-based Scheduler
MAHS	Y. Fallah 2004 [114]	Packet-based Scheduler
FAHPS, DACE-T	Y. Fallah 2005 [113]	Packet-based Scheduler
LLEPS	H. Wu et al. 2006 [54]	Packet-based Scheduler
CAPS	Y. Fallah 2007 [112]	Packet-based Scheduler
CATS	Y. Seok et al. 2007 [120]	Packet-based Scheduler
D-CATS	Y. Seok et al. 2007 [121]	Packet-based Scheduler
D-CATS+	Y. Seok et al. 2007 [121]	Packet-based Scheduler
T-WFQ	Y. Seok et al. 2007 [121]	Packet-based Scheduler
QP-CAT	S. Shin et al. 2008 [117]	Packet-based Scheduler
APP-Aware	N. Bayer 2010 [111]	Packet-based Scheduler
Proposed VPQ	K. Nisar et al. 2011 [49]	Packet-based Scheduler

CBQ, CB-SCFQ, LLEPS, CAPS, DACE-T, FAHPS, MAHS, WFQ, WF2Q, DO-WF2Q, QP-CAT, T-WFQ, Application-Aware (APP-Aware), CATS, D-CATS and VPQ are packet-based scheduling algorithms for real-time and non real-time traffic over IP-based networks. DRR, NDRR, DDRR, DTT, and Efficient Scheduler are frame-based packet scheduling algorithms and these are very similar to packet-based scheduling algorithms. RCSP and Dynamic-R&S are regulative packet scheduling algorithms.

Priority Packet-based is a utility that set up priority filters to process high priority network traffic before normal traffic. Using Priority Packet set up priority filters to give priority to time-critical traffic such as VoIP. In regulating real-time traffic stream within the network is to control jitter or the instantaneous rate over network. Bit-by-bit scheduler based on bit and data transfer into bit as compare to packets over networks. Frame-based scheduler very similar with packet-based scheduler and main difference frame are combination of many packets over the networks.

Several packet schedulers, focusing on fairness and throughput maximization, packet scheduling algorithm that can achieve power-efficient transmission while providing both system throughput gain and fairness improvement. In a mixed traffic system, a classifier is necessary for the efficiency of packet scheduling. The classifier sets independent queues based on traffic types, and each queue is given its own priority. Thus, each traffic type can be independently handled.

2.3 Real-time Traffic Scheduler

Real-time schedulers provide a guaranteed fairness facility and offer multiple sessions over the same transmission node link. These traffic schedulers divide the bandwidth in the small traffic flows and forward the traffic flow with priority over IP-based networks. Real-time flows require adequate bandwidth allocations and strict delay on the network.

Real-time traffic scheduling system model and algorithms are designed to share the bandwidth over the network. The motivation behind real-time applications considers a long-term fair index and manages the real-time application such as when the VoIP flows more smoothly on the network. There are several real-time traffic schedulers that have been introduced for a WLAN based on IEEE 802.11 standards [57].

2.4 VoIP Traffic Scheduler Issues

In WLANs, the voice traffic flow is categorized into different classes [58], [59], [60], [61] and [62]. The traffic flow guarantee issue in WLAN addresses how to assign bandwidth resources among these traffic flows so that each flow can have its own Quality of Service (QoS) requirements satisfied. Most of the schedulers perform well when the network load is not high [63], [64]. Some VoIP traffic scheduler issues relate to the minimum cost voice traffic delivery, MAC layer scheduling issues, wired to wireless traffic issue and voice traffic over loaded.

VoIP traffic scheduler algorithms are an important mechanism to guarantee the QoS requirements on high-speed packet-switched networks. The scheduling algorithm should be scalable and sufficiently manageable as the number of connections or nodes sharing the channel increases on the networks.

2.5 Traffic Scheduler

The main concern of the VoIP over WLAN is for a well-organized traffic flow on the network. Traffic scheduler techniques manage the multiple flows in a flexible approach with a bandwidth guarantee and it provides QoS over the networks. Some of these traffic schedulers are discussed in the following subsections. It will also show the difference between these schedulers and proposed scheduling system model.

2.5.1 Class Based Queue

S. Floyd et al. [65] introduced the Class Based Queue (CBQ) for hierarchical bandwidth link-sharing and the resource technique of packet-switch networks. CBQ is a class based algorithm and shares the bandwidth for each class in a well organized manner over IP-based networks. CBQ manages bandwidth link-sharing rations for all classes. It maintains each queue and provides fair link-sharing.

CBQ is one of the best solutions for data traffic over the networks [66], [67], [68] and [69]. It is implemented with a gateway technique and link-sharing. CBQ is a combination of classifier, estimator, selector and over limit processor which is used to schedule the traffic flow classes which have extended beyond the link sharing limits [70], [71] and [72].

Figure 2.1 shows how CBQ provides the solution for multiple types of traffic flow over IP-based networks. CBQ divides the bandwidth or allocates the link-sharing according to the traffic requirements. In this figure, CBQ has 6 (Mbps) bandwidth and it needs to share this bandwidth in three different kinds of traffic flows like audio (1Mbps), video (2Mbps) and File Transfer Protocol (FTP) (3Mbps) over wireless nodes.

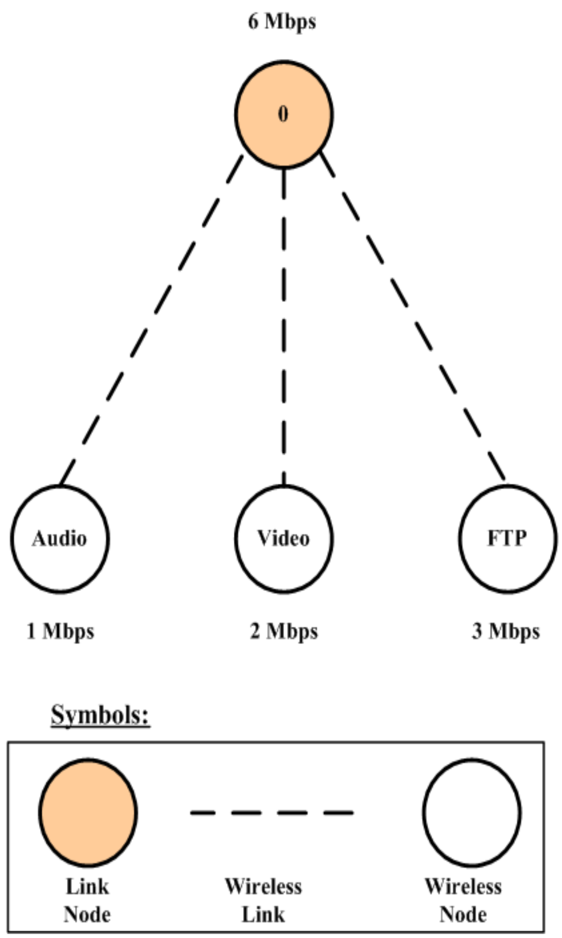


Figure 2.1 Priority, Structure of Link-Sharing Allocation

However, the drawbacks of CBQ are as follows: If bursty traffic arrives on a real-time application, it will become a cause of delay on the network. The real-time application requires a specific buffer to control the bursty traffic on IP-based networks but the bulky buffer introduces longer playback delay. The bulky buffer creates more delay time and it slows down for real-time applications particularly research areas such as the VoIP over WLANs. Hence, CBQ is not appropriate for fine grained real-time applications due to bursty traffic and therefore, not suitable for VoIP applications. CBQ does not assume the delay of packets whereas the delay of packets is required to be less and scalable for the VoIP application. Furthermore, for real-time application such as video conferencing, Internet Protocol TV (IPTV) and High definition TV (HDTV), CBQ will not fulfil the constant and low delay on WLANs [73].

2.5.2 Fair Queue

S. Keshav et al. [74] and A. Demers et al. [75] proposed the Fair Queue (FQ) scheduling algorithm based on the First-In-First-Out (FIFO) technique. It is a cost-effective and efficient scheduler. The FQ algorithm is executed at the server that manages packets on the output link of a switch or router. The flow of data from sender to receiver can be notable and saved in a logical distinct per-conversation queue.

This procedure assigns bandwidth in a fair manner; every link gets its fair share of the bandwidth over the network correctly. FQ implements traffic flow as a practical way of providing packet-by-packet transmission. The packet-by-packet transmission means that whenever a packet is completely sent, then a new packet is sent to the destination. It has three basic components; bid number computation, round number computation and packet buffering. Furthermore, it has a few alternative buffering methods; an ordered linked list (LINK), a binary Tree (TREE) and a double heap (HEAP) [76], [77] and [78].

The drawbacks of the FQ algorithm are with the high time complexity on the multiple flow structures which are extremely difficult to implement in a large scale due to the access delay. FQ only precedes the cost metric and does not consider any variance. FQ is also an initial stage of queueing algorithms and it will not be suitable for VoIP applications.

2.5.3 Weighted Fair Queue

A. Demers et al. [57] introduced the Weighted Fair Queue (WFQ) scheduling algorithm which is a sort packet-based scheduling algorithm with the latest updating of the Generalized Processor Sharing (GPS) [79], [80]. WFQ offers a number of queues at the same time with different bandwidth service rates by providing each queue a weight and arranging different percentages of output port bandwidth over the network. WFQ calculates the departure time of each packet and manages multiple sizes of packets. List of WFQ notations are as shown in Table 2.2.

Table 2.2 List of important WFQ Notations [81]

Notations	Description
$d_{i,}^{k}\ WFQ$	The time at which the $K'th$ packet on the session departs under WFQ.
$d_{i,}^{k}\ GPS$	The time at which the $K'th$ packet on the session departs under GPS.
$W_{i\,WFQ}\ (0,r)$	The total amount of the services received by the session under WFQ
$W_{i\,GPS}\ (0,r)$	The total amount of the services received by the session under GPS
i	Arrival flow
r	Link speed
t	Time of arrival flow
L_{max}	The maximum packet length.
$O(\log n)$	The work complexity of a WFQ scheduler
$L\ (i,k,t)$	The size of the k-th packet and it arrives on flow i at time t.

Meanwhile, the traffic flows of smaller packets are assigned extra bandwidth. WFQ follows the GPS fairness distribution of bandwidth in IP-based networks [82]. WFQ is also a weight-based traffic scheduler as it offers the fairness traffic flow approach with the weight of traffic class on high-speed networks. An important function of WFQ schedulers is to provide a weight to each traffic flow and get fairness by following the GPS scheduler over IP-based networks. WFQ and GPS scheduler packet system equations are as described in equation 2.1 and 2.2:

$$d_i^k, WFQ - d_i^k, GPS \leq \frac{L_{max}}{r} \quad \forall i, k \tag{2.1}$$

$$W_{i\,WFQ}\ (0, r) - W_{i\,GPS}\ (0, r) \leq L_{max} \quad \forall i, r \tag{2.2}$$

Where d_i^k, WFQ and $d_i^k,\ GPS$ are the time of the packet on task i sent to WFQ and GPS. Particularly, $W_{i\,WFQ}\ (0, r)\ and\ W_{i\,GPS}\ (0, r)$ are the total amounts of service received on task i at time r on WFQ and GPS. Finally, the length of the packet with the maximum amount is denoted as L_{max} on an IP-based network.

The WFQ complexity of computing the virtual time in the worst-case complexity is $O(n)$, where n is the number of traffic flows on the same output sharing link. Compared with a related scheduler like Self-Clocked Fair Queueing (SCFQ), Frame-Based Fair Queueing (FFQ) and Worst-Case Fair Weighted Fair Queueing (WF^2Q), the complexity of computing the virtual time is $O(1)$.

The drawback of WFQ is inefficiency in calculating the timestamps and adding work complexity of $O(\log n)$, where task n is the number of flows. Additionally, WFQ is a sorted-priority scheduler that requires expensive hardware to implement with real-time applications.

Furthermore, due to the slow process of sorting the timestamps, WFQ will not be suitable with real-time applications such as VoIP applications [84].

2.5.4 Worst-case Fair Weighted Fair Queueing

J. Bennett et al. [81] introduced Worst-case Fair Weighted Fair Queueing (WF^2Q). It is better than WFQ and GPS schedulers and offers identical service to GPS [85]. WF^2Q allocates the start and end process time of the packet when choosing a suitable packet from the queue. WFQ and GPS are different from each other in terms of sending packets over an IP-based network. GPS sends traffic flow bit-by-bit and WFQ sends traffic flow packets-by-packets over the IP-based network. The main difference between the GPS fluid system and the packet-based system is that the fluid system can serve only one packet at the time of transmission. While, in the packet system, at any time multiple packets can be served. The traffic flow of Worst-case Fair Weighted Fair Queueing (WF^2Q) is illustrated in Figure 2.2.

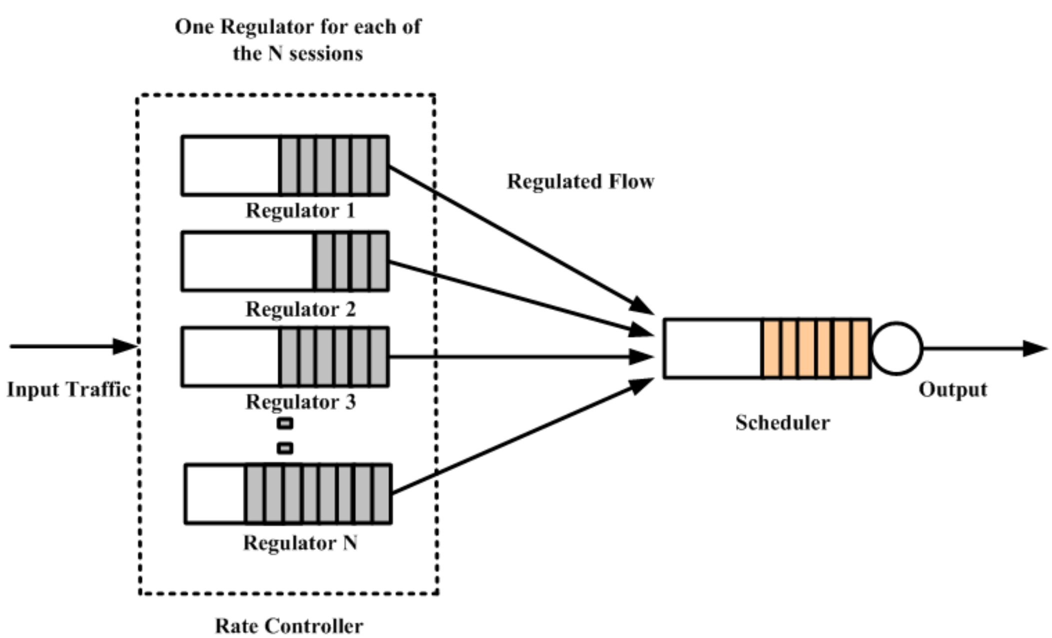


Figure 2.2 WF^2Q Traffic Flows [81]

Table 2.3 List of important WF^2Q Notations [81]

Notations	Description
N, W	Number of sessions, Work conserving
$\emptyset 1, \emptyset 2, \emptyset 3, \ldots \emptyset N$	Positive real numbers
r	The server operates at a fixed rate r
$\mathrm{W}_i\ (t_1,\ t_2\ \ )$	The amount of session traffic served in the interval
$[\ t_1, t_2\ \ ]$	The interval
i	Flow or session
$RC_{Rate\ Controller}$	Rate Controller
$R_{Regulator}$	Regulator
$S_{Scheduler}$	Scheduler
V_{GPS}	Virtual GPS
$c_{WF2Q} = \frac{\mathrm{L}}{r}$	The Packet completes transmission

List of notations are as shown above in Table 2.3. The GPS process is on real numbers, $\emptyset 1, \emptyset 2, \emptyset 3, \ldots \emptyset N$. The server operates the fixed size of rate as r, work conserving W and the amount of sessions number is denoted as i flow is sent in the interval $[\ t_1,\ t_1\ \ ]$. The GPS process is as shown in 2.3 [81]:

$$\frac{Wi\ (t_1,\ t_1)}{Wj\ (t_1,\ t_1\,)} \geq \frac{\emptyset j}{\emptyset j}\ \ j = \emptyset 1, \emptyset 2, \emptyset 3, \ldots \emptyset N \qquad (2.3)$$

GPS is a basic idea to manage multiple sessions on a single platform. To address the drawbacks of GPS, the suitable packet at time t, which is not greater than time t at the time of selection on the network, WF^2Q selects a packet that takes less time starting from the queue and moves forward to the ending point. WF^2Q is applied on wireless networks. WF^2Q is a combination of rate controller, regulator and scheduler; equations are as described in equation 2.4.

$$RC_{Rate\ Controller} = R_{Regulator} + S_{Scheduler} \qquad (2.4)$$

WF^2Q has a few drawbacks, e.g. the time complexity in applying the virtual time on high-speed networks but in WFQ applying the process on a virtual clock time V_{GPS}. WF^2Q needs another virtual clock time that is less difficult and better than V_{GPS}. WF^2Q has limited functions to fulfil the requirement of fairness index $c_{WF2Q} = \frac{L}{r}$ on high-speed networks. The analytical and simulation results show that WF^2Q has a worst case fairness index and queueing delay than WFQ and is not suitable for VoIP applications [86].

2.5.5 Deficit Round Robin Scheduler Scheme

Round Robin (RR) has many variations and modifications of scheduling algorithms. These all improve each other in fairness and efficacy, such as the Surplus Round Robin (SRR) [92], Nested Deficit Round Robin (NDRR) [93], Elastic Round Robin (ERR) [94], Dynamic Deficit Round Robin (DDRR) [95], Ordered Round-Robin (ORR) [96] and Airtime Deficit Round Robin (ADRR) [97].

Deficit Round Robin (DRR) was introduced by M. Shreedhar et al. [87] for active queue flows in the priority Round Robin (RR) formation [88], [89]. The DRR scheduler is classified as a frame-based packet scheduling algorithm for high-speed networks. DRR fulfil the short comings of the simple RR scheduler. Compared to the simple RR, DRR maintains a variable size of packets while RR maintains a constant size of packets.

For each queue, DRR offers a quantum size and deficit counter (DC). This counter counts the number of bytes of traffic that could be serviced in a current round. On the serviced time, DRR will insert a quantum size to the DC. The DRR weight is allocated by the quantum size of traffic flows in IP-based networks [90] and [91]

DRR and its modifications are not preferable for a real-time application such as VoIP because all of them follow the round robin technique. DRR and its modifications cause latency of packets which cannot provide short packet delay performance and better throughput, the delay bound is longer and depends on the bandwidth sharing rations. DRR and its modifications also focus on the fairness and cannot support the efficiency of performing the delay response of real-time VoIP quality.

2.5.6 Deficit Transmission Time

R. Garroppo et al. [98] proposed the Deficit Transmission Time (DTT) scheduling algorithm for only WLANs based on the IEEE 802.11 standard. DTT proposes to make sure that each node gets fairness on WLANs [99]. DTT supports Basic Service Set (BSS) infrastructure.

The DTT experiment is based on the test-bed traffic shaping scheme, wired host and Linux-based Access Point (AP) as centralized BSS communication with a number of nodes. These nodes change their locations and adjust to be near to the AP. The DTT evaluated with a classic First-In First-Out (FIFO) queue management technique.

The DTT scheduler offers the required traffic flow isolation for UDP and TCP traffic on high-speed networks [100]. DTT is implemented on two-way calculations; one way with nodes and another way with AP. Details are in Figure 2.3.

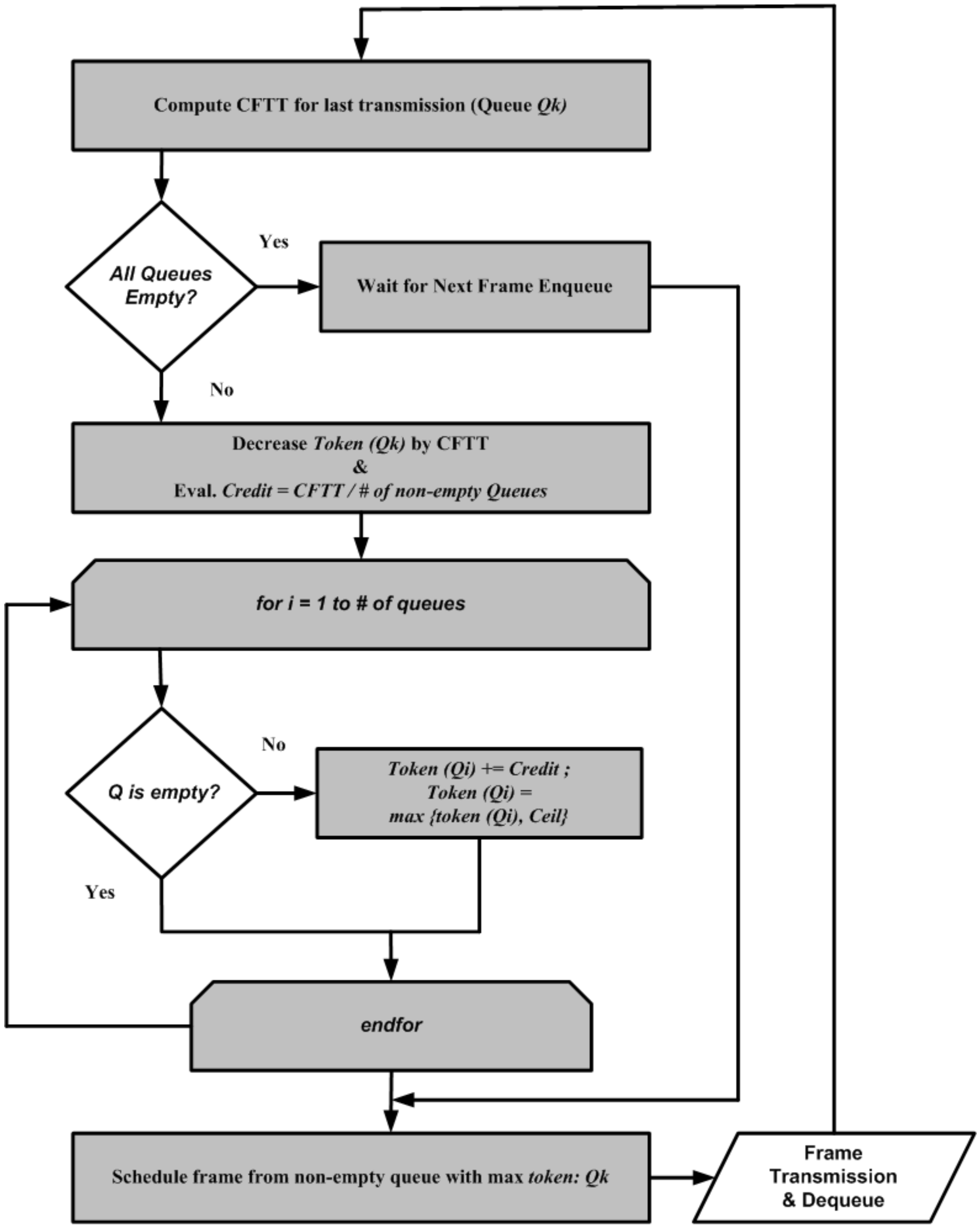


Figure 2.3 Flowchart Diagram of DTT Scheduler [98]

The first way obtains the Signal-to-Noise Ratio (SNR) values from the Wireless Network Interface Cards (NIC's) to evalue the maximum predicted throughput from each node. The second way calculates the overall amount of time immediately needed to send frame rate retransmissions. This type of calculation needs to provide Acknowledgement (ACK). This piece of information is used to compute the optimal schedule list.

The main drawback of DTT is that it is applied on both UDP and TCP traffic flows and it does not apply properly on VoIP traffic flow. Furthermore, it is comparable only to commercial AP such as IEEE 802.11 a/b standards. Finally, the DTT scheduler needs to improve various token sharing polices on high-speed networks.

2.5.7 LLEPS scheduling Algorithm

H. Wu et al. [54] proposed the Low Latency and Efficient Packet Scheduling (LLEPS) algorithm for real-time applications to offer bandwidth assurance service proficiently. It is introduced as a sort-based packet scheduling algorithm. Also, it assumes the long-term fairness and arranges the real-time stream of traffic well-ordered. LLEPS calculates the transmission rate of each task and ensures that each task can obtain the kept bandwidth. Every time, LLEPS manages the traffic queue with the highest priority and forwards packets for the queue.

LLEPS offers the pre-emption method for packets. Once a packet of a real-time stream like, VoIP is delayed for any particular reason, the LLEPS is able to forward the packet earlier than other packets. The frame-based packet scheduling does not have this mechanism [54].

2.5.7.1 LLEPS Queue Process

LLEPS manages multiple queues applied only for a traffic stream. LLEPS selects the highest priority packets from a queue and transmits the packets for that queue. Also, LLEPS introduced the history of traffic flow and it's based on the Start-Time Fair Queue (SFQ) [99]. The history of traffic flow indicates the deadlock in real-time traffic flow. LLEPS queue process is demonstrated in Figure 2.4.

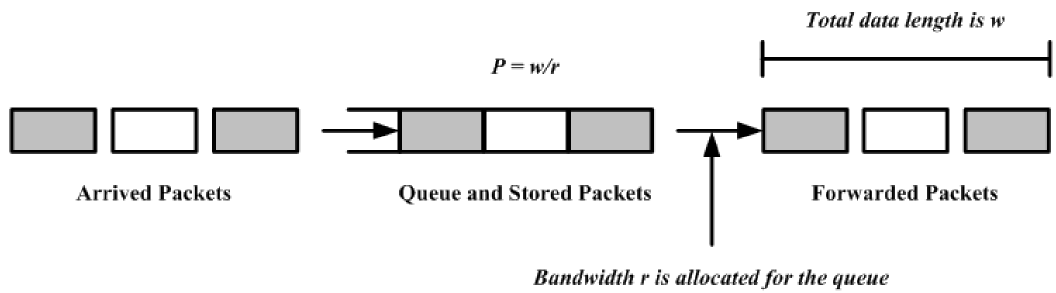


Figure 2.4 LLEPS queue process [101]

The arrived packets will be forwarded to the queue and stored. The forwarded packets are equal to the total data length which is w. LLEPS queues are associated with p, w, r and $flow$ as can be seen in the figure above. Let p_i be the queue priority of task i. Every time the p_i value will be bigger than the other queues. LLEPS provides the p_i value for each queue the same as for streaming transmissions on IP-based networks.

2.5.7.2 LLEPS Architecture and Components

LLEPS introduces a number of queues and forwards these queues with packets in the shape of a time segment. These queues show a time interval $(t_f - t_s)$ and occupy bandwidth for each task as described. LLEPS architecture and components are based on the description in Figure 2.5.

- Queues
- Min Heap
- System Timer
- Scheduler

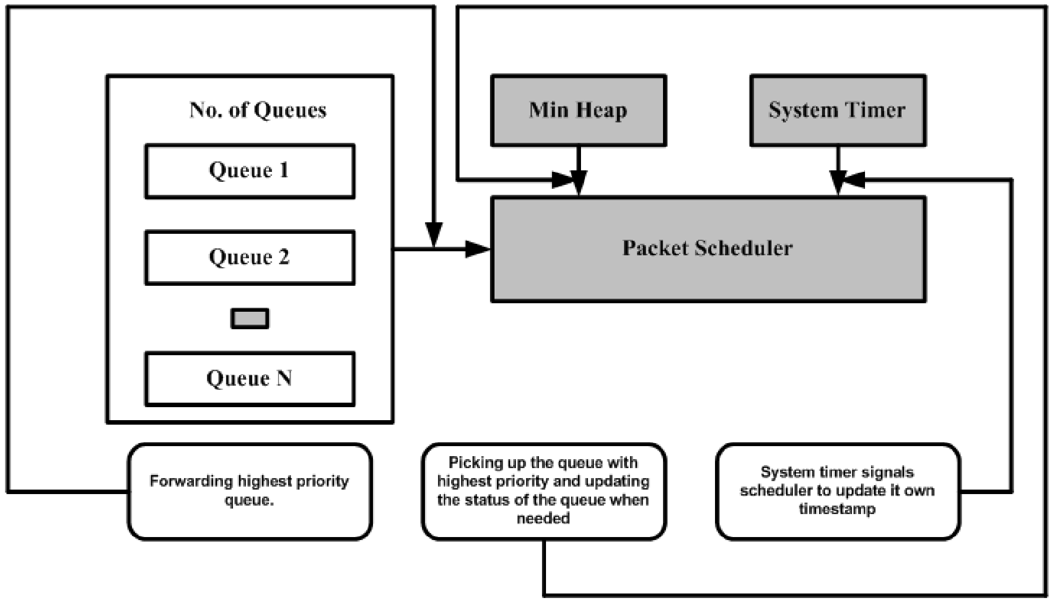


Figure 2.5 Architecture of LLEPS [101]

LLEPS has a number of queues in the architecture. Firstly, these queues will be sorted according to the higher priority queue. Secondly, the Min Heap (MH) function will be active to the manage the queue and provide the scheduler with a selection for the high priority queue. Thirdly, the System Timer (ST) upgrades the timer, after the selection of the high priority queue system.

Lastly, the scheduler keeps in touch with all three components. LLEPS is compared with two scheduling algorithms WFQ and NDRR. LLEPS has performed better than both algorithms where the queueing delay is in milliseconds (ms) [102] [103] and [104].

The LLEPS drawbacks are as follows: LLEPS does not provide a high-speed timer and it's not appropriate for high-speed and real-time applications. Furthermore, LLEPS calculates the transmission rate of each task in a time interval. If the time interval is very small, then the calculation will be become very inappropriate. This means LLEPS is not suitable for high-speed and real-time applications.

2.5.8 Credit Based-SCFQ Scheduler

E. Palacios et al. [105] presented a new Credit Based Self-Clocked Fair Queueing (SCFQ) which is a combination of SCFQ and Weighted Fair Queueing (WFQ) [106]. The main function of Credit Based SCFQ architecture is for arranging the bandwidth when the traffic mechanism refuses to queue more on the broadband wireless networks. Credit Based SCFQ provided hardware architecture and scheduler architecture [107], [108] and [109]. Credit Based SCFQ is an updated version of SCFQ which added two more parameters as follows:

- Wr_k = Wireless network weight to be allocated to the $K'th$ traffic flow (Weight allocation)

- C_k = Credit allocated to the $K'th$ traffic flow (number of packets add wireless weight)

The process of the Credit Based SCFQ Basic Service Set (BSS) is very simple and fair as after the packets are enqueued from the network, they are simply classified and dequeued to the destination. It is implemented in the MAC layer and provides priority, packet length and allocation to the traffic flow. Figure 2.6 shows the tag computations which assign the smallest tag worth to the biggest fair queue.

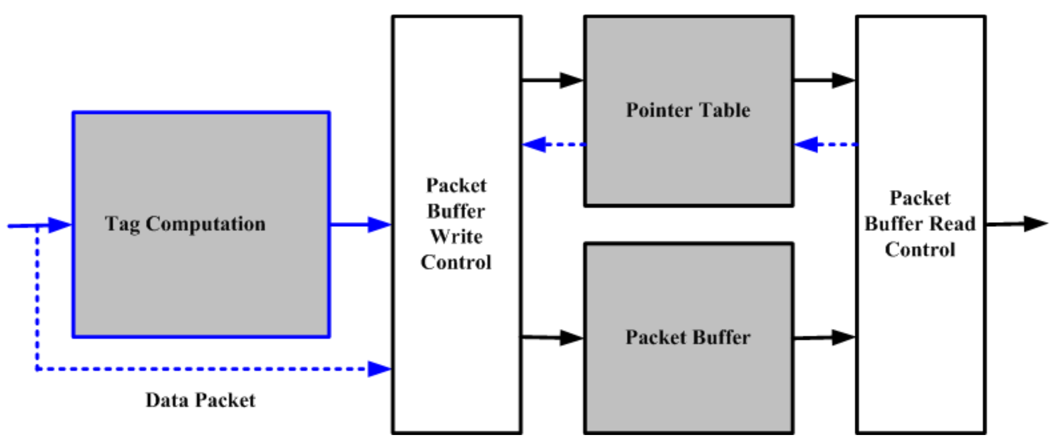


Figure 2.6 Credit Based-SCFQ Scheduler Architecture [105]

The main function of packet buffer write control is for temporarily storing the upcoming packets. Content addressable memory manages the finished tag and packet pointer values. Again, the packet buffer stores the packets before sending them to the destination. These packets arrive to the packets buffer read control and then dequeue to the destination.

The drawback of the Credit Based-SCFQ scheduler is that, it is designed in the hardware with traffic flows. Credit Based-SCFQ without specific hardware cannot be implemented on the network and that hardware increases its cost for implementation on the network. Another drawback of Credit Based SCFQ is that it only works based on the IEEE 802.16 standard and proposed research based on the IEEE 802.11 standard.

2.5.9 Proportional Fairness Packet Scheduler

E. Lee et al. [110] introduced the packet scheduler Proportional Fairness (PF) algorithm over an IP-based network. The PF is specifically applied for the communication schemes and explains the reservation based function. The architecture of the PF packet scheduler for the traffic flow of VoIP and the traffic flow of data is shown in Figure 2.7.

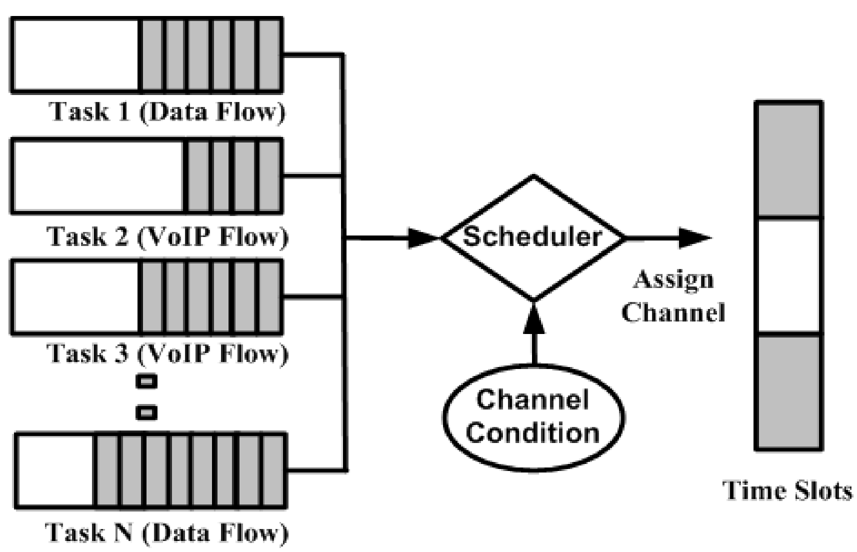


Figure 2.7 Architecture of VoIP and Data Traffic Flow [110]

This algorithm is purely for VoIP traffic flow and considers the time delay with the channel condition. It provides the highest priority to time delay traffic on networks. The scheduler works as following: The scheduler chooses the highest priority from the flow. The average transmission data rate of the user is calculated based on the following equation 2.5 [110]:

$$j = arg\ \ MA_iX\ \left(\frac{C_i\ (t)}{R_i\ (t)} \right) \qquad (2.5)$$

Where $j =$ arg and $C_i\ (t)$ is the condition of the channel respectively the user as $i,\ t$ as time and the $R_i\ (t)\ arg$ transmission data rate of users over the IP-based network. PF has Reservation Based (RB) and Integrated Packet Scheduling Algorithms. They introduced an algorithm, which is applied on voice and data packets traffic. This algorithm provides the highest priority to VoIP traffic.

The weight of the queue decreased and increased based on the weight of the delay. In the experimental and simulations setup the proposed scheduling algorithm is with PF and RB. They were compared with the voice and data traffic channels from 1 to 5. The applied parameters are the number of nodes, throughput (bps), drop probability (%), and priority. The proposed algorithm was approximately better than both PF and RB.

The drawback of this scheduling algorithm is that the PF cannot classify VoIP and data traffic. It is not properly compared with the PF and RB algorithms. Furthermore, it has the main drawback of priority-based functions. If real-time traffic flow is increased or bursty then a problem will arise for other low priority traffic flows.

2.5.10 Application-aware Scheduling for VoIP

N. Bayer et al. [111] introduced an application-aware Scheduling scheme for VoIP traffic over the 802.11-based standard. This type of scheduling technique is applied on TDMA-based access control networks. This scheduling technique also provides a new VoIP-aware resource coordination method and is validated through the NS-2 simulation. This coordination method is scalable and provides a better quality for VoIP traffic flow over a wide range of networks. This approach provides on-demand and continuous mechanisms. In the scheduler, in order to reserve the bandwidth for the data transmission between two nodes, a three-way handshake (TWHS) mechanism is applied. The list of notations is as shown in Table 2.4.

Table 2.4 List of important Abbreviations of Scheduling Scheme [111]

Notations	Description
TWHS	In order to reserve the bandwidth for the data transmission between two nodes.
REQ	The TWHS by sending request contained within the mesh nodes to transmit distributed scheduling signalling request.
AVA	Availability of node
GNT	Bandwidth requirement and will reply with a grant.
CONF	It is simply a copy of the GNT.

The architecture of the scheduler in Figure 2.8 shows the process of the scheduler and all processes implemented on the base station. This scheduler has three ways of communication, one-hop node A, one-hop node B and extended two-hop node A and B. This scheduling scheme becomes three-way handshaking due to three ways of node communication.

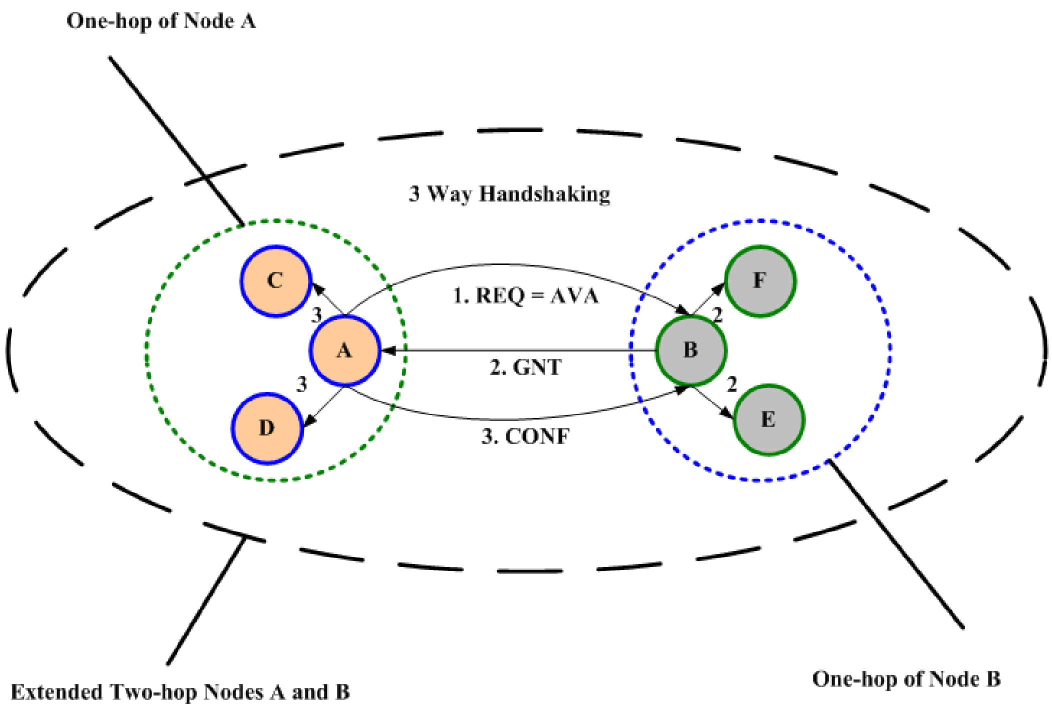


Figure 2.8 Architecture of Application-aware Scheduling [111]

In this scheduler, the three-way handshaking technique is introduced and reserves the bandwidth for the data transmission between nodes. The findings of the application-aware scheduling are in Table 2.5.

Table 2.5 The Findings of the Application-aware Scheduling

Term	Description
On-demand	On-demand traffic flow of VoIP traffic as applied in WLAN.
Continuous Scheduling	Application-aware scheduling is well suitable to serve VoIP traffic.
Optimised	This scheduling scheme provides carrier-grade services and high voice quality. Independent number of hops between the nodes and the base station.
Application-aware	As applied by the VoIP aware resource for coordination scheme and is manageable to increase scalability over the network.

The drawbacks of the application-aware scheduling scheme is that, this scheduling scheme is based on the IEEE 802.16-2004 standards, Wireless Mesh Networks (WMN), and it does not support the IEEE 802.11a/b WLAN. It was applied on chain and grid topologies on the NS-2 simulation.

2.5.11 Controlled Access Phase Scheduling

The IEEE 802.11-based WLAN requires enough bandwidth, higher throughput and less delay for high-speed applications. The 802.11e provides two types of Quality of Service (QoS) approaches, contention-based and controlled-based. Y. Fallah et al. [112], [113] proposed the Controlled Access Phase Scheduling (CAPS) algorithm that adopts a controlled-based approach for implementation. The IEEE 802.11-based WLAN is introduced for distributed networks but Medium Access Control (MAC) facilities the Access Point (AP) centrally located for access in both ad-hoc and an infrastructure model. The CAPS is implemented on a centralized access method.

The CAPS algorithm is maintained on Virtual Packet (VP) and mixed-mode scheduling of uplink and downlink traffic flows in the AP. CAPS is applied on upstream and downstream traffic flows and reserves the bandwidth during the process of the task. The algorithm allocates the remaining capacity to be shared with all nodes and flow in a contention method and is based on Enhanced Distributed Channel Access (EDCA). Furthermore, CAPS introduces a queueing structure and a new scheduling architecture. This structure is known as Multiple Access Hybrid Scheduling (MAHS) [114]. The CAPS algorithm is briefly discussed in the following sub-sections:

2.5.11.1 Mixed Downstream / Upstream Scheduling

The CAPS concept is to describe packets from the distant nodes e.g. the downstream flow. The downlink cannot create a schedule but the AP can create a schedule from the Stations (STAs) and concerns the polling messages to STAs at the times indicated by the schedule. This scheduler also represents packets from the remote STAs (e.g. the upstream packets) through virtual small packets in the AP. The scheduler then schedules virtual packets along with actual packets (downstream packets). This concept is presented in Figure 2.9.

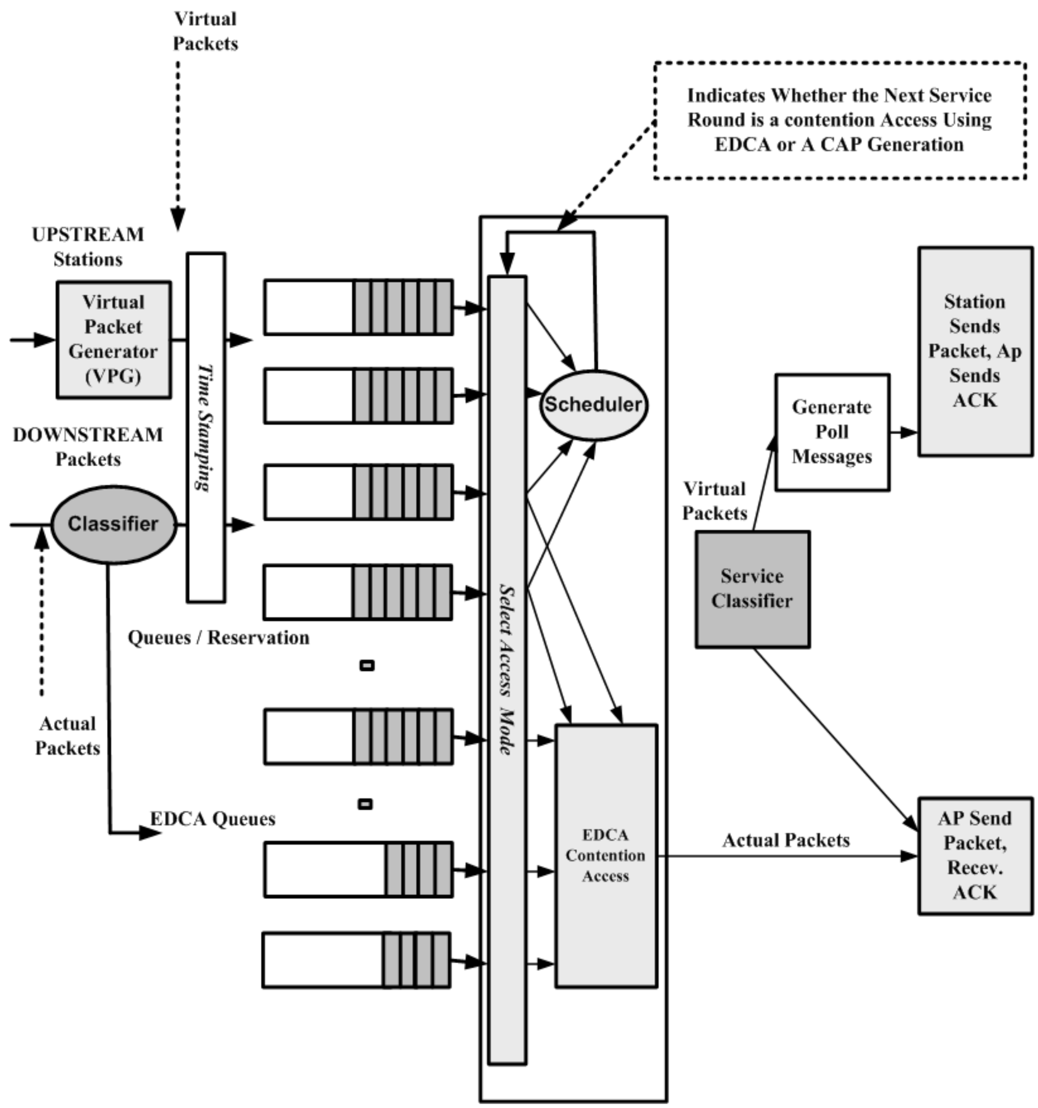


Figure 2.9 Architecture and queueing model of CAPS [112], [113]

Furthermore, Y. Fallah et al. [112], [113] applied Weighted Fair Queueing (WFQ) and an algorithm to enhance the capacity of the voice traffic over WLANs. This process is called Hybrid Scheduling (HS) because CAPS combines the upstream packets and downstream packets on a single platform for real-time applications.

2.5.11.2 Queueing, Scheduling, and Traffic Shaping

CAPS is basically an integrated scheduler, it contains a scheduler, shaper, EDCA and HCF Controlled Channel Access (HCCA) operation to realize both fairness and service guarantee over IP-based WLANs. Furthermore, the process of the CAPS algorithm can be divided into three steps; each run in a separate process. The first step is responsible of admission control and generating virtual packets (VP) according to the confirmed task information. The second step is responsible of time-stamping and queueing the arriving packets in equation 2.6 [112].

$$F_i^k = \frac{L_i^k}{r_i} + \max\left(F_i^{k-1}, V(t)\right) \qquad (2.6)$$

Where F_i^k is the timestamp for the $K\ th$ and packet from the $i\ th$ traffic flow r_i is the rate, L_i^k is the adjusted packet length, $V(t)$ is the virtual time function and shows the process time of the Generalized Processor Sharing (GPS) Scheduling mechanism. The third and main step is choosing the packet to be provided and controls the switching between HCCA and EDCA access [112].

Y. Fallah et al. [114] also introduced algorithms and one of them is the Multiple Access Hybrid Scheduling (MAHS) algorithm. The MAHS applies the Virtual Packet (VP) that is very similar to the CAPS algorithm. Additionally, it arranges an upgraded adaptation of Weighted Fair Queueing (WFQ). Y. Fallah et al. [115] introduced a few more: Distributed Adaptively Configured EDCA (DACE-T) and Fair HCCA Priority Scheduling (FAHPS). DACE-T and FAHPS are applied on WLAN 802.11e standards and based on the results it was found that both algorithms perform better than the Enhanced Distributed Channel Access (EDCA) method.

The drawbacks of CAPS, MAHS and other algorithms are as follows: CAPS is a very complicated scheduler as it includes a Virtual Packet Generator (VPG), multiple queues with EDCA and the HCCA access method. It also has an internal scheduler and many more that provide some complication and cause the delay in real-time application. It is implemented only on IEEE 802.11e and IEEE 802.16; however, they are working on WLAN with IEEE 802.11a/b standards.

2.5.12 Efficient Fair Scheduling

S. Ahmed [116] introduced Efficient Fair Scheduling (EFS) which is a packet-based traffic scheduling algorithm. It provides an end-to-end quality of Service (QoS) and it is evaluated by an analytical and simulation environment. This algorithm is based on the Deficit Round Robin (DRR) scheduler. It contains classification, token bucket as well as enqueue and dequeue processes.

However, the drawback of the efficient fair scheduling algorithm is that it is for a wired network only. Therefore, it's not suitable for the WLAN environment. Furthermore, it's based on the DRR scheduler.

2.5.13 Queue size Prediction-Computation of Additional Transmission

S. Shin et al. [117] proposed the Queue size Prediction-Computation of Additional Transmission (QP-CAT), which is a packet-based traffic scheduling algorithm for a VoIP network. QP-CAT is a good queue size calculation technique in an AP over WLANs with IEEE 802.11 standards [118], [119].

QP-CAT manages the overflow of VoIP calls over IP-based networks. It applies the Call Admission Control (CAC) technique to improve QoS over WLANs. QP-CAT coordinates between the AP and a number of nodes in a Basic Service Set (BSS). They reduced wasted bandwidth and evaluated with a simulation tool (Qualnet). In addition, they verified QP-CAT in the MadWifi driver and validated the functioning on the WLAN test-bed called Open Access Research Test-Bed for Next-Generation Wireless Networks (ORBIT).

The ORBIT is applied to recognize the correlation between the same queue flow size of an AP and downlink delay on a number of nodes in a BSS. In addition, the correlation between the AP and the number of nodes in the BSS by analysis is validated. The list of WFQ notations is as shown in Table 2.6.

Table 2.6 List of important QP-CAT Notations

Notations	Description
(D)	Downlink delay
(D_Q)	Queueing delay
(D_T)	Transmission delay
T_c	Time Checking
T_r	Remaining time
n_p	Transmitted frames
Q_p and Q_A	Actual queue size of the uplink

Where downlink delay is (D), queueing delay (D_Q), and transmission delay (D_T).

Subsequently, $D = D_Q + D_T$

Where, $Queue\ Delay\ Calculate = D_Q * D_T$

Also, $Queue\ Size\ (Q) = D_{System} = Q_{system}\ / \mu_{\ system}$

Furthermore, the calculations of the queueing delay of the AP are performed based on the equation 2.7 and 2.8.

$$D_Q = Q.\ \frac{1}{\mu_{\ AP}} = Q.\ D_T \qquad (2.7)$$

Where, the delay of the N of nodes in a BSS delay applies the equation:

$$D = Q.D_T + D_T = (Q + 1).D_T,\ (2.8)$$

The QP-CAT calculates the upcoming queue size of the AP applying the emulation of the upcoming VoIP traffic flow. Also an explanation of the voice flows is shown in Figure 2.10.

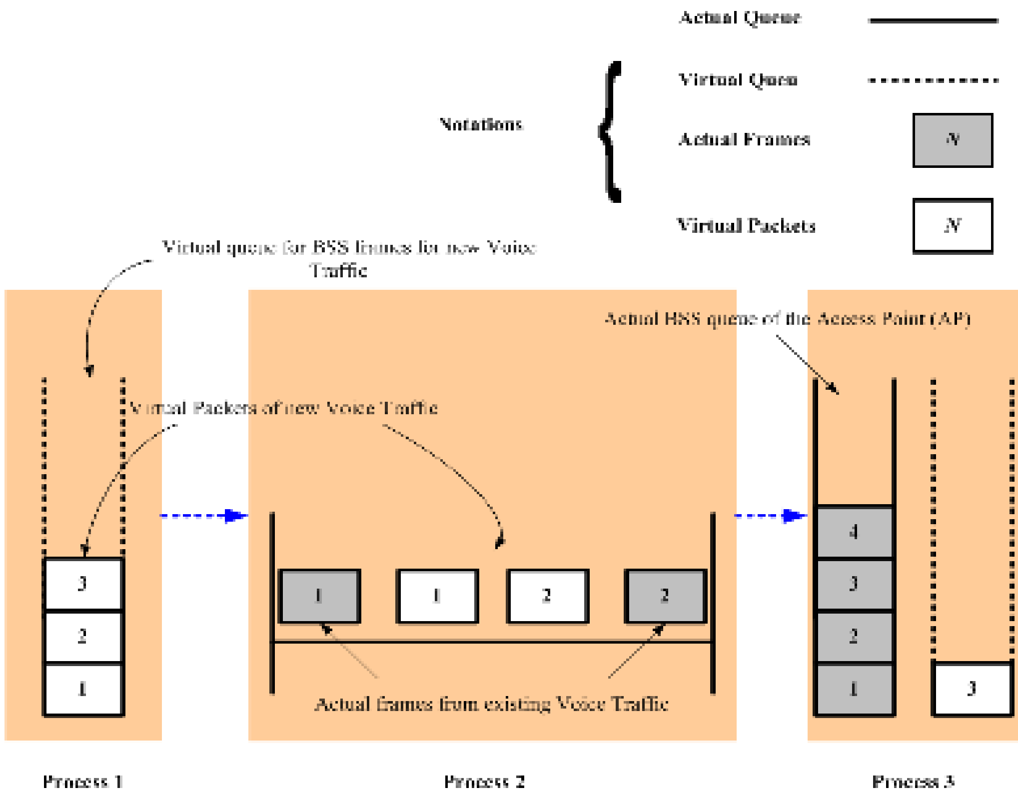


Figure 2.10 Actual and Virtual queue flow of QP-CAT [117]

Figure 2.11, the counter for a packet of upcoming voice traffic over WLANs is presented. In addition, the QP-CAT is described in the following algorithm.

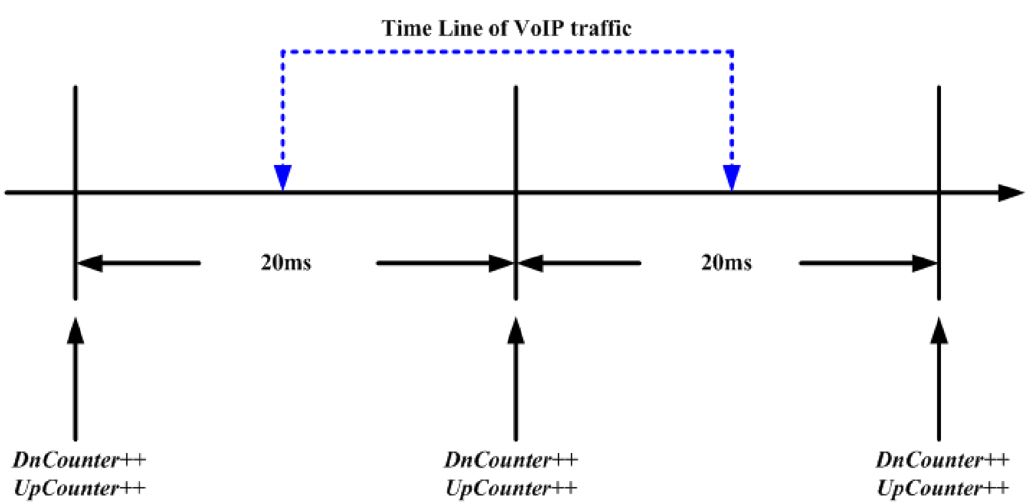


Figure 2.11 Emulation of VoIP traffic 20ms time interval [117]

In Figure 2.12, let T_c be time checking, T_r be the remaining time from an earlier calculation, n_p be transmitted frames and $DnCounter$ utilizing n_p, the upcoming queue size Q_p and Q_A be the actual queue size of the uplink and then they are calculated.

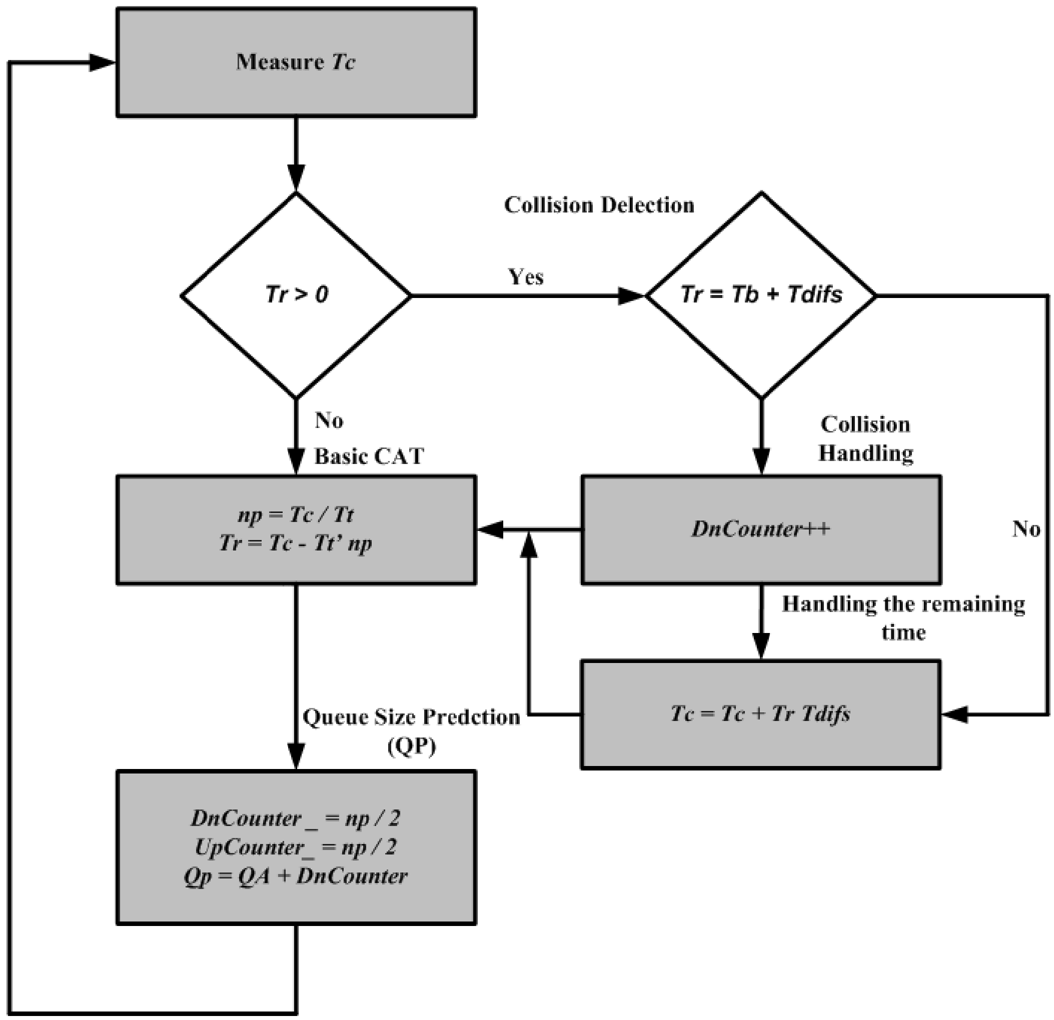


Figure 2.12 Flowchart of QP-CAT Algorithm [117]

The drawback of PQ-CAT is that it's only applied on a MadWifi wireless card driver on an Open Access Research Test-Bed for Next-Generation Wireless Networks (ORBIT). In addition, it's very complicated due to its evaluation on the QualNet simulator 3.9, MadWifi wireless card, ORBIT test-bed and emulations on real-time traffic. Furthermore, QP-CAT does not fully support WLAN standards and therefore, there is a need for enhancement of QP-CAT.

2.5.14 Contention-Aware Temporally fair Scheduling

Y. Seok et al. introduced [120] a packet based algorithm named Contention-Aware Temporally fair Scheduling (CATS). CATS offers a fairness traffic flow over WLAN with IEEE 802.11a/b standards. CATS introduced equal time sharing for each flow over IP-based networks. CATS decides the packet scheduling order after the virtual finish time. In addition, the scheduler is capable of performing in multi-rate [121], [122] WLAN environments. CATS provides a solution for the Carrier Sense Multiple Access / Collision Avidness (CSMA/CA) technique. The list of WFQ notations are as shown in Table 2.7.

Table 2.7 List of CATS important Notations [120]

Notations	Description
$F(i,k) = S(i,k)$	A virtual finish time the normalized amount of service
$\frac{P(i,k)}{\emptyset(i).C(i)}$	The packet size for the flow Kth packet of flow i , and $\emptyset\,(i)$ is its weight
$C\,O\,(t)$	the contention overheard
$C\,O\,(t)'$	the exponential weight
$currentACK$	Mobile station updates current ACK variable
$lastACK$	Mobile station updates last ACK variable

CATS is applied on the updated version of the Weighted Fair Queue (WFQ) and the Temporally-Weighted Fair Queueing (T-WFQ) algorithms. CATS defines two algorithm; the first algorithm is for the enqueue technique and the second one is for the dequeue technique. The CATS algorithms are presented in Figures 2.13 and Figure 2.14.

Enqueue Technique
1. **if** $Qi^* = \emptyset$ **then** 2. $Qi^{**} \leftarrow P^{***}$; 3. $S_i \leftarrow max\ (f_{i,}\ R\ (t))$; 4. $f_i \leftarrow S_i + L / (C\ (i) \times \emptyset i) + CO\ (t) / \emptyset i$; 5. $R\ (t) \leftarrow max\ (min_{k \epsilon B\ (t)}\ S_k\ , R\ (t))$ 6. **end i** 7. insert $(P,\ Q_i\ )$; 8. **return;**
Algorithm Description
*i: *Flow id,* ***Qi: Queue for flow i* **** P: Newly arrive packets,*

Figure 2.13 CATS algorithm for Enqueue technique [120]

Dequeue Technique

1. $i* \leftarrow \min_{k \in B\ (t)\ and\ sk\ \leq R\ (t)} f_k$;
2. delet from (P, Qi)
3. nextPacket $\leftarrow$ Qi;
4. **if** $nexPacket\ \neq\ \emptyset$ **then**
5. $S_i\ \leftarrow max(fi, R\ (t))$;
6. $fi\ \leftarrow\ S_i +\ L\ /\ (C\ (i)\ \times \emptyset i) + CO\ (t)\ /\ \emptyset i$;
7. **end if**
8. $R\ (t) \leftarrow$ max $(min_{k \in B\ (t)\ \ sk,}\ R\ (t)$
   $+\ L\ /\ \sum_{k \in B\ (t)}^{\cdot}(C\ (K)\ \ \times\ \emptyset k) + CO\ (t) **/\ \ \sum_{k \in B\ (t)}^{\cdot} \emptyset k)$;
9. *transmit* **P*****;*

Algorithm Description

**i: Flow id,*

** $C\ O\ (\ t\ )$: *is the contention overheard,*

*** P: Newly arrived packets,*

Figure 2.14 CATS algorithm for Dequeue technique [120]

The basic equation 2.9 is as follows [120], [121]:

Where,

$$F\ (i\ ,k\ ) = S(i\ ,k)\ +\ \frac{P\ (i\ ,k\ )}{\emptyset\ (i\ ).C\ (i\ )}\ +\ \frac{C\ O\ (t\ )'}{\emptyset\ (i\ )} \qquad (2.9)$$

Let $C\ O\ (t\ )$ be the contention overheard, and $C\ O\ (t\ )'$ is the exponential weight, where the time-slot is as below:

$$time - slot\ state = successful\ access, collision\ \&\ backoff$$

Furthermore, the contention overhead $C\ O\ (t\ )$ of each node is calculated in equation 2.10 as shown below:

$$C\ O\ (t\ ) = \ currentACK - lastACK \qquad (2.10)$$

$$-PLCP\ LENGHT\ field$$

The CATS algorithm is briefly described in Figure 2.15.

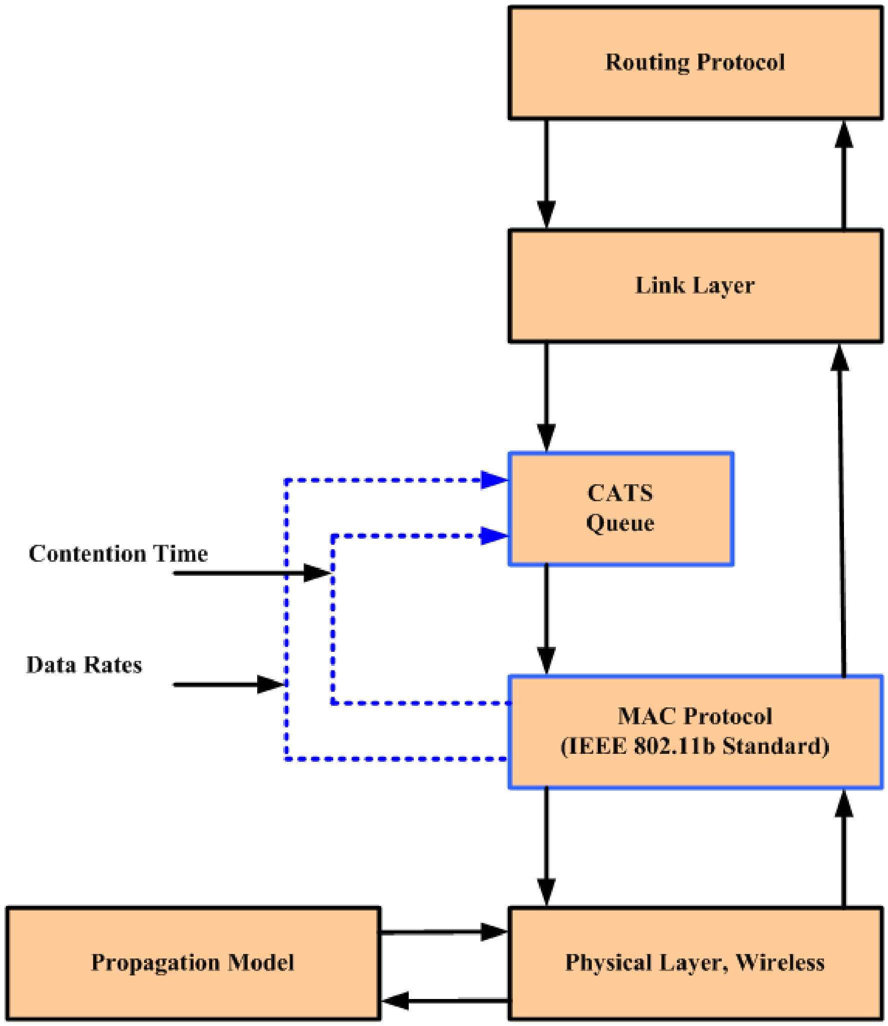


Figure 2.15 CATS Protocol Stack [121]

In addition, CATS is also applied in an upgraded version known as Decentralized Contention-Aware Temporally fair Scheduling (D-CATS). It's developed for WLAN with IEEE 802.11 Distributed Coordination Function (DCF) [123] and provides information of each node. D-CATS consider the uplink traffic flow. The D-CATS scheduling algorithm manages the WLAN without modifying the Medium Access Control (MAC).

The drawbacks of CATS is that it is based on Generalized Processor Sharing (GPS) that is applied on wired based networks and it is based on a fluid flow mechanism. GPS is based on a fixed link of capacity and cannot facilitate the per-flow process. CATS applies T-GPS and cannot provide each flow calculation on the WLAN. Meanwhile, the Temporally-Weighted Fair Queueing (T-WFQ) that is based on GPS has a lower performance than CATS.

Another drawback of CATS and its upgraded version D-CATS is that if the backoff method is not applied they need to be altered using the 802.11 MAC protocol. Due to this problem, S. Seok et al. introduced another CATS version D-CAT+ that added more complexity to schedule voice traffic and did support multi-hop over WLANs.

In order to address these significant issues over WLANs for voice traffic, a new Voice Priority Queue (VPQ) scheduling algorithm is proposed. In the next chapter, the proposed scheduling algorithm is discussed in detail and compared with the above mentioned scheduling techniques. A summary of the related schedulers are as shown in Table 2.8.

Table 2.8 Summary of the related schedulers

Scheduler Names	Researched by with Year	Advantages	Disadvantages
CBQ	S. Floyd et al. 1998	CBQ provides solutions for data traffic, link-sharing and resource management mechanisms for packet networks.	Not suitable for real-time and VoIP applications due to bursty traffic misconduct The bulky buffer creates a bigger delay time and it's overloaded for real-time applications.
FQ	A. Demers et al. and S. Keshav 1989, 1991	FQ is presented in both ways the theoretical and practical. FQ sends transmission packet-by-packet.	FQ experience with high time-complexity. FQ only preceded the costs metric and it will not be suitable for VoIP applications.
WFQ	A. Demers et al 1989	WFQ is a sort-based packet scheduling algorithm. It is an updated scheduler of Generalized Processor Sharing (GPS).	WFQ is inefficient to calculate the timestamps and slows the process of sorting among the timestamps. It is expensive to implement on real-time applications.

Table 2.8 Continue

Scheduler Names	Researched by with Year	Advantages	Disadvantages
WF^2Q	J. Bennett et al. 1996	WF^2Q fulfils the drawbacks of the GPS. It is better than WFQ and GPS schedulers.	WF^2Q has time complexity and limited functions for the requirements of the fairness index.
DRR	M. Shreedhar et al. 1995	DRR is classified as a frame based packet scheduling algorithm. DRR fulfils the short-comings of the simple round robin (RR) scheduler.	The delay bound is longer and depends on the bandwidth sharing ration. DRR focuses on the fairness and cannot support the efficiency of the scheduler.
DTT	R. Garroppo et al. 2007	DTT is a scheduling algorithm introduced for WLANs. DTT supports an infrastructure BBS method.	DTT is applied on both UDP and TCP traffic flows but it was not applied on the VoIP traffic flow. DTT needs to improve token sharing polices.

Table 2.8 Continue

Scheduler Names	Researched by with Year	Advantages	Disadvantages
LLEPS	H. Wu et al. 2006	LLEPS algorithm offers bandwidth assurance service. Assumes the long-term fairness and makes the real-time stream of the traffic in an acceptable manner.	LLEPS did not provide a high-speed timer. It is not appropriate for high-speed applications like VoIP traffic Flow over WLANs.
CB-SCFQ	E. Palacios et al. 2004	Credit Based SCFQ updated scheduler of Self-Clocked Fair Queueing (SCFQ) combined with WFQ. It rearranges bandwidth, provides hardware and scheduler architecture.	CB-SCFQ is based on hardware and this hardware makes it cost expensive. Another drawback is that it is only implemented on the IEEE 802.16 standard.
PF	E. Lee et al. 2009	Proportional Fairness (PF) algorithm is applied on voice communication schemes. It is also implemented on VoIP traffic.	It cannot classify the voice and data traffic. It does not properly compare with PF and RB algorithms. On bursty traffic flow, it only focuses on high priority and low priority getting delayed over networks.

Table 2.8 Continue

Scheduler Names	Researched by with Year	Advantages	Disadvantages
App-Aware Scheduling	N. Bayer et al. 2010	Application-aware scheduling scheme for VoIP traffic. It provides a VoIP-aware resource coordination method.	It is applied on TDMA-based access control over networks. It is based on IEEE 802.16-2004 standards and implemented only on Wireless Mesh Networks (WNM).
CAPS	Y. Fallah, 2007	CAPS is an adopted controlled-based approach for implementation. It is based on virtual packet and mixed scheduling of uplink and downlink. It provides distributed traffic flow inside the Access Point (AP). It provides traffic flow for both ad-hoc and infrastructure models.	The disadvantage of CAPS is that it is very complicated. It includes virtual and actual packets, Virtual Packet Generator (VPG), multiple queues with EDCA and HCCA access methods. It has schedulers inside a scheduler and becomes complicated over networks.

Table 2.8 Continue

Scheduler Names	Researched by with Year	Advantages	Disadvantages
QP-CAT	S. Shin et al. 2008	QP-CAT is applying a queue size calculation technique in an Access Point (AP). It is based on the Call admission Control (CAC) technique. It manages the overflow of VoIP calls over networks. It is simulated on the Qualnet simulation tool.	QP-CAT is only applied on the MadWifi wireless card driver on ORBIT. It needs enhancement of QP-CAT. It needs an enhancement to fulfil the VoIP scheduling requirement over WLANs.
CATS	Y. Seok et al. 2007	CATS is a packet based algorithm and offers fairness traffic flow over WLANs. It supports IEEE 802.11 a/b standards. It is based on T-WFQ and has a primary version of CATS. It has upgraded versions of Decentralized-CATS, D-CATS+ etc.	Drawbacks of CATS are as follows: CATS is based on GPS scheduler that was never applied practically and GPS is proposed for wired networks. CATS is applied on T-GPS and cannot provide each flow calculation on WLANs.

2.6 Summary

This chapter discussed related traffic schedulers and algorithms. As well as, they discussed fundamental and basic characteristics of each scheduling technique [124], [125], [126], [127], [128], [129] and [130]. In particular the classifications of scheduling algorithms of real-time traffic schedulers, Voice over IP (VoIP) traffic scheduler issues and representative schedulers. Some of the related scheduling algorithms are introduced to improve QoS over WLANs. These scheduling schemes improved the performance of IP-based traffic, increased throughput, increased the number of mobile nodes, and the efficiency on high-speed networks but with limitations over the network.

Furthermore, these scheduling algorithms have some limitations, as discussed above in this chapter. This chapter provided a foundation for the new scheduling algorithm for serving the VoIP over a time shared WLAN packet-based scheduling algorithm. A new Voice Priority Queue (VPQ) scheduling technique is proposed and will be presented in the following chapter to improve the VoIP scheduling technique over WLANs.

CHAPTER 3

VOICE PRIORITY QUEUE (VPQ) SCHEDULER

3.1 Introduction

The scheduling system model plays a major role in the Voice over IP (VoIP) over Wireless LANs (WLANs). It fulfills the Quality of Service (QoS) requirements of the VoIP over WLAN through the scheduler, efficient algorithms and managing traffic flow. The VoIP over WLAN is another emerging application besides the Internet Protocol TV (IPTV) and the High Performance-Video Conferencing (HP-VC). This chapter presents a new VoIP scheduling system model and algorithm in order to enhance performance of voice traffic over WLANs.

The scheduling system model is an important technique to achieve efficient throughput and fairness over WLANs based on IEEE 802.11 standards [131], [132]. Scheduling techniques manage voice traffic over WLANs. It will be able to offer bandwidth link-sharing to tolerate the status of changing traffic queues and to be scalable over IP-based networks. As discussed in chapter two, a number of related schedulers have been proposed to support traffic flow over IP-based networks. Most of the existing schedulers support limited services and do not meet the requirements of real-time applications especially for the VoIP over WLANs.

The bottleneck of the scheduling mechanism is the downlink path and uplink path of traffic flow over WLANs. In the scheduling mechanism, the downlink path is comparatively easy due to the Access Point (AP) having complete information regarding traffic flow. As far as the uplink path is concerned, traffic flow comes from Stations (STA) which makes it difficult due to diverse techniques, procedures, parameters and traffic flow over networks [133].

3.2 Proposed VoIP Scheduler

A number of traffic scheduling system models has been introduced to enhance traffic flow over WLANs. Since in the WLAN, the VoIP Flow (VF) and Non-VoIP Flow (NVF) traffic flows are sharing the same transmission media, therefore, there must be a traffic scheduling system model to differentiate between the flows so that they can be successfully transmitted to the proper destination.

In order to achieve QoS capability, the standard describes a number of parameters like traffic flow techniques, bandwidth request technique and channelizing for traffic flow. As studied, these standards have limited QoS skills over WLANs. As have come to realize, utilizing a traffic scheduling algorithm is a way to enhance performance of the VoIP over WLANs; therefore, a new scheduling algorithm is necessary to improve the VoIP flow over networks [134].

When multiple traffic flows are sharing a common transmission link, there must be a traffic scheduler to make a decision and be accountable for improving the traffic source share to the individual traffic flows [135].

New VoIP traffic schedulers propose for WLANs that provides the traffic flow with intelligence on WLANs. The proposed scheduler assigns priority to the VF traffic in order to meet the performance requirements. Furthermore, the scheduler provides a traffic flow performance technique to allocate the Priority Queue (PQ) for VF traffic over IP-based networks.

In this chapter, propose a new Voice Priority Queue (VPQ) scheduling system model and algorithms. The different traffic flows arrive at the Access Point (AP) and combine with the VF and NVF flows over WLANs. Furthermore, the VPQ classifies the VF and NVF traffic flows for specific proposes for the VoIP traffic flow over WLANs using IEEE 802.11 standards. The proposed VPQ system model is unique as compared to most of the existing schedulers. The details of the system model will be presented in the following section.

3.3. Voice Priority Queue System Model

The designing of the scheduling system model is challenging due to the diverse QoS requirements such as throughput and fairness. VoIP traffic scheduling is the sharing of a transmission link among a number of wired and wireless mobile nodes over WLANs using IEEE 802.11 standards. However, because of variation and contention in the set of different traffic flows in each group, it cannot provide maximum fairness guarantees.

A methodology present for the VoIP traffic scheduler and algorithm for WLANs using IEEE 802.11 standards and apply this methodology to derive a new algorithm based on classification and Priority Queue (PQ) management.

In this chapter, the system model considered in research. New VoIP traffic scheduling system model, scheduler and algorithms: the Voice VPQ scheduler for IP-based networks. The VPQ provides bidirectional voice traffic communication over uplink and downlink connections. It can make an end-to-end guarantee of delay to a session of the best-effort traffic and classify the traffic flow into the VF and NVF.

The VPQ has pre-packet delay bounds and provides both bounded delay and fairness over WLANs using IEEE 802.11 standards. The VPQ concurrently provides both throughput and fairness. It also provides throughput guarantees for error-free flows, long term fairness for error-free flows.

The VPQ provides Fair Queueing (FQ) in the VF and NVF over IP-based networks. The main rule of FQ is the output bandwidth fair-sharing among multiple queues over WLANs. Further progress has been fulfilled towards noticing a low delay explanation by appropriating the VPQ system model over WLANs. In the VPQ, it differentiate the VF packets based on packet size.

3.3.1 First Stage of VPQ System Model

The first stage of the Voice Priority Queue (VPQ) system model propose and initial operations. The VPQ is based on some special components, initialization of traffic, classification of enqueue traffic flow, VoIP Flow (VF) and Non-VoIP Flow (NVF), Traffic Shaping (TS) and Token Bucket (TB), Voice Priority Queue (VPQ) Scheduler, VoIP Flow Buffer (VFB), Non-VoIP Flow Buffer (Non-VFB) and dequeue traffic flow for the end user over WLANs using IEEE 802.11 standards. It will describe in detail these components in this chapter. Figure 3.1 articulates the working of the VPQ scheduling system model.

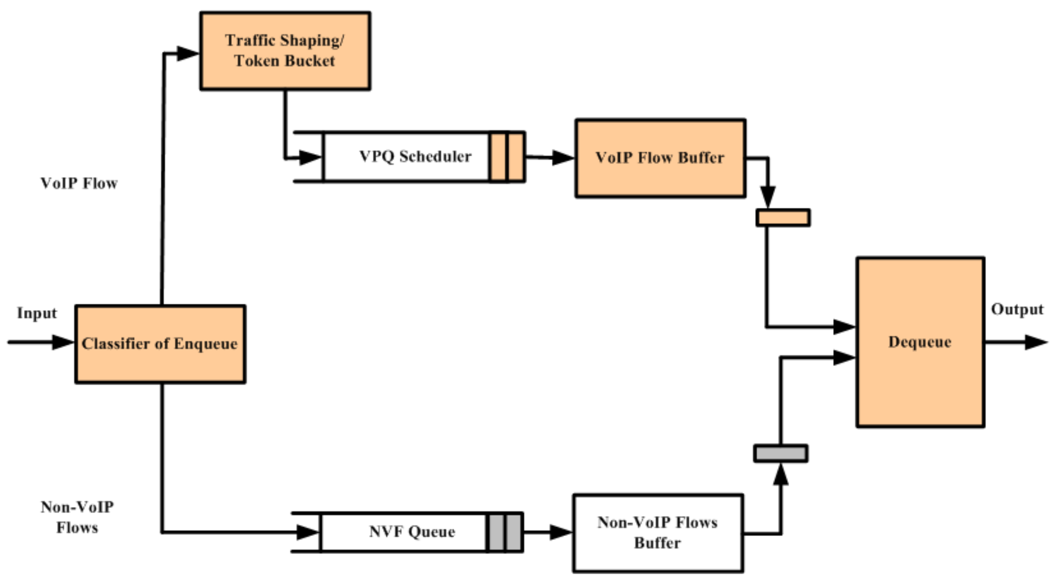


Figure 3.1 First Stage of the VPQ System Model

In Figure 3.1, classification of enqueue traffic flow is sorted-based and the index of the incoming packet is checked. The classification model supports a mechanism that works like Differentiated Services (DiffServ) [135]. After classification, the traffic enqueue sends the VF traffic to the shaping and Token Bucket (TB) flow. The shaping controls the amount of flow and flows sent to the TB flow. The TB applies bursty traffic as the regulated maximum rate of the traffic over WLANs using IEEE 802.11 standards.

Then the VF moves ahead to the main VPQ scheduler flow. The VPQ scheduler forwards these packets into the VoIP Flow Buffer (VFB). The VFB is a temporary buffer flow for VF and NVF traffic. Lastly, the VPQ flow dequeues the traffic flows and sends them to their destinations.

3.3.1.1 Voice Priority Queue Scheduler

To resolve the related work drawbacks, Also, introduce a Voice Priority Queue (VPQ) scheduler that provides the priority to the VoIP Flow (VF) over IP-based networks.

It has two types of traffic flow, the Actual-VoIP Flow (Actual-VF) and Virtual-VoIP Flow (Virtual-VF) over WLANs using IEEE 802.11 standards. Figure 3.2 depicts the simplified model of the proposed VPQ scheduler.

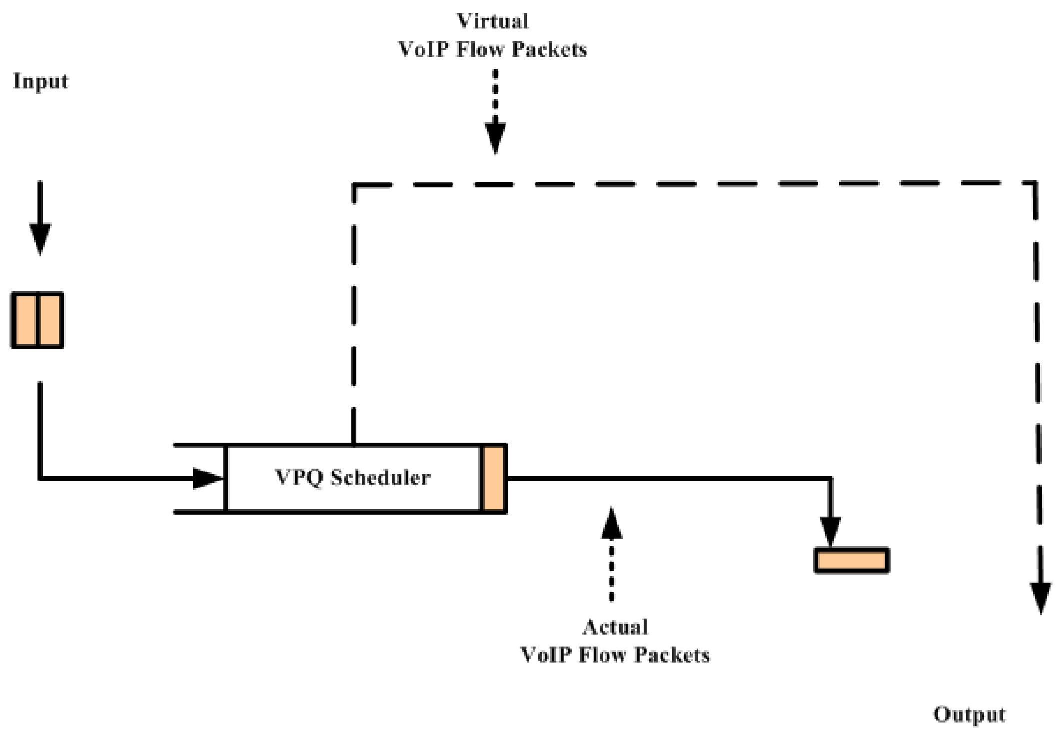


Figure 3.2 VPQ Scheduler

Firstly, packets arrive in the Actual-VF traffic, then the traffic is passed through the dequeue to the end user. Meanwhile, the VPQ receives more bursty traffic due to peak time flow and the VPQ scheduler proposes the Virtual-VF for heavy bursty traffic flow.

Furthermore, a new VPQ scheduler will provide high-speed traffic flow for the VoIP traffic over networks. Scheduler will assign a guaranteed VF with fair scheduling. The VPQ is a scheduler that provides each flow a fair share of the network bandwidth over WLANs. It will resolve the fundamental problems of the VF queue management flow which applies only for the VF traffic. The proposed VPQ scheduler is packet-based with real-time traffic flow that will increase the throughput and fairness of the VF. When bursty traffic arrives then the VPQ will resolve this bursty traffic. The VPQ provides for the VF a constant bandwidth for a fair flow of IP-based traffic over WLANs. Furthermore, the details are as shown in Figure 3.3.

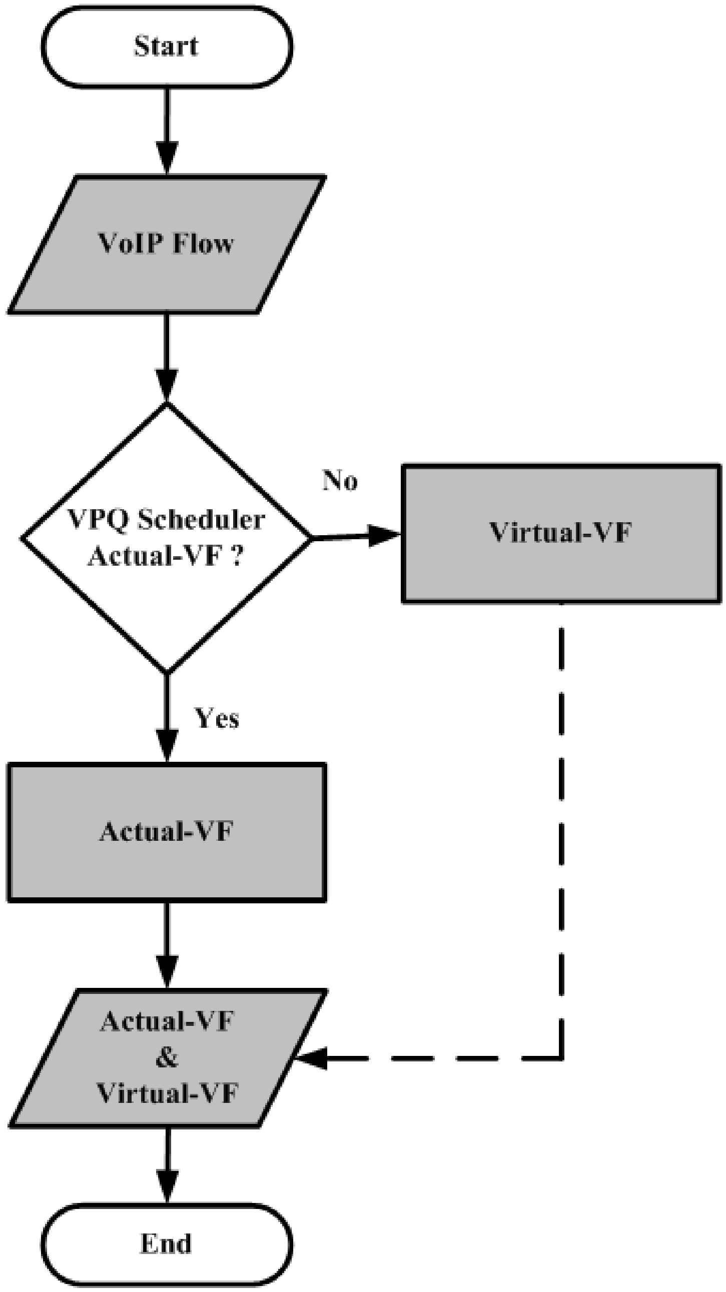


Figure 3.3 Flow Chart of the VPQ Scheduler and First Stage

3.3.1.2 First Stage of the VPQ Scheduling Algorithm

In Figure 3.4, the first stage of the VPQ scheduling algorithm proposes to enqueue and dequeue the VF and NVF traffic flows over WLANs. Firstly, the VPQ scheduler initializes the traffic flows and sends them to the enqueue. Proposed classification component takes the traffic from the enqueue and classifies it into a VF and NVF traffic flow based on the nature of the traffic.

First Stage of the VPQ Scheduling Algorithm

Initialize Traffic Flows over WLANs: (ITF-WLANs)

(Invoke When the VPQ Scheduler is Initialized over WLANs)

Traffic Flow Arrives

On the arrival of the traffic flow Packets (pkt), the *VF and NVF Traffic Flows to the VPQ**

Traffic flow = 0;

*Classification (c) = the VF and NVF;***

***for** (pkt = 0; pkt < n; pkt = pkt + 1)*

pkt = 0;

Enqueue:

(Invoke When Packet Arrives Inside to Enqueue for the Classification of Traffic Flows)

If** (Classification (c) = the VoIP Flow (VF)) **then**

*send the VoIP Flow (VF) to the token bucket (TB) & Traffic Shaper (TS);*****

Else

*send the Non-VoIP Flow (NVF) to the Queue 1 .....N;******

End if;

on arrival of the VoIP Flow(VF) to the token bucket

If** (The token bucket size (pkt) < = the VoIP Flow (VF)) **then

send to the VPQ scheduler for the VoIP Flows (VF);

End If;

If** (The VPQ scheduler for the VoIP < = the VoIP Flo Buffer (VFB)) **then****

send to the VoIP Flow Buffer (VFB);

End if;

on arrival of the Non-VoIP Flows to the Queue 1 .....N

If** (The Non-VoIP Flow < = the Non-VoIP Flow Buffer (Non-VFB)) **then*****

send to the Non-VoIP Flow Buffer (Non-VFB);

End if;

Dequeue:

(Invoke the Packet (pkt) Queue Corresponding to a Different Flow)

If** (The VoIP Flow Buffer (VFB)) < = the Dequeue Traffic Flows (DTF)) **then******

send to the Dequeue Traffic Flow (DTF)Processor;

End if;

If** (The Priority Queue for the VoIP finishes (VF) and (NVF)) **then

Again go to the Initial Traffic Flow;

End if;

Notations

*VPQ = Voice Priority Queue Scheduler

**Classification (c) = VF and NVF

*** VF = VoIP Flow

****TB & TS = Token Bucket (TB) & TS =Traffic Shaper (TS)

*****NVF = Non-VoIP Flow

******VFB = VoIP Flow Buffer

*******Non-VFB = Non- VoIP Flow Buffer

********DTF = Dequeue Traffic Flow

Figure 3.4 First Stage of the VPQ Scheduling Algorithm

After classification of the traffic flow, the VF is sent to the Token Bucket (TB) and the Traffic Shaper (TS), and other NVF traffic is sent to one of the other numbers of queues over the network. The TB < = the VF is then sent to the VPQ scheduler component for traffic flow, and then the VPQ scheduler component < = the VoIP Flow Buffer (VFB) is sent to the VFB traffic flow. The same process works for the NVF traffic flow over the network. In the dequeue stage, the packet queue corresponds to different flows and the VFB < = the Dequeue Traffic Flow (DTF) is then sent to the DTF for all traffic flows. Finally, the traffic flows dequeue to the final destination.

3.3.2 Second Stage of the VPQ System Model

In order to enhance the performance of the Voice Priority Queue (VPQ) system model, in this section, a second stage of the VPQ described as a Virtual-VoIP Flow (Virtual-VF). The VPQ changes into the Virtual-VF mode once the bursty traffic arrives in the VoIP Flow (VF) over WLANs. To provide continuous VF, the Virtual-VF component for peak time bursty traffic flow over the network. This Virtual-VF shows that system model meets the requirements for bursty and Non-Bursty (NB) traffic flow over WLANs using IEEE 802.11 standards.

In Figure 3.5, the Virtual-VF for bursty traffic flow that directly starts from the VPQ scheduler and the dequeueing of the end user traffic flows. To achieve long term fairness, a Virtual-VF is compulsory over IP-based networks and especially for the VF traffic over WLANs.

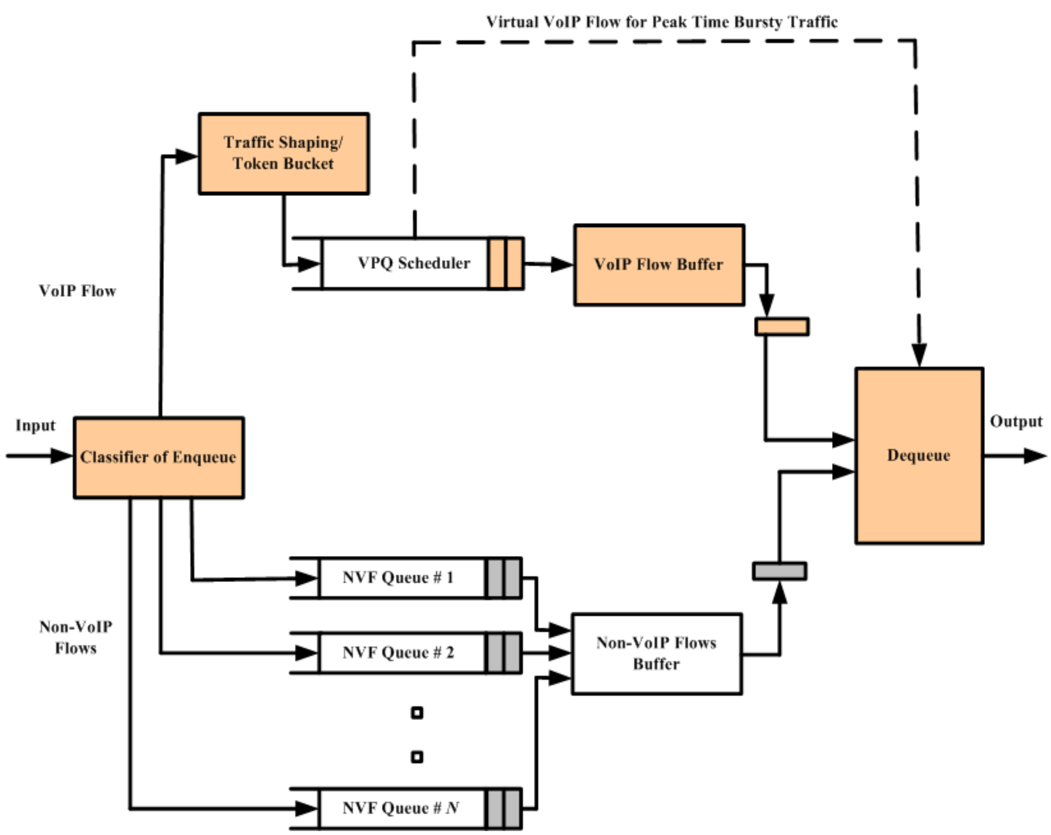


Figure 3.5 Second Stage of the VPQ System Model

In order to support better QoS requirements of the VoIP traffic, the Virtual-VF fulfills the needs of the traffic flow in the bursty environment. The VoIP is a delay-sensitive application for Next Generation Networks (NGNs). The VoIP has the full prospect of replacing the traditional telephonic system and the VoIP flow is a high priority traffic flow over WLANs. It performs the Virtual-VF to meet the real-time traffic flow requirements.

In second stage, propose the system model for bursty traffic flow over WLANs. As discussed in the first stage, with normal traffic flow there is no need for the Virtual-VF but the VoIP is a real-time traffic flow that is why need to introduce the Virtual-VF flow to the most efficient traffic system model such as the VPQ over IP-based networks. The performances of the scheduling algorithms of related work under bursty traffic flow were studied. It applied the Virtual-VF mechanism to generate the voice packets to be safely dequeued to the final destination. For fairness, it assume that the Virtual-VF mechanism applied to the Access Point (AP) has accurate information regarding each node over WLANs using IEEE 802.11 standards. In the following, there will be a detailed discussion about the VPQ traffic scheduler used in the proposed method. Furthermore, the details are as shown in Figure 3.6.

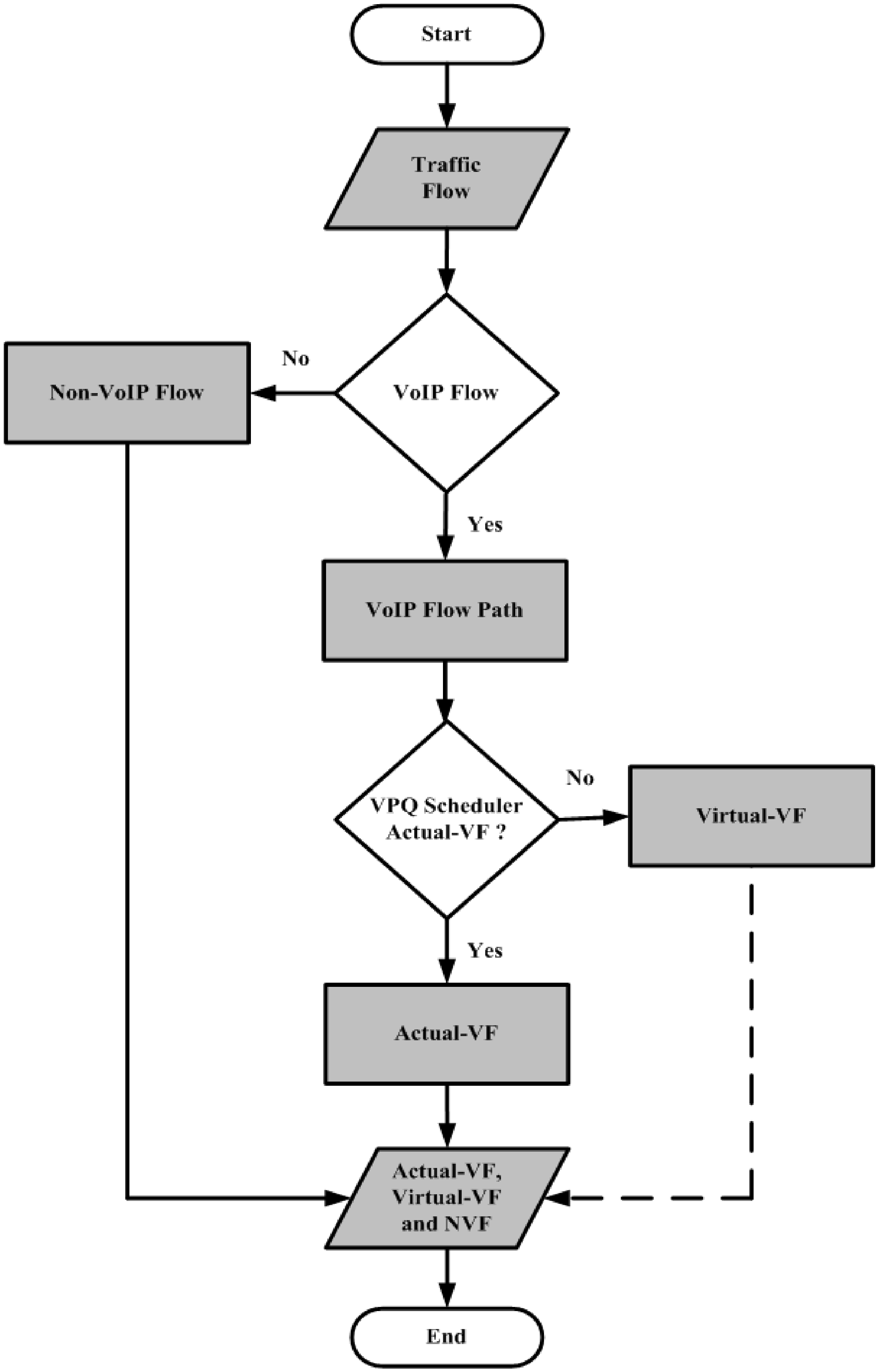


Figure 3.6 Second Stage of the VPQ Flow Chart

3.3.2.1 Second Stage of the VPQ Scheduling Algorithm

Figure 3.7 presents the second stage of the VPQ scheduling algorithm to enqueue and dequeue the VF traffic flow in a bursty environment over WLANs. The second stage of the VPQ scheduling algorithm works as follows: On arrival of the traffic flow to the VF and Virtual-VF, if the Token Bucket (TB) size of packet (pkt) = > the VF then send to the Virtual-VF component for bursty traffic over the network. Meanwhile, if the Virtual-VF < = the Dequeue Traffic Flow (DTF) then proceed to dequeue for the end user.

Second Stage of the Voice Priority Queue (VPQ) Scheduling Algorithm

Initialize The Virtual-VoIP Flow (Virtual-VF) for Bursty Traffic*

(Invoke When the VPQ Scheduler is Initialized for the Virtual-VF)

On the arrival of the traffic flow Packet (pkt) the Virtual-VF to the VPQ

Traffic flow = 0;

*Classification (c) = the VF and NVF;***

the Virtual VoIP Flow = the Virtual-VF

***for** (pkt = 0; pkt < n; pkt = pkt + 1)*

pkt = 0;

Enqueue:

(Invoke When the Packet Arrives for the VF and Virtual-VF)

If** (Classification (c) = the VoIP Flow (VF)) **then**

*send the VoIP Flow (VF) to the token bucket (TB) & Traffic Shaper (TS);*****

Else

*send the Non-VoIP Flow (NVF) to the Queue 1.....N;******

End if;

If** (The token bucket size (pkt) < = the VoIP Flow (VF)) **then

send to the VPQ scheduler for the VoIP Flow (VF);

End If;

If** (The VPQ scheduler for the VoIP < = the VoIP Flow Buffer (VFB)) **then****

send to the VoIP Flow Buffer (VFB);

End if;

on arrival of the VoIP Flow(VF) to the token bucket

If** (The token bucket size (pkt) = > the VoIP Flow (VF)) **then

send to the Virtual-VoIP Flow (Virtual-VF)

the Virtual-VF component for the VoIP Flow (VF) for Bursty Traffic over the Network;

End if;

Dequeue:

(Invoke the Packet (pkt) flow to the Virtual-VF)

If** (The Virtual-VF < = the Dequeue Traffic Flow (DTF)) **then****

send to the Dequeue Traffic Flow (DTF)Processor;

End if;

If** (The Virtual-VF and VF finish) **then

Again go to the Initial Traffic Flow;

End if;

Notations

***Virtual-VF =** Virtual VoIP Flow*

****Classification (c)** = VF and NVF*

*****VF =** VoIP Flow*

******TB & TS =** Token Bucket (TB) & TS =Traffic Shaper (TS)*

*******NVF =** Non VoIP Flow*

********DTF =** Dequeue Traffic Flow*

Figure 3.7 Second Stage of the VPQ Scheduling Algorithm

3.3.3 Third Stage of the VPQ System Model

A third stage of the Voice Priority Queue (VPQ) system model propose over a WLAN. It also provides a switch that moves to the VoIP Flow (VF) from the Non-VoIP Flow (NVF) when the NVF flows are empty due to non-real-time traffic over the network.

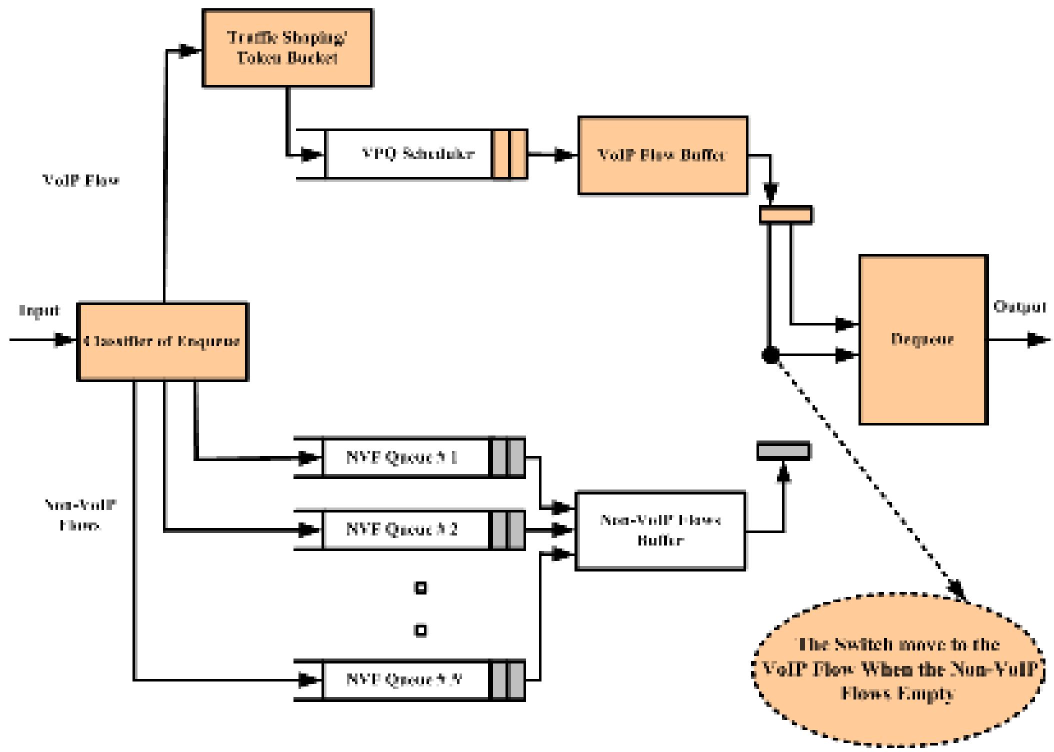


Figure 3.8 Third Stage of the VPQ System Model

In Figure 3.8, the switch mechanism proposes more fairness in the VF traffic over WLANs using IEEE 802.11 standards. This switch controls the bursty traffic flow at the peak time. It provides a key technique proposed by VPQ scheduler. The key technique shows the high priority and low priority queue management system over IP-based networks. Furthermore, the details are as shown in Figure 3.9.

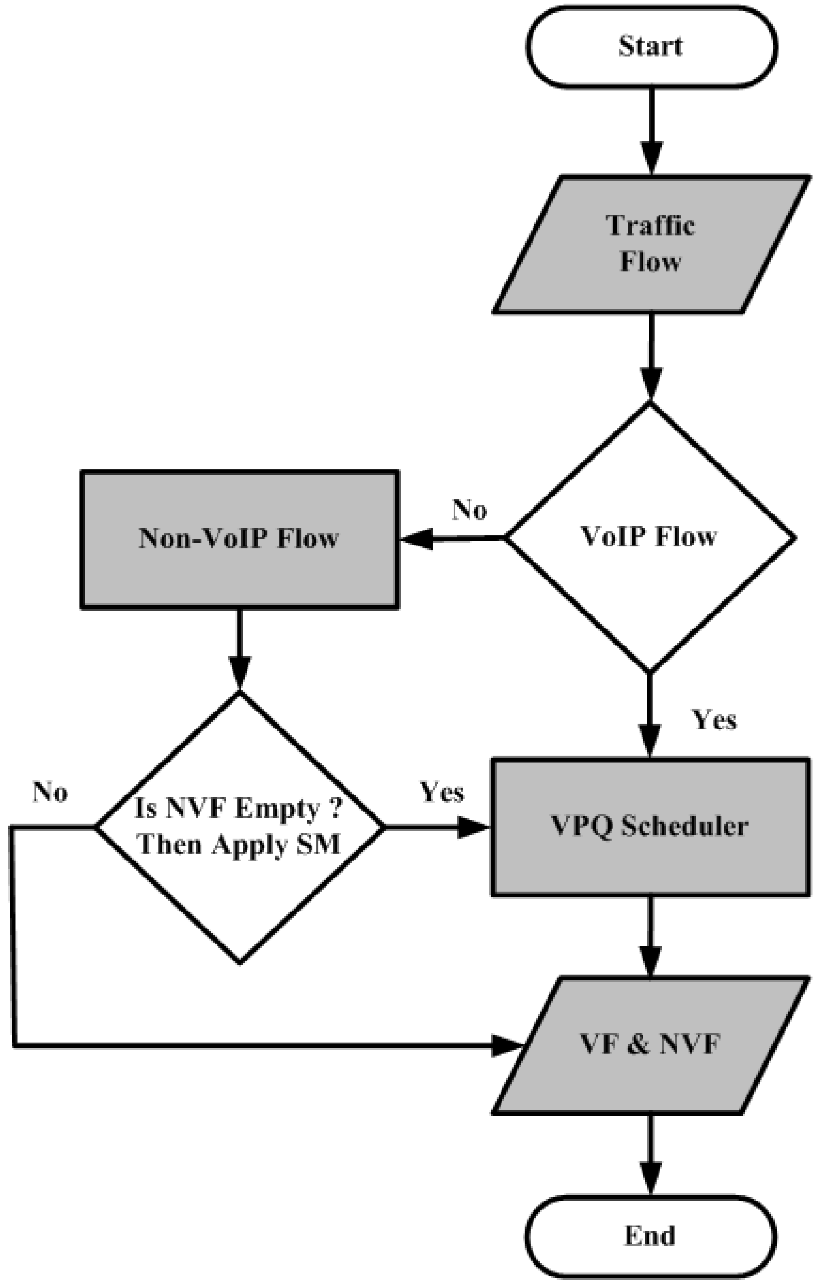


Figure 3.9 Third Stage of the VPQ Flow Chart

3.3.3.1 Third Stage of the VPQ Scheduling Algorithm

In Figure 3.10, the third stage of the VPQ scheduling algorithm proposes to enhance the performance of the VoIP over WLANs. Firstly, initialize the Switch Movement (SM) to the VoIP Flow (VF) for bursty traffic over the network.

Third Stage of the VPQ Scheduling Algorithm

Initialize the Switch Movement (SM) to the VoIP Flow (VF) for Bursty Traffic*

(Invoke When the VPQ Scheduler is Initialized to Switch)

On the arrival of the traffic flow Packet (pkt) to the VPQ

Traffic flow = 0;

*Classification (c) = the VF and NVF;***

Switch Movement = SM

***for** (pkt = 0; pkt < n; pkt = pkt + 1)*

    pkt = 0;

Enqueue:

(Invoke When the Packet Arrives for the VF)

  If** (Classification (c) = the VoIP Flow (VF)) **then**

    *send the VoIP Flow (VF) to the token bucket (TB) & Traffic Shaper (TS);*****

  End if;

  If** (The token bucket size (pkt) < = the VoIP Flow (VF)) **then

    send to the VPQ scheduler for the VoIP Flows (VF);

  End If;

  If** (The VPQ scheduler for the VoIP < = the VoIP Flow Buffer (VFB)) **then****

    send to the VoIP Flow Buffer (VFB);

  End if;

 If** (The Non-VoIP Flow (NVF) Empty to the Queue 1.....N) **then

    The Switch moves to the VoIP Flow (VF);

  End If;

Dequeue:

(Invoke the Packet (pkt) flow to Switch Movement (SM))

  If** (The Switch Movement < = the Dequeue Traffic Flow (DTF)) **then****

    send to the Dequeue Traffic Flow (DTF)Processor;

  End if;

  If** (The VoIP Flow (VF) finishes) **then

    Again go to the Initial Traffic Flow;

  End if;

Notations

***SM = *Switch Movement*

****Classification (c)* = *VF and NVF*

*****VF = *VoIP Flow*

******TB & TS = *Token Bucket (TB) & TS =Traffic Shaper (TS)*

*******NVF = *Non VoIP Flow*

********DTF = *Dequeue Traffic Flow*

Figure 3.10 Third Stage of the VPQ Scheduling Algorithm

The SM allocates the Non-VoIP Flow (NVF) bandwidth to the VF when the NVF buffer is empty. This will further enhance the performance of the VoIP over WLANs.

3.4 Voice Priority Queue Components

The Voice Priority Queue (VPQ) has the following components: the VPQ traffic initialization, the traffic flow classifier, the VoIP Flow (VF) and Non-VoIP Flow (NVF), Traffic Shaping (TS), the Token Bucket (TB), VPQ Scheduler, the VoIP-Flow buffer, the Non-VoIP Flow Buffer and the dequeue traffic flow for the end user over WLANs using IEEE 802.11 standards. These components are discussed in detail below.

3.4.1 Traffic Classifier of Enqueue Component

The traffic classifier works once the traffic is initialized. They are classifying traffic flows based on the traffic flow id of the incoming packet in the enqueue component. The classification of the enqueue component to identify the multiple traffic flows over networks. The Voice Priority Queue (VPQ) divides the traffic flow in two; the two types of traffic flow are called VoIP Flow (VF) and Non-VoIP Flow (NVF) as shown in Figure 3.11. The classification is applied on the segment of the packet header.

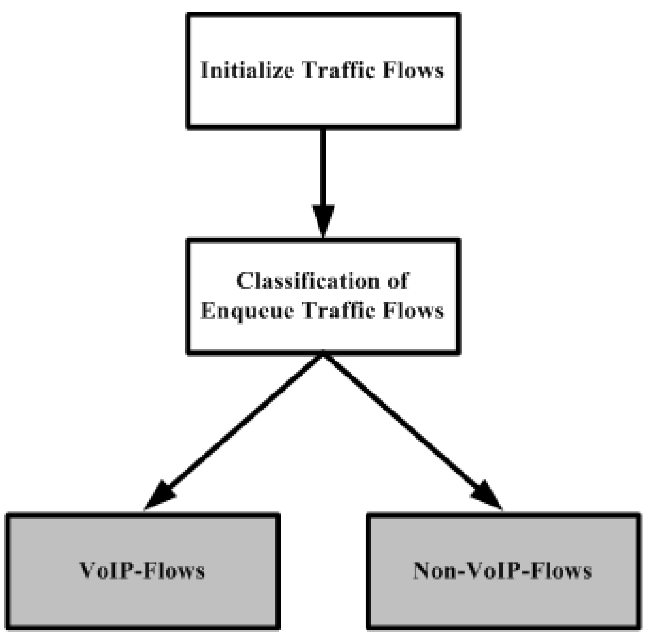


Figure 3.11 Initialize traffic classifications as VF and NVF Component

The classification model supports a mechanism that works like Differentiated Services (DiffServ) [135]. The classification of the enqueue differentiates between the VF and NVF traffic flow over WLANs using IEEE 802.11 standards. Figure 3.12 shows the block diagram of the classifier model.

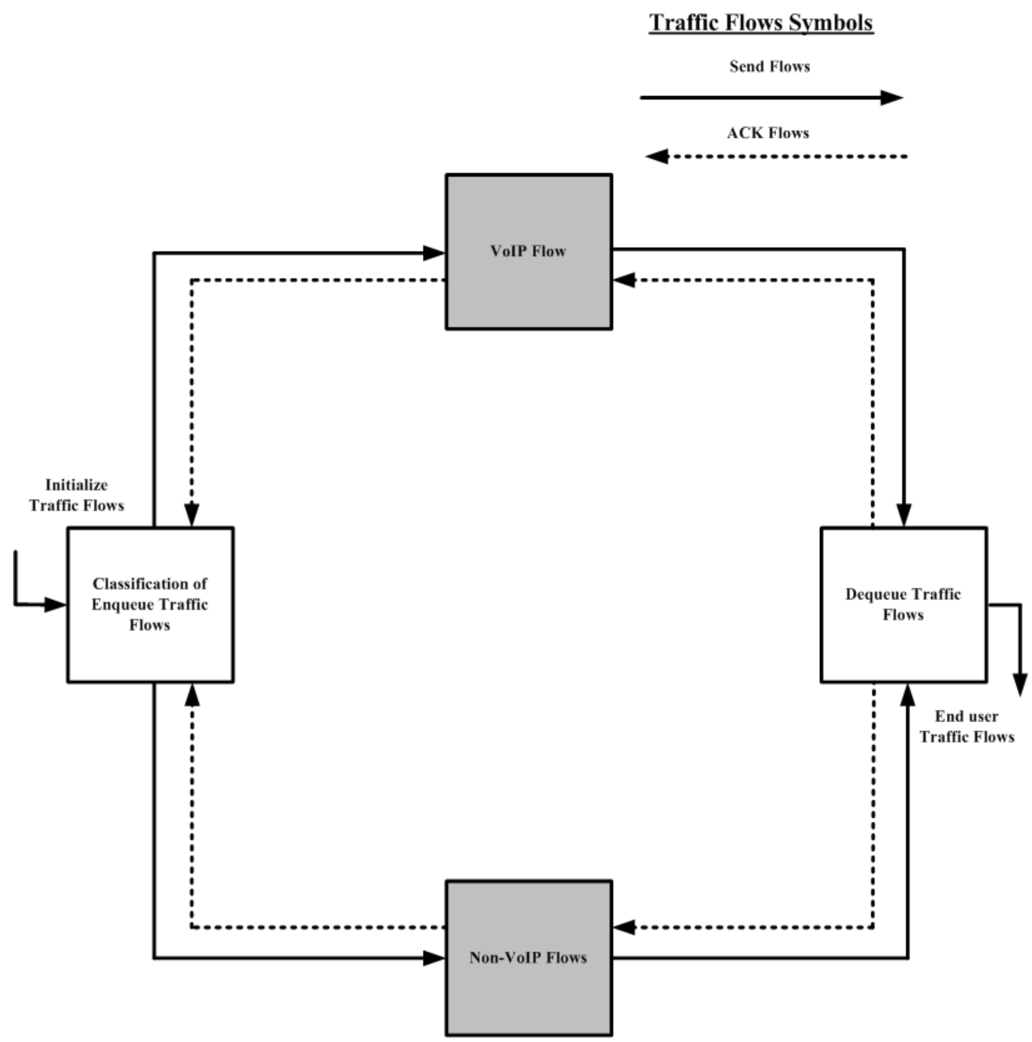


Figure 3.12 Traffic classifier of Enqueue to Dequeue Traffic flow Component

It classifies the packets into the VF and NVF and then forwards the packets to the next component for onward processing. It provides both Fair Queue (FQ) and strict priority (SP) queuing to the VF over WLANs using IEEE 802.11 standards. The classifier performs the following tasks: classification of the enqueue traffic into the VF and NVF and classification of the dequeue traffic.

3.4.1.1 VoIP-Flow and Non-VoIP-Flow Components

After classification of the traffic, it manage the VoIP Flow (VF) and Non-VoIP Flow (NVF) traffic in bursty and Non-bursty traffic over IP-based networks. The classifier sends the VF to Traffic Shaping (TS) and the Token Bucket (TB). Furthermore, the classifier sends the NVF traffic to a number of queues over WLANs using IEEE 802.11 standards. Figure 3.13 shows the block diagram of the enqueue and the dequeue of the VF and NVF traffic flow over IP-based networks.

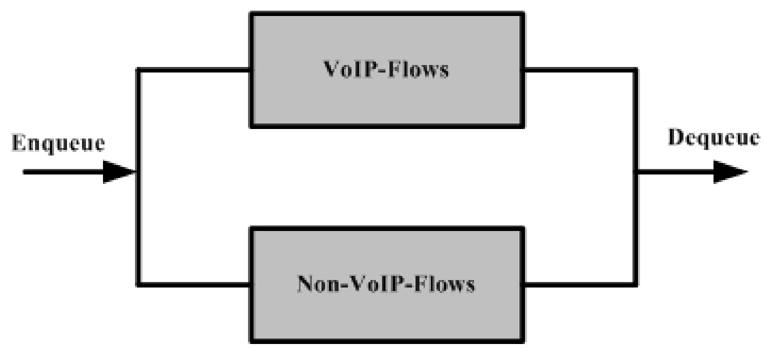


Figure 3.13 VF and NVF, Enqueue to Dequeue

It provides the algorithm that supports both the VF and NVF traffic in bursty and Non-Bursty (NV) traffic over IP-based networks.

3.4.2 Traffic Shaping and the Token Bucket Component

Traffic Shaping (TS) is a technique to enhance performance of a scheduler on a VoIP over WLANs. In this section, discuss TS and Token Bucket (TB) techniques that are applied to enhance the Quality of Service (QoS) of a new Voice Priority Queue (VPQ) scheduler.

Traffic Shaping (TS) is a method that manages the amount and the rate of the VoIP Flow (VF) traffic that should be sent over the network. In Figure 3.14, the Traffic Shaping (TS) has introduced the TB technique over a WLAN and apply it to the VPQ scheduler.

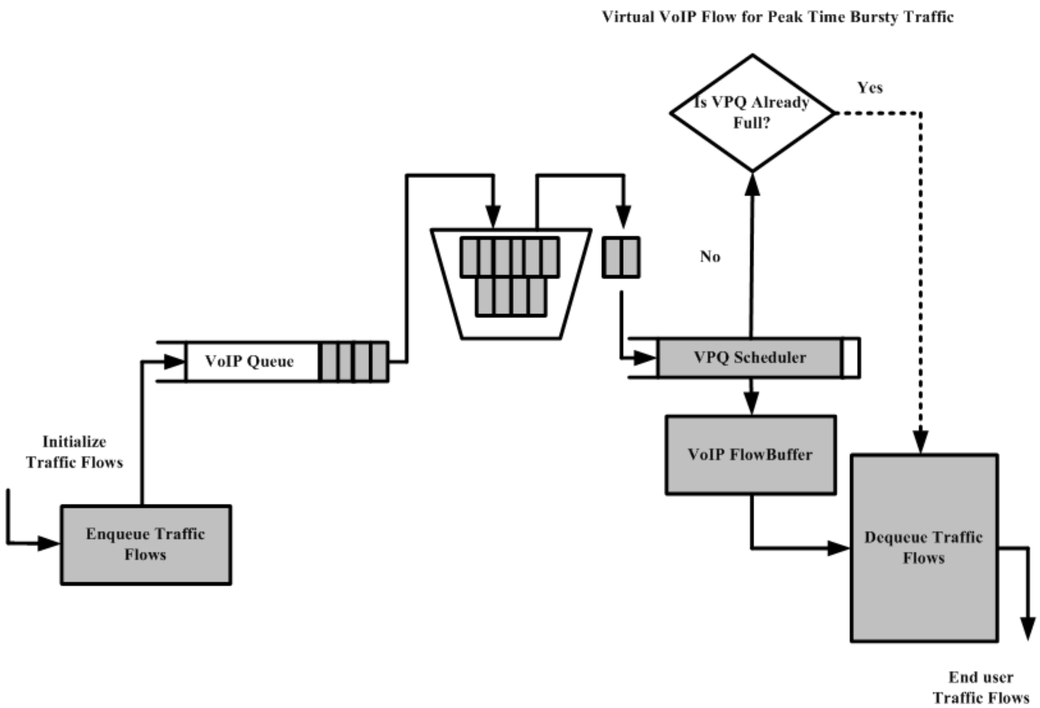


Figure 3.14 Process of the TB for the VoIP Flow Component

The TB is applied to the VF in bursty traffic situations and regulates the traffic rate over WLANs. Furthermore, the token bucket process begins with a counter that starts from zero in the bucket. Step by step, the token bucket counter is incremented by 1token. If the counter is zero, the host cannot send the data to the VF traffic.

3.4.3 VPQ Scheduler Component

It has traffic flow, the VoIP Flow (VF) over WLANs using IEEE 802.11 standards. Figure 3.15 depicts the simplified architecture of the proposed VPQ.

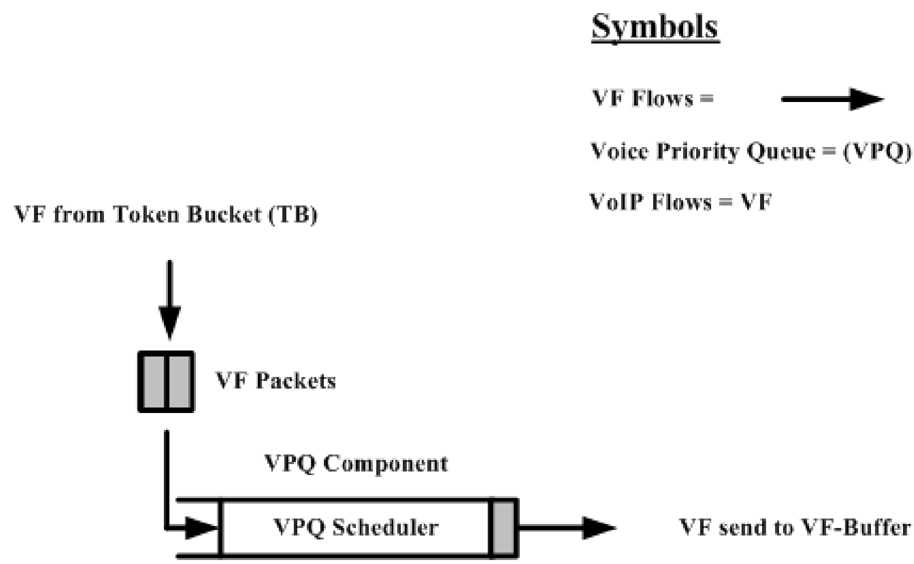


Figure 3.15 VPQ Scheduler Component for the VF

Firstly, packets arrive in the initialized traffic then the classifier differentiates the traffic into the VF and NVF traffic. The VF packets from the TB are moved to the VPQ component that temporarily stores and forwards the packets to the VoIP Flow Buffer (VFB).

3.4.4 VFB and Non-VFB Components

The VoIP Flow Buffer (VFB) temporarily stores the VF and the Non-VoIP Flow Buffer (Non-VFB) stores the NVF concurrently in WLANs. It has been discussed briefly in the previous section but it need to discuss the VFB and Non-VFB in detail. It divide the link-sharing of the VF and NVF among multiple types of traffic, for example, the NVF may have Video conferencing, FTP, IPTV and data traffic all at the same time over IP-based networks as in Figure 3.16.

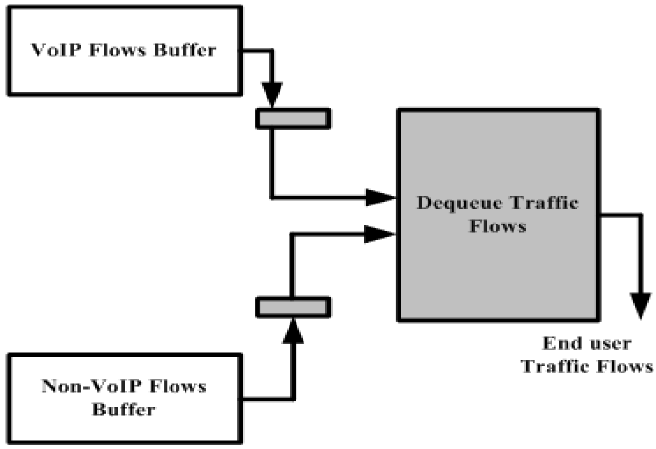


Figure 3.16 VFB and Non-VFB Components

It will deal with classification of high-level mechanisms and low-level mechanisms that provide the simple resource management model that allows for evaluation. The buffer applies for temporary storage of both types of traffic, the VF and NVF over a network. The VPQ scheduler's classification can make better use of both its buffers, the VF and NVF.

Furthermore, these buffers are useful for bursty traffic as they help to manage bursty traffic. The buffer takes the minimum action required to ensure that the traffic receives its allocated link-sharing bandwidth over the relevant time interval.

3.4.5 Virtual-VoIP Flow for the Bursty Traffic Component

A unique link-sharing mechanism referred to as the Virtual-VoIP Flow (Virtual-VF); it is only for bursty traffic over WLANs. The traffic flow is divided into the VF and NVF flows that use the link-sharing classification mechanism. In the VF, incoming packets wait at the gateway to the most suitable flow for that output link. For normal VF flow, the classification mechanism is sufficient but for bursty traffic flow it need to introduce a component named the Virtual-VF as shown in Figure 3.17.

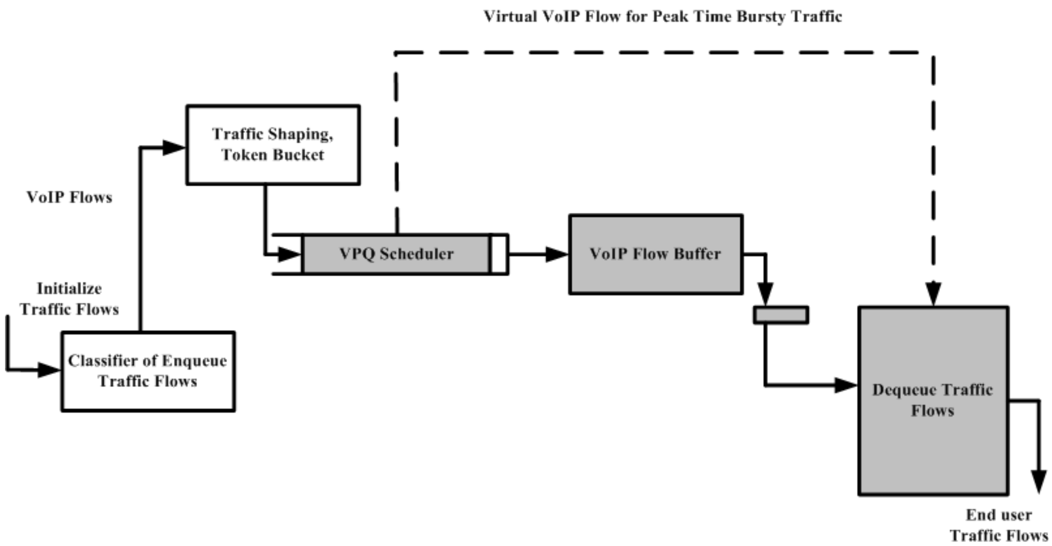


Figure 3.17 Virtual-VF Component for Bursty Traffic

A key attribute of the Virtual-VF component is the ability to share bandwidth with the VF and NVF traffic over WLANs. If it's a bursty traffic flow, then the Virtual-VF starts from the VPQ scheduler compnent and sends it directly to the most suitable flow for that output link as dequeued traffic flow for the end user. In addition, the Virtual-VF will be applied on the same bandwidth that is used for the VF flow.

Figure 3.17 shows both the flows i.e. the VF flow and the Virtual-VF flow. The Virtual-VF manages the bursty traffic. The Virtual-VF bypasses two components, the Voice Priority Queue (VPQ) scheduler component and the VoIP Flow Buffer (VFB).

3.4.6 VoIP Flow-Switch Component

In this section, it provides details of the VoIP Flow-Switch (VFS) component over WLANs. As it has two types of traffic classifications, the VF and NVF, it introduce the Switch Movement (SM) concept to the VF flow when the NVF flow is empty as shown in Figure 3.18.

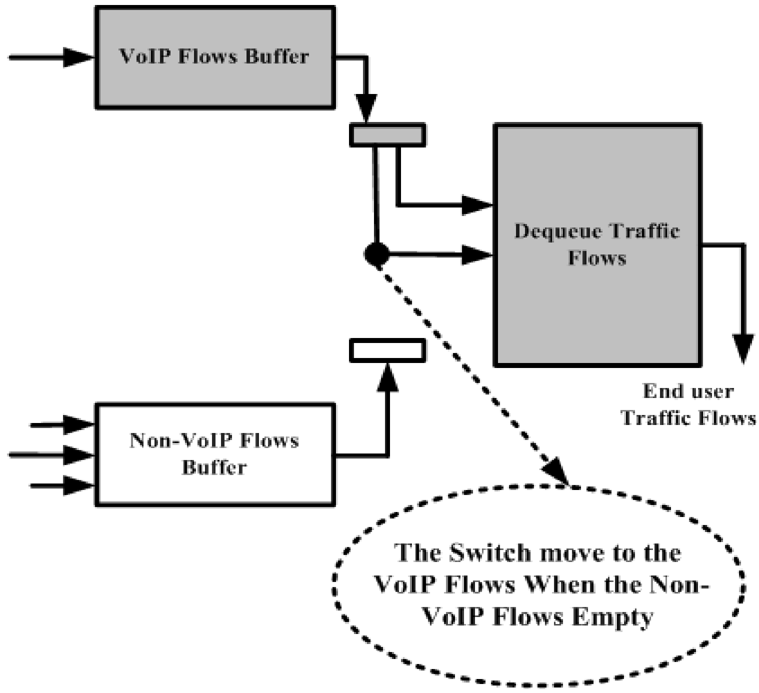


Figure 3.18 VoIP Flow-Switch Component

The VFS is a moveable switch for VF traffic over WLANs. Normally, the NVF traffic is non real-time traffic and is low priority traffic as compared to the VF traffic. The NVF does not use the allocated bandwidth all the time and is normally not delay sensitive.

It can utilize those flows for the VF traffic to handle brusty traffic. Now, it has three types of flows; they are the VF, the Virtual-VF and the moveable switch over WLANs using IEEE 802.11 standards.

3.4.7 VF and NVF Dequeue Traffic Flow Component

Finally, the VF and NVF traffic will move to the Dequeue Traffic Flow (DTF) component and be received by the end user from three ways VF, NVF and Virtual-VF with the Voice Priority Queue (VPQ) scheduler as shown in Figure 3.19.

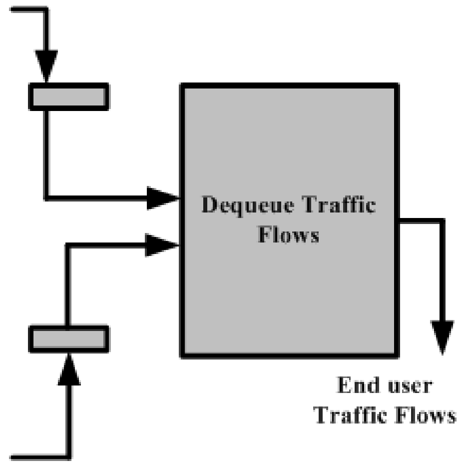


Figure 3.19 VF and NVF Dequeue Traffic Flow Component

The DTF flows both types of traffic and it will discuss these components in experimental and simulation chapter.

3.5 Final Stage of VPQ System Model

In Figure 3.20, the Final Stage of the Voice Priority Queue (VPQ) Scheduling system model is a combination of all the components. The final stage of the VPQ has the first, second and third stages of the scheduler system model over WLANs. The final stage of the VPQ supports the VoIP Flow (VF) in three ways.

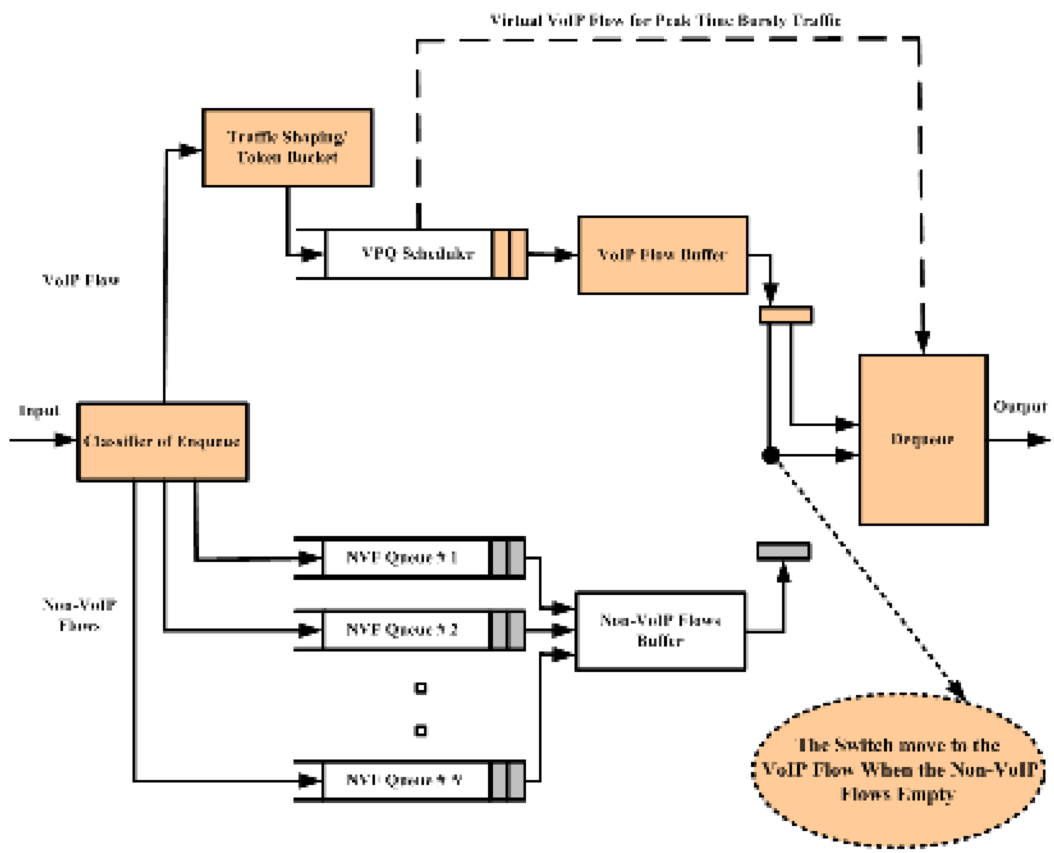


Figure 3.20 Final Stage of the VPQ System Model

The final stage of the VPQ initializes traffic in the VF and Non-VoIP Flow (NVF) for enqueueing and dequeueing of the traffic over WLANs. The second stage of the VPQ proposes a Virtual-VoIP Flow (Virtual-VF) for bursty and Non-Bursty (NB) flows over networks. The third stage of the VPQ introduces a switch that moves to the VF from the NVF when NVF flows are empty over the network. The final stage of the VPQ fully supports the VF and NVF traffic flow over WLANs. Furthermore, the details are as shown in Figure 3.21.

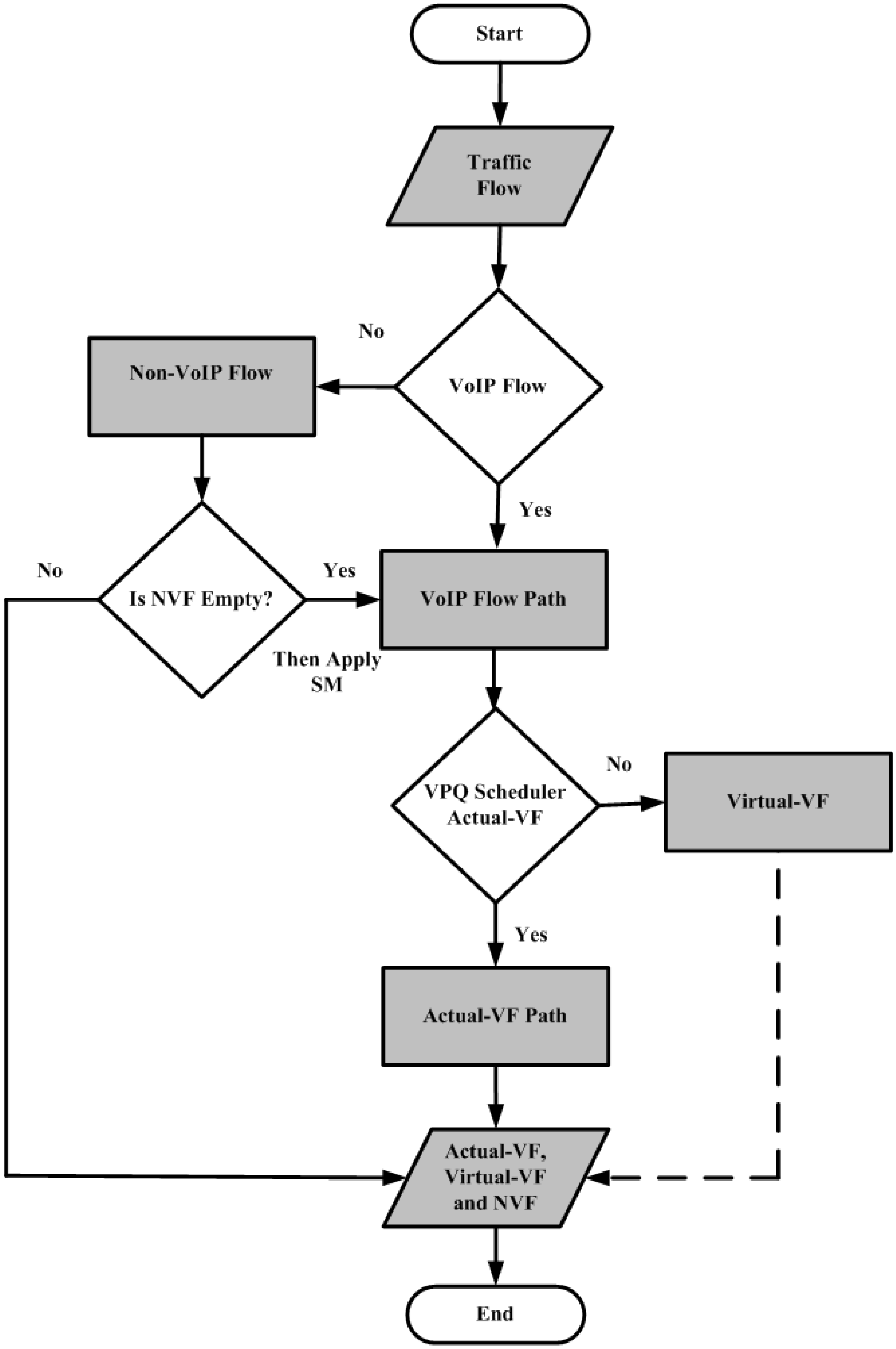


Figure 3.21 Final Stage of the VPQ Flow Chart

3.5.1 Final Stage of the VPQ Scheduling Algorithm

Firstly, the VPQ scheduling algorithm initializes the traffic flow over WLANs. After the initialization, the traffic flow arrives to enqueue and the classification component divides the traffic flow into the VoIP Flow (VF) and Non-VoIP Flow (NVF). After that, invoke the packets to the Token Bucket (TB) and Traffic Shaper (TS), if the TB size < = to the VF then the traffic is sent to the VPQ component. At the same time if the TB size = > the VF then send it to the Virtual-VF for bursty traffic over WLANs. The details of the final stage of the VPQ are as shown below in Figure 3.22.

Final Stage of the Voice Priority Queue (VPQ) Scheduling Algorithm

```
Initialize*
Traffic Flows over WLANs: (TF-WLANs)**
The Virtual VoIP Flow (Virtual-VF) for Bursty Traffic over a Network***
The Switch Movement (SM) to VoIP Flow (VF) for Bursty Traffic****
(Invoke When the VPQ Scheduler is Initialized over WLANs)
Traffic Flow Arrives
On arrival of the traffic flow Packets (pkt) the VF and NVF Traffic Flows to the VPQ*****
Traffic flow = 0;
Classification (c) = the VF and NVF;
Virtual VoIP Flow = Virtual-VF
Switch Movement = SM
for (pkt = 0; pkt < n; pkt = pkt + 1)******
        pkt = 0;
Enqueue:*******
(Invoke When the Packet Arrives Inside to Enqueue for Classification of Traffic Flow)
   If (Classification (c) = the VoIP Flow (VF)) then********
         send the VoIP Flow (VF)  to the token bucket (TB)
         & Traffic Shaper (TS);*********
     Else
         send the Non-VoIP Flow (NVF) to the Queue 1.....N;**********
     End if;
         on arrival of the VoIP Flow(VF) to the token bucket
   If (the token bucket size (pkt) < = the VoIP Flow (VF)) then
         send to the V PQ Scheduler for the VoIP Flow (VF);
```

Else

send to the Virtual-VoIP Flow (Virtual-VF)

the Virtual-VF component for the VoIP Flow (VF) for Bursty Traffic over a Network;

End If;

If** (The Non-VoIP Flow (NVF) Empty to the Queue 1.....N) **then

The Switch moves to the VoIP Flow (VF);

End If;

If** (The Priority Queue for the VoIP < = the VoIP Flow Buffer (VFB)) **then************

send to the VoIP Flow Buffer (VFB);

End if;

on arrival of the Non VoIP Flow to the Queue 1.....N

If** (The Non-VoIP Flow < = the Non-VoIP Flow Buffer (Non-VFB)) **then*************

send to the Non-VoIP Flow Buffer (Non-VFB);

End if;

Dequeue:

(Invoke the Packet (pkt) Queue Corresponding to a Different Flow)

If** (The VoIP Flow Buffer (VFB)) < = the Dequeue Traffic Flow (DTF)) **then

send to the Dequeue Traffic Flow (DTF)Processor;

End if;

(Invoke the Packet (pkt) flow to Virtual-VF)

If** (The Virtual-VF < = the Dequeue Traffic Flow (DTF)) **then****************

send to the Dequeue Traffic Flow (DTF)Processor;

End if;

(Invoke the Packet (pkt) flow to *the* Switch Movement (SM))

If** (The Switch Movement < = the Dequeue Traffic Flow (DTF)) **then

send to the Dequeue Traffic Flow (DTF)Processor;

End if;

If** (The Priority Queue for the VoIP finishes the (VF) and (NVF)) **then

Again go to the Initial Traffic Flow;

End if;

Notations

***Initialize** = *Start the Traffic Flow*

****TF-WLAN** = *Traffic Flow over WLANs.*

*****Virtual-VF** = *Virtual VoIP Flow (Virtual-VF)*

******SW** = *Switch Movement (SW)*

*******VPQ** = *Voice Priority Queue Scheduler*

********Pkt** = *Packet (pkt)*

*********Enqueue** = *Enqueue is a Standard Queue Operator*

******** **VF** = *VoIP Flow*

***********TB & TS** = *Token Bucket (TB) & TS =Traffic Shaper (TS)*

************NVF** = *Non VoIP Flow*

*************VFB** = *VoIP Flow Buffer*

**************Non-VFB** = *Non- VoIP Flow Buffer*

************* **DTF** = *Dequeue Traffic Flow*

Figure 3.22 Final Stage of the VPQ Scheduling Algorithm

Meanwhile, if the NVF queue is empty then Switch Movement (SM) to the VF traffic flow and dequeue all VF traffic over WLANs.

3.6 Summary

A new Voice Priority Queue (VPQ) scheduling system model and algorithms were proposed in this chapter. The VPQ was basically considered to satisfy the unique requirements of the VoIP traffic flow over IP-based networks. It has also provided a detailed system model of the VPQ for the performance and fairness over networks. The propose method will resolve some of the issues on VoIP scheduling as real-time traffic over WLANs.

In the following chapter, it will discuss the simulation and experiment environment. Therefore, the next chapter introduces an experimental step to evaluate the above scheduling system model and algorithm techniques. There will also be a comparison and evaluation among the related scheduling algorithms.

CHAPTER 4

SIMULATION AND EXPERIMENTAL SETUP

4.1 Introduction

This chapter will discuss the simulation and experimental setup of proposed scheduling system model and algorithms. It will perform verification and validation of the developed simulation scenarios. It will explain all stages of the new Voice Priority Queue (VPQ) scheduling system model and algorithms over WLANs using IEEE802.11 standards.

Furthermore, it will provide an experimental setup of the scheduling models considered in research. It will explain all stages of the VPQ and simulate VPQ scheduler components. The VPQ is based on the two types of traffic flow namely the VoIP Flow (VF) and Non VoIP Flow (NVF) as discussed in the chapter three. VPQ traffic will initialize from classification of enqueue traffic flows to dequeue traffic flows for the end user.

4.2 Simulation Tools

Simulation tools are helpful for validation and verification of scheduling model and algorithms over IP-based networks. Simulation tools provide multiple topologies, scenarios, models and situations. These tools will act like a real environment but with low cost and simple implementation [136], [137], [138] and [139].

Simulation tools are commonly used paradigms to study communication and networks. They are used to study existing systems or to model newly proposed models and algorithms. Simulation tools study without building a test-bed over IP-based networks. A number of credible published research works have been done using network simulation. They have appeared in IEEE/ACM journals and proceedings. It is tremendously important to select suitable simulator tools for an enhanced performance of the VoIP over WLANs. They have found some simulator tools for network performance over IP-based networks. Some of the commonly used simulation tools are as shown below in Table 4.1.

Symbols:

License = L

License Free = LF

Support = √

Not Support = X

Partial Support = P √

Table 4.1 Selection of Simulation Tool

Simulator	VoIP Support	WLANs Support	License or Free
OMNET++ [140]	P √	√	LF
J-SIM [141]	P √	P √	LF
QualNet [142]	P √	√	L
OPNET [143]	P √	√	L
MATLAB [144]	X	P √	L
TinyOS [145]	X	X	LF
VipTos [145]	X	P √	LF
NS-2 [146]	√	√	LF

4.3 Comparison of Techniques with Network Simulation

Measurement can be done on a test-bed network or an equipped network. Measurement needs real hardware equipment, codes and time to run for experiments [147]. The limitations of a measurement test-bed can be difficult to configure and reconfigure. Test-beds can be very expensive. The details are as shown in Table 4.2.

Table 4.2 Comparison of Techniques [147]

	Simulation	Measurement	Analytical Modeling
Cost	Free / Moderate	High	Medium
Accuracy	Good	Moderate	Moderate
Time Required	Moderate	High	High

Analytical modeling is based on mathematical notations and describes performance aspects of the system under study. Analytical modeling has limitations that require too many simplifications and assumptions while ignoring network dynamics such as flow interactions over WLANs based on IEEE 802.11 standards.

Simulation tools are one of the most commonly applied paradigms in the learning of communication networks. The network simulation tools used to ensure that functional requirements of newly proposed algorithms, protocols etc. are working properly.

Simulation tools provide cheaper communication and can simulate the systems. The benefits of the simulation define the network system under study, identify system workload, design the experiments and present the results.

4.4 NS-2 Simulations and Results Analysis Process

NS-2 is based on OTcl scripts to setup network topologies for the VoIP over WLANs using IEEE 802.11 standards. Normally, the NS-2 simulation process consists of the following steps: The Tcl Simulation Codes, NS-2 Executable Tcl Interpreter, NS-2 Simulator Library, Simulation Results, Results Processing and finally production of results into two different formats i.e. trace file analysis and Network Animator (NAM). Figure 4.1 illustrates in two-way simulations and results process over networks.

The NAM presents the results in a visual format. The NS-2 supports the real-time flow especially for the VoIP traffic schedulers over WLANs. It can create multiple topologies using nodes and a packet forwarding technique. It can also connect the nodes to form links. The NS-2 provides a queue management mechanism where packets are temporarily stored. The packet scheduling and queues show the locations where packets may be held or dropped over IP-based networks.

The NS-2 is an event-based simulator tool that supports the scheduling technique using different data structures such as heap, simple linked-list, calendar queueing and real-time over network. The unit of time applied by the scheduler is seconds (sec). With a real-time scheduler such as (class Scheduler/RealTime), it can create a number of topologies for real-time applications especially for the VoIP application over WLANs. The NS-2 also supports the classification method that should map the values of departing interface objects that are next in line for receiving packets downstream. The NS-2 manages simple and multiple classification methods and queues that represent the location where packets maybe held over a network [146].

It supports drop-tail First-In-First-Out (FIFO), Class Based Queue (CBQ), RED Queue management, Fair Queue (FQ), Stochastic Fair Queue (SFQ) and Deficit Round Robin (DRR) as they have already discussed in detail in chapter two. Furthermore, The NS-2 supports differentiated traffic services like classification of traffic over WLANs. It will implement new Voice Priority Queue (VPQ) scheduling model and algorithms over WLANs using IEEE 802.11 standards.

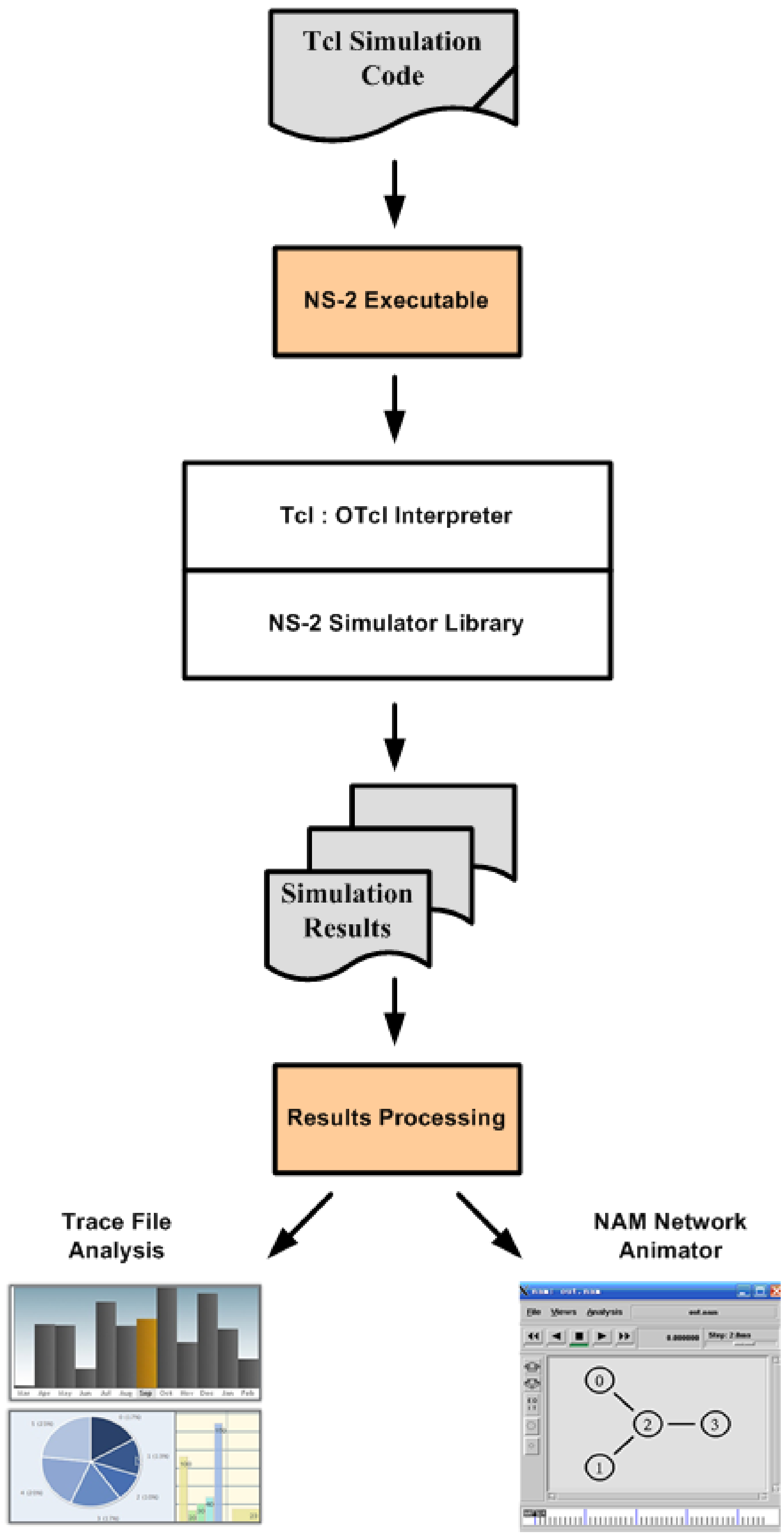


Figure 4.1 NS-2 Two-Way Simulations and Results Analysis Process

4.5 Fairness Measurement

In this section, it will measure the fairness of the VPQ scheduler and compare it with related schedulers. They have noticed that most of the published schedulers do not offer the actual quality of fairness [148]. The VPQ, offers the actual quality of a fair scheduler due to always reaching the upper bound to achieve maximum fairness. The other schedulers, on the other hand, hardly reach the upper bound of relative fairness.

Fairness could be measured in time allocation, which is known as the temporal fairness measurement; this is an important parameter in the field of computer networks. Temporal fairness could be measured as throughput fairness or delay fairness. The fairness can be computed as in the following in equation 4.1 [149]:

$$fairness\ index\ = \frac{\left(sum\ (Xi)\right) ** 2}{number\ of\ flows\ sum\ (Xi ** 2)} \qquad (4.1)$$

Let, $(Xi = T\char`^i\ /\ T\char`^i\ \ )$ be the relative allocation.

It is a normalized measure that ranges between zero and one. The maximum fairness is 100% and the minimum is 0%. This makes it intuitive to interpret and present. If all $Xi's$ are equal, the allocation is fair and the fairness index is one. The second well known fairness index was proposed by Jain et al. [149] and named as Jain's Fairness Index (JFI).

Furthermore, the performance of VoIP flows changes their condition frequently from an active to an inactive condition. The VPQ measures the fairness under these conditions in a very proficient way over WLANs. In simulation experiments, it will implement the VPQ fairness on the temporal fairness index and Jain's fairness index. It will also compare VPQ results with the results of other efficient and fair schedulers such as Contention-Aware

Temporally fair Scheduling (CATS) and Temporally-Weighted Fair Queueing (T-WFQ). It shall first present simulation model over WLANs [150], [151].

4.6 Throughput of Flows

In experiments, throughput is the amount of data correctly received by the end-user. It classify the VF and NVF traffic. The first (4) flows are from the VF traffic flows and the last (4) flows are from the NVF traffic flows. The total channel link sharing is 100 Mbps. The total simulation time in these experiments is 1200 seconds. At first it considered and evaluated the maximal realizable throughputs in the NS-2 with the simulation in T-WFQ and CATS. The scenario is a VPQ consisting of a QoS and only one VF station and the NVF from one to a number of stations.

The throughput is to measure the final state of the packet sent or dropped over the network. In the simulation, it can add up the number of packets, multiply by their size and divide by the time of the simulation to get the average throughput over the network. The VPQ will provide the long-term throughput that should be assured for the VF and NVF when the sufficient bandwidth is supplied over the network.

It took the following scenario to verify that VPQ model behaved correctly for different numbers of competing nodes: In the simulation scenario, the number of wireless stations was increased from 1 to 15. Each station sent two types of flows: a VF and NVF flow. For the following simulations, it chose an 802.11b physical layer with a basic rate of 1 Mbps and a data rate of 11Mbps. To guarantee the accuracy of simulation topology, they evaluated them with the work of T-WFQ, CATS and CAPS. WLAN networking technologies based on the IEEE 802.11 standards transmit data packets via the air. Each data packet is immediately acknowledged if it is received without errors. To avoid potential packet delay effects, in this simulation, the maximal number of retransmissions was set to zero.

4.7 Simulation Setup

In this section, the simulation setup is obtained with the NS-2 simulator tools. It will present a detailed simulation-based evaluation of the Voice Priority Queue (VPQ) scheduling system model and algorithms in comparison with others of the most related schedulers to VPQ work. It initially presents a concise description of the VPQ measure of fairness which captures the immediate performance of a traffic scheduler over WLANs. After then present a simulation using the NS-2 topology gateway traffic traces which compares the fairness characteristics of the VPQ based on this new metric with other efficient schedulers. In addition, they also compare the throughput of VPQ scheduler with most related schedulers already discussed in chapter two.

It will discuss the implementation details of WLANs using IEEE 802.11 standard. It performs extensive simulations in all three stages to show the effectiveness of the VPQ scheduling algorithm. They have selected the correct bandwidth model for all three stages of the VF and performance comparison with the most related traffic schedulers. It assumes that all the nodes have an Omni-directional antenna with available WLANs for the NS-2 simulation. Based on the above specifications, it send packets to the VF and NVF type of flows with different type of packets, frames, bytes and bit-rates.

In simulations, the effectiveness and efficiency of the proposed algorithms will be measured in terms of throughput, fairness index, packet delay and Packet Error Rate (PER). The comparison will show in simulation clearly in topologies and tables to evaluate the VPQ scheduler over IP-based networks.

The VPQ scheduling algorithm is provided as an essential technique in the VoIP communication networks to guarantee the QoS requirements. The design of the VPQ is managed by the limited bandwidth utilization and has been proven to have an efficient performance over WLANs. It will discuss the following features and principles that should be implemented in the simulation. The efficient bandwidth implementation is the most important in the VPQ design. It applied the VF and NVF techniques to channelize the traffic proficiently over the network.

The VPQ algorithm will assign available resources fairly across the VF and NVF connections over the network. In this manner, they are giving preference to the VF due to the higher priority due to delay sensitive traffic over WLANs. In this section, it will present the simulation analysis for VF flows. They have focused on two things, delay roundedness and maximum achievable throughput. The simulations have performed for both types of traffic i.e. the VF and NVF.

In chapter three, the 3rd stage of the VPQ scheduling model has discussed in detail. The switch mechanism provides the move to the VF when the NVF is in the empty condition. This switch mechanism provides more fairness in VF traffic over WLANs using IEEE 802.11 standards. This switch manages the bursty traffic flow at peak times. It provides a key technique introduced by VPQ scheduling model. The key technique shows the high priority and low priority queue management system over IP-based networks.

4.8 Link Sharing between the VF and NVF

The proposed VPQ algorithm automatically shares the link between the VF and NVF traffic over IP-based networks. For the VF flow, the Real-time Transport Protocol (RTP) for the VoIP traffic. For the NVF flow, it has a variety of traffic such as Transmission Control Protocol (TCP), User Datagram Protocol (UDP) and Constant Bit Rate (CBR) for the NVF traffic over WLAN networks.

The main advantage of the VPQ traffic scheduler is that it provides extra bandwidth to the VF flow through link sharing without affecting the NVF flow over WLANs in stages. Furthermore, this priority will not affect the fairness when the VF and NVF flows are in active conditions. The newly proposed VPQ mechanism provides the guaranteed link sharing between the VF and NVF flows.

4.9 Simulation Configuration of VPQ scheduler

In these simulations, the wireless nodes were connected with an AP in the WLAN. The AP was also connected to the wired nodes so that it could easily carry the VF and NVF traffic and hence no loss occurred on this link over the network. The simulations were configured in such a way that each node used bidirectional traffic flow. This was represented as VF traffic and transmitted using the fundamental VoIP protocol stack architecture to implement a VoIP network system over WLANs.

Voice packets are transmitted over IP-based networks. The VoIP is a real-time application and transmits the voice on the Real-Time Transport Protocol (RTP), User Datagram Protocol (UDP) and Internet Protocol (IP) (RTP/UDP/IP) over WLANs. Voice packets are small in size and they have RTP (12 bytes), UDP (8 bytes), and IP (20 bytes) headers. Lastly, the data-link layer Medium Access Control (MAC) has a (34 bytes) header. All of these headers calculate to a total of 74 bytes of overhead in the VoIP protocol.

It setup the start and stops time in seconds (sec) because the NS-2 supports time in seconds. It configures the channel types as set opt (chan) over the network. It set in the simulation the two-ray ground reflection method for wireless node communication over a network. This is a single Line-of-Sight (LoS) path between wired and wireless nodes that are the only means of propagation.

The benefit of the two-ray ground reflection method is that it assumes both a direct path and a ground reflection path over a network. The Two-ray round reflection method equation is as follows 4.2, 4.3 and the details are as shown in Table 4.3.

Table 4.3 List of important Two-ray Ground Reflection Notations

Notations	Description
P_r	It is consider as propagation.
d	It is consider as the distance from the transmitter over a network.
P_t	It is consider as a transmitted signal power over radio propagation.
G_t G_r	These two are the antenna gains of the transmitter and the receiver.
ht^2 hr^2	These are the heights of transmitter and receiver antennas.
L	It is consider with the free space model
W	It is consider as a wavelength.

$$P_r\,(d) = \frac{P_t\; G_t\; G_r\, ht^2\; hr^2}{d^{\,4} L} \qquad (4.2)$$

The free space propagation method communicates between the transmitter and receiver. The free space method works in a circle around the transmitter and if a receiver node is within the circle, it will receive all of the communication packets. Otherwise, it will lose all the packets over the networks.

$$P_r\,(d) = \frac{P_t\; G_t\; G_r\, W^{\,2}}{(4\pi)^{\,2} d^{\,2} L} \qquad (4.3)$$

The energy model represents the level of energy in a mobile host. The initial energy of nodes is set in the NS-2 using the command (intialEnergy). It also calculates the energy usage for every packet it transmits and receives. These are called (txPower_) and (rxPower_). The network interface option applies as (Phy/WirelessPhy) between a wired and wireless node over a network. IT configures (Mac/802.11) for wireless communication and for the interface queue type, it configure it as (PriQueue) with a Link Layer (LL) over the network. The Mac object simulates the medium access protocols that are essential in the shared medium situation such as the wireless, wired and local area network.

It has used the Antenna/OmniAntenna model for communication. Furthermore, it define a wired node as a (set num_wired_nodes), a wireless node as a (set num_wireless_node) and an AP or Base Station (BS) node as a (set num_bs_node). As it discussed earlier in this chapter, the NS-2 has two types of output, namely trace file (set tracefd [open VPQ-out.tr.w]) and Network Animator or NAM (setnamtrace [open VPQ-out.nam w]) in the simulation of the VPQ scheduler over the WLAN. After that, it configure the wired, wireless and base-station nodes over the network. It will set the base-station as (set BS (n) [$ns_node [Index $temp 0]]) in the simulation.

4.10 The Simulation Topology Scenario Description

The simulation of the Voice Priority Queue (VPQ) scheduler's scenario is the setting of two wired nodes connected with two Access Points (APs); these two APs classify the traffic into the VoIP-Flow (VF) and Non VoIP Flow (NVF). Next, it has the VF mobile node and NVF mobile nodes from queue one to a number of queues over WLANs using IEEE 802.11 standards.

In this section, it will present simulation analysis for the VF flows. It has focused us on two things, delay roundedness and maximum achievable throughput. It has performed the simulations in both types of traffic the VF and NVF modes. The details are as shown in Table 4.4.

Table 4.4 Selection of options in the NS-2

Opt	Configuration	Description
set opt(chan)	Channel/WirelessChannel	channel type
set opt(prop)	Propagation/TwoRayGround	radio-propagation model
set opt(netif)	Phy/WirelessPhy	network interface type
set opt(mac)	Mac/802_11	MAC type
set opt(ifq)	Queue/DropTail/PriQueue	interface queue type
set opt(ll)	LL	link layer type
set opt(ant)	Antenna/OmniAntenna	antenna model
set opt(ifqlen)	50	max packet in ifq
set opt(nn)	1 to …N (Number of nodes)	number of mobilenodes
set opt(cp)	cp	connection pattern file
set opt(sc)	./mobility/scene/scen-3-test	node movement file.
set opt(stop)	250	time to stop simulation
set num_wired_nodes	2	wired nodes
set num_bs_nodes	2	base station nodes
set num_wireless_nodes	1 to number (N)	wireless nodes

Figure 4.2 shows the details of VPQ simulation topology over the networks. It is based on backend nodes that are connected with a wired network and frontend nodes that are connected with two Access Points (APs) or a based-station (BS) over WLANs. BS nodes are similar to gateways between wired and wireless nodes and permit packets to be exchanged between two types of nodes. They are going to simulate a VPQ scheduler over WLANs. The VPQ topology consists of two wired nodes called node (Initialization of Traffic Flow) and

node (Classification of Traffic Flow). The node (Initialization of Traffic Flow) provides initialization of traffic flow over a wired node and the node (Classification of Traffic Flow) provides classification of traffic over a wired node. Again for simulation of the VPQ, it has two more nodes named the node (VoIP Flow AP) and the node (Non-VoIP Flow AP). The node (VoIP Flow AP) provides the VoIP Flow (VF) and the node (Non-VoIP Flow AP) provides the Non- VoIP Flow (NVF) over gateway nodes.

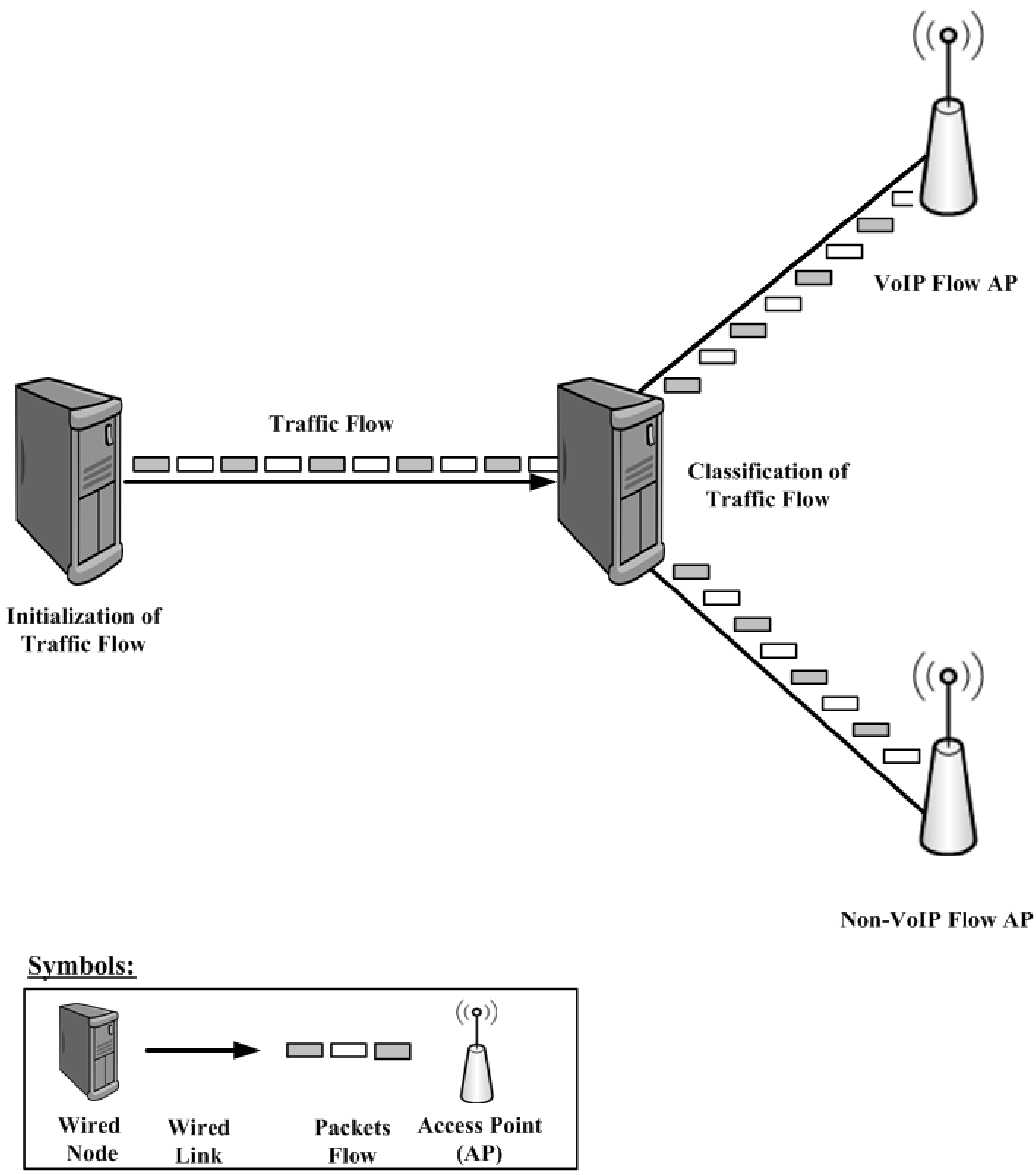


Figure 4.2 The VF and NVF Topology in the NS-2

In Figure 4.3, it has extended VPQ scenario up to wireless nodes. Meanwhile, the wireless nodes named the node (VF Queue), the node (NVF Queue # 1) and the node (NVF Queue # N) as a number of nodes over a wireless network. The node (VF Queue) is a mobile node that provides traffic flow between the VoIP Flow-AP (VF-AP) and the VoIP Flow-Queue (VF-Queue) over the WLAN. The node (NVF Queue # 1) and the node (NVF Queue # N) are considered as the NVF queue # 1 to NVF queue # N, number of flows over the network. It implemented the AP IEEE 802.11b mixed-mode technology.

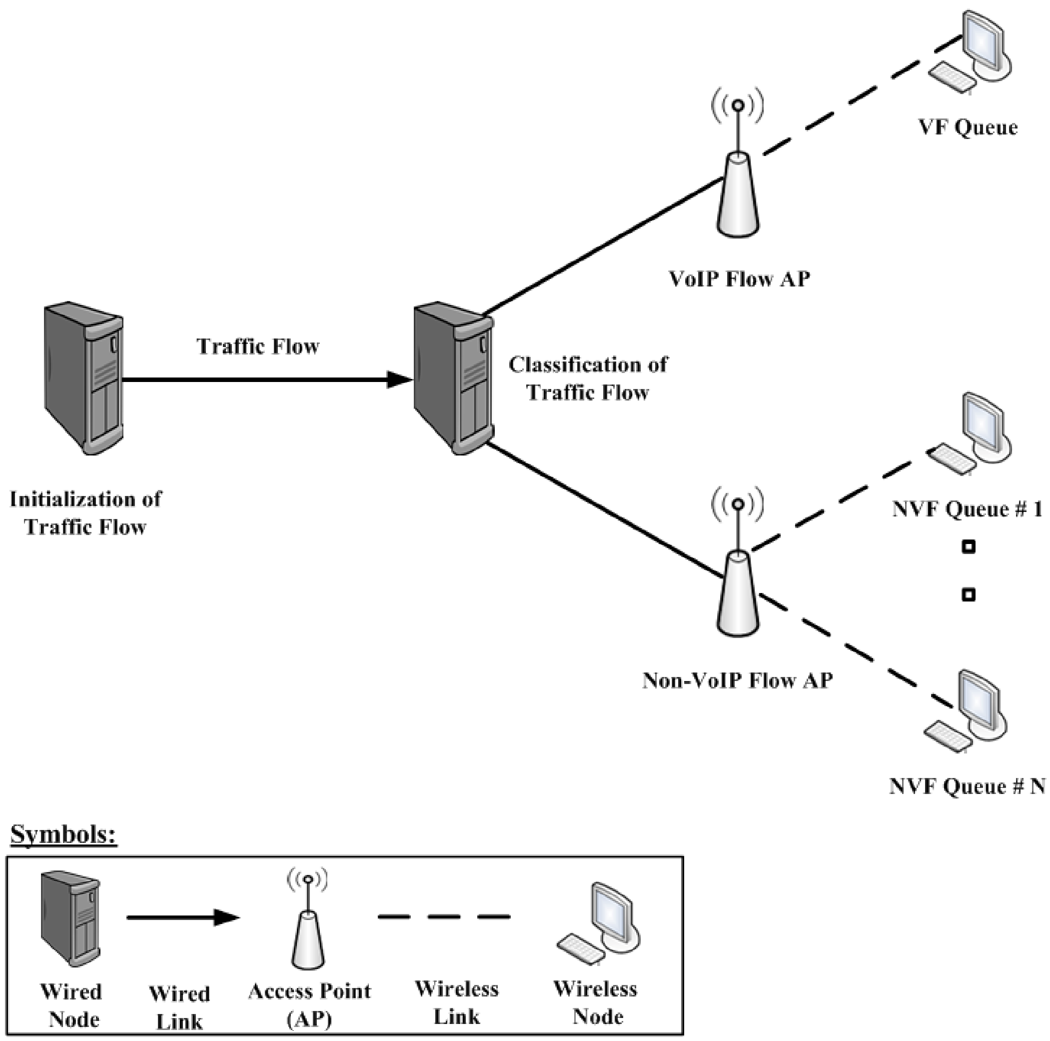
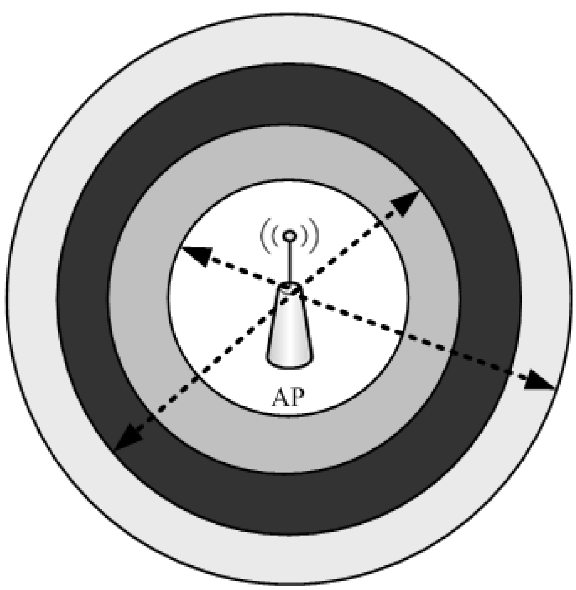


Figure 4.3 The VF and NVF Topology in the NS-2 over the WLAN

The mobile nodes move about within an area whose boundary is defined in the VPQ scheduler simulation as 500mX500m.

The node (NVF Queue # 1) starts out initially from the VF flow to the AP at a distance of around 50 meters and a data rate of 11Mbps. Then the node (NVF Queue # 1) moves far from the AP step by step starting at a distance of 70 meters and a data rate of 5.5 Mbps, a distant of 90 meters and a data of rate 2 Mbps, a distant of around 115 meters and a data of rate 1 Mbps from the AP over the WLAN. The VoIP connection is setup between the VoIP Flow-AP (VF-AP) and the VoIP Flow-Queue (VF-Queue).

After that, the Wired-VF and Wireless-VF start bidirectional communication with each other and packets are exchanged between the node VoIP Flow-AP (VF-AP) and the VoIP Flow-Queue (VF-Queue) as they reach within range of each other as it mentioned above over the WLAN using IEEE 802.11b standard. In simulation, it configure the IEEE 802.11 Omni-directional AP. Transmission ranges are based on data rate and distance. Furthermore, the details are expressed in Figure 4.4.



Transmission Ranges of Access Point (AP) IEEE 802.11b

Distance	Data Rate
50 Meters	11 Mbps
70 Meters	5.5 Mbps
90 Meters	2 Mbps
115 Meters	1 Mbps

Figure 4.4 IEEE 802.11b Transmission Ranges of Access Point

At the same time, the node (NVF Queue # 1) provides a link between the NVF-Wired (NVF-W) and the Wireless-NVF (W-NVF) over the WLANs. The node (NVF Queue # 1) and the node (NVF Queue # N) are consider as (NVF queue # 1) to NVF (queue # N), number of flows over the network. It implemented the AP IEEE 802.11b mixed-mode technology.

These nodes also communicate with each other in a bidirectional manner and packets are exchanged between the node (Non-VoIP Flow AP) as a Wired-NVF and the node (NVF Queue # 1), to number of nodes (NVF Queue # N) as they reach within range of each other as it mentioned above over the WLAN using IEEE 802.11/b standards.

VPQ initialized traffic server sends two types of flow, the VF and NVF. The VF gets 10Mbps and the NVF get the 90Mbps of traffic flow. The VF received flows in the shape of packet flows. The NVF received flows in multiple shapes based on the nature of the traffic like packets, frames and bytes from the initialized traffic server.
The flow weights are sent based on the average rate of each flow.

Furthermore, the bandwidth of the link between the nodes from the server to the VPQ MN is 10Mbps in the initial stage. Due to brusty traffic, the VF flow will increase the bandwidth and share with the NVF in an inactive condition. It also shares bandwidth with NVF due to an inactive bandwidth of NVF 90Mbps.

4.11 Experimental Setup of the VoIP Test-Bed

For the experimental setup measurements, it developed a Voice over IP (VoIP) test-bed. The test-bed is located in the VoIP Research & Development Lab (VR&DL), on the 2nd floor, building 2, Department of Computer and Information Sciences (CIS) at Universiti Teknologi Petronas (UTP). The test-bed is illustrated in Figure 4.5 and 4.6.

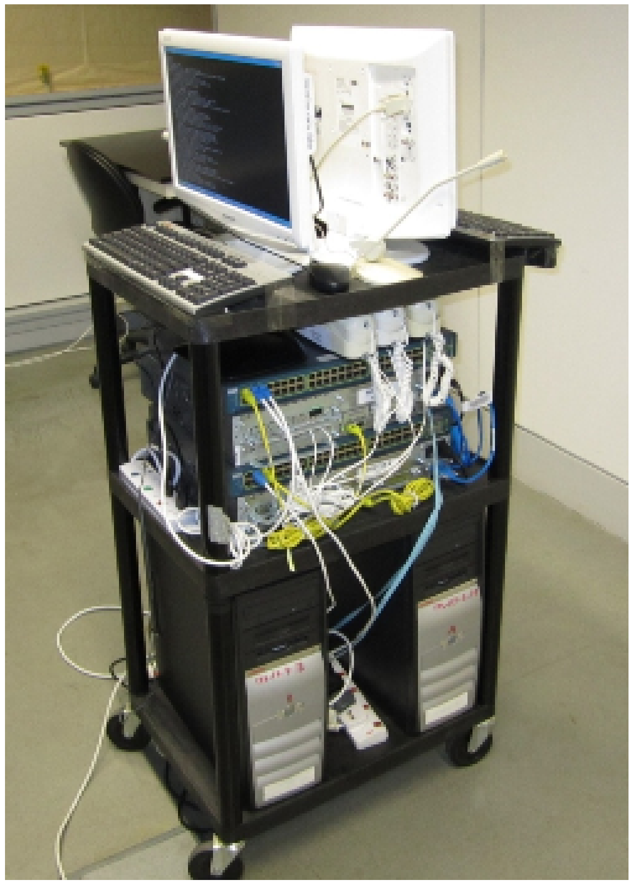


Figure 4.5 The VoIP Test-Bed in the CIS Department

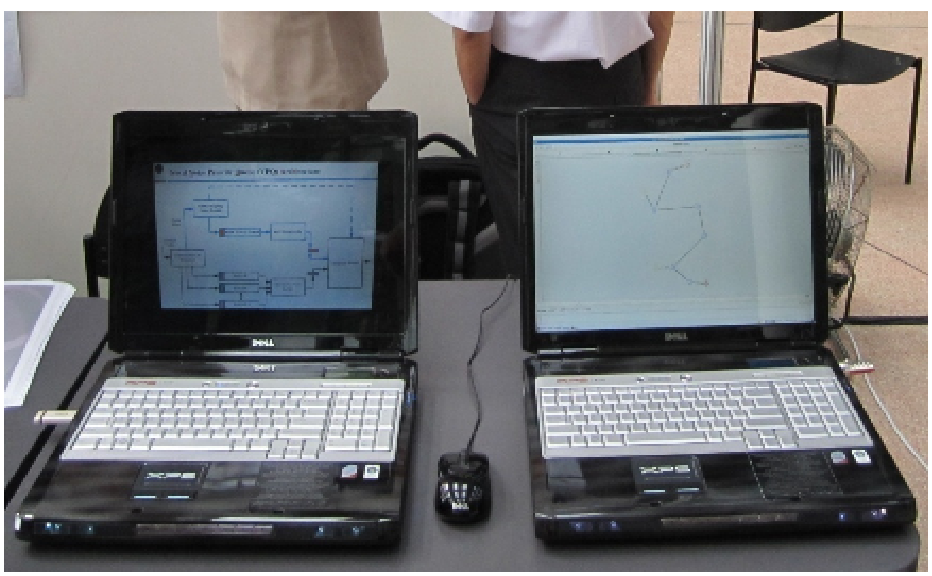


Figure 4.6 The VoIP Test-Bed's Mobile Nodes for WLANs

In the test-bed, two desktop computers were used as Server A (IP 192.168.0.1) and Client B (IP 172.17.7.7). The desktops were a 2.0 GHz Intel with 2 GB of RAM running Putty terminal for configure Routers A (IP 10.20.20.1), Router B (IP 10.10.10.1), Access Point (AP) IEEE 802.11a/b (IP 192.168.0.100) switches, Soft-Phones, Hard-Phones and a few laptops as a Mobile Node (MN) for the VoIP experimental environment. The details are as shown in Tables 4.5 and 4.6.

Table 4.5 The VoIP Test-Bed Equipment and Configuration Details

Equipment	Configuration Details
PC as Server A	IP 192.168.0.1
PC as Client B	IP 172.17.7.7
Routers A CISCO 3700 Series	IP 10.20.20.1
Routers B CISCO 3700 Series	IP 10.10.10.1
Access Point (AP) 802.11a/b	IP 192.168.0.100
Switches CISCO 3600 Series	Two Switches 48 Ports
Soft-Phones	SJ-Phone and Net-Meeting
Hard-Phones	5561001, 5561002, 5561003 & 5561004
A Few Dell Laptops	As Mobile Nodes (MN) for WLAN
Cables	Console, Serial, Fast Ethernet 0/0, 0/1, RJ11

Table 4.6 Device Details of the VoIP Test-Bed

Device	Quantity	Description
Desktop	2	Compaq inc. Processor: Intel core (TM)2 Duo CPU E 7300 @ 2.60 GHz, Memory 2 GB RAM
Laptop	2	Dell inc. Processor: Intel(R) Core (TM)2 Duo CPU T9300 @ 2.50 GHz, Memory 4 GB RAM, 32-bit Operating System, Service Pack 1.
Router	2	Cisco 3700 Series, Voice Network Module,
Switch	2	Cisco Catalyst 2950 series, 48 Ethernet
Access Point	2	Cisco IEEE 802.11 Mix-mode AP

Furthermore, the measurements for the VPQ scheduler on the test-bed were taken in the VR&DL. Experiments performed in outdoor and indoor environments. It performed VPQ experiments 50 times in a multiple of days for verification and validation. The main purpose to performed experiments multiple locations due to indoor, outdoor environments that why experiments 50 times. It also compared VPQ test-bed results with other related schedulers. The Server, Client and MN sent the VF and NVF traffic flows requests to each other for communication. One of the goals in developing the VoIP test-bed was to verify VPQ scheduler's performance in any infrastructural environment. It classifies the VF and NVF traffic flow over the test-bed and the VoIP test-bed equipment and configurations.

In this section, they are doing experimental setup as the VoIP test-bed that shows how well the VPQ manages over a real WLAN. For these setups, it applied the IEEE 802.11b mixed-mode standard and related parameters. The main reason for this is to get the maximum bit-rate of 11Mbps when combined with a test-bed. All the experiments were run for at least 1200 seconds (sec) in order to achieve average experimental results.

Throughput is the most obvious measure of performance in the network. The VoIP has a clear throughput requirement of 32Kbps and if this is not being met then the performance will definitely suffer. It should mention that due to the bursty nature of the traffic, the throughput requirement is actually 64Kbps while the station is transmitting and 0Kbps otherwise.

4.12 Summary

In this chapter explained the NS-2 simulation and test-bed experimental setup to evaluate the Voice Priority Queue (VPQ) scheduler over WLANs. The evaluation of the VoIP-Flow (VF) and Non-VoIP-Flow (NVF) measured via simulation applying the NS-2 simulator. The NS-2 is a license free network simulations tool and is known to have more realistic topologies than other tools. It also made a comparison and evaluation between different scheduling algorithms. It has defined parameters like, fairness index and throughput to evaluate the VPQ technique for the VF and NVF traffic flow over WLANs.

In the following chapter will present the results, discussion and findings. Therefore, in the next chapter different among different scheduling algorithms will be compared and evaluated.

CHAPTER 5

RESULTS, DISCUSSIONS AND FINDINGS

5.1 Introduction

This chapter presents achieved results which are based on the various scenarios of topologies performed in the Network Simulation-2 (NS-2). In the previous chapter, they have implemented all stages of the Voice Priority Queue (VPQ) using the NS-2 simulation and test-bed experimental setup. They have described the topologies in the simulation. They have compared the VPQ with Contention-Aware Temporally fair Scheduling (CATS) [120], Decentralized-CATS (D-CATS) [120], Decentralized-CATS+ (D-CATS+) [120], Temporally-Weighted Fair Queueing (T-WFQ) [120] and Controlled Access Phase Scheduling (CAPS) [112], [113] traffic schedulers. Furthermore, they have also compared the VPQ test-bed results with the other schedulers. They will discuss in detail these results in the following sections.

5.2 Performance Parameters of the VPQ Evaluation, Validation, and Verification

The following parameters are used to evaluate the performance of the VPQ through the detailed simulation in the NS-2. They have applied a few parameters to verify and validate the VPQ with related schedulers and algorithms. Throughput is an important parameter to evaluate the research work in the IP-based networks. Throughput can be measured in Kbps and Mbps for the VoIP over WLANs.

End-to-end delay is a term of time in second (sec) or milliseconds (ms) needed for each transmitted packet to arrive at its destination. In the VoIP, the maximum end-to-end delay is based on less than 150 ms [8]. The Fairness indexes which they used are the Gini fairness index [93], temporally fairness index [120] and Jani's fairness index [148] to evaluate research work. The Fairness index is based on 0 to 1 digits and from 0 up to what it can consider better fairness of networks. The details are as shown in Table 5.1.

Table 5.1 Performance Parameters of the VPQ

Parameters	Description
Throughput	Throughput is the amount of traffic correctly received by the end user.
End-to-end Delay	End-to-end Delay is the time needed for each transmitted packet to reach its destination.
Fairness Index	Fairness can be calculated in fairness of the time share 0 to 1 digits. The highest value of the fairness index, that means good fairness of traffic.
Bandwidth	The amount of data that can be carried from sender to receiver in a given time normally in bit per second (bps).

5.3 Simulation Results According to the Mobility of Mobile Station

The VPQ results were plotted according to the flow 1 to flow 4, the distance (50 meters, 70 meters, 90 meters, and 115 meters), it see that the transmission ranges approximately from each data rate (11 Mbps, 5.5 Mbps, 2 Mbps and 1 Mbps). This was according to a commercial data sheet. Furthermore, they implemented all the stages of the VPQ and the details are as shown in Table 5.2.

Table 5.2 Performance metrics of VPQ

Flow	Distance	Data Rate	VPQ Stage
Flow 1	50 Meters	11 Mbps	First Stage
Flow 2	70 Meters	5.5 Mbps	Second Stage
Flow 3	90 Meters	2 Mbps	Third Stage
Flow 4	115 Meters	1 Mbps	Final Stage

Figure 5.1 shows the throughput of the VoIP-Flow (VF) over WLANs using IEEE 802.11 standards. Flow 1 shows the throughputs (Kbps) and time (sec) during the NS-2 simulation. The graph shows the results of the proposed VPQ along with T-WFQ and CATS scheduling algorithms over IP-based networks.

Flow 1 is based on the data rate of 11 Mbps and a distance of around 50 meters. The time period of the algorithms starts from 1 sec and ends at 1200 sec. The throughput of algorithms starts from 1 Kbps and ends at 2000 Kbps over WLANs. As shown in Figure, the CATS scheduling algorithm throughput starts at 700 Kbps and continues until the end at 700 Kbps. The T-WFQ algorithm throughput, starts from 1000 Kbps until 400 sec, after that it moves up gradually from 1100 Kbps to 1200 Kbps. It can notice that the throughput of T-WFQ is better than the CATS algorithm.

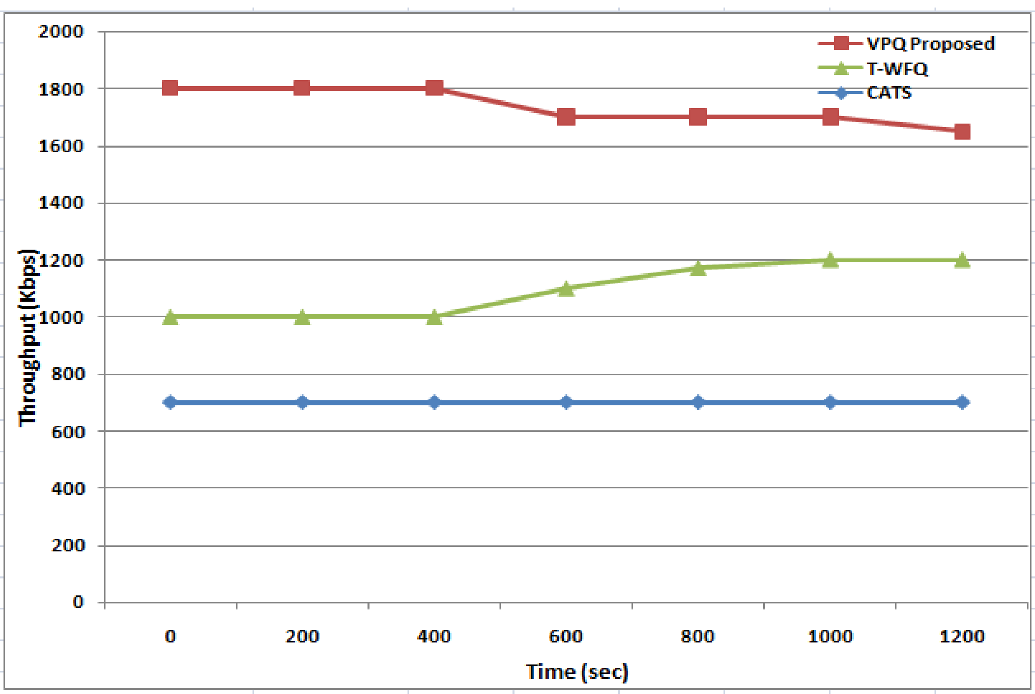


Figure 5.1 Throughput of Flow 1 per Flow when using the VPQ in Access Point

Now, it can see in Figure 5.1, the VPQ proposed scheduler's throughput starts from 1800 Kbps and it remains consistent until 400 sec then the throughput gradually decreases to 1700

Kbps and then to 1650 Kbps. Overall, when compared with CATS and T-WFQ in flow 1, the performance of the VPQ is better than the related schedulers.

Figure 5.2 shows the throughput Kbps of flow 2 and the time sec of the VPQ (proposed), T-WFQ and CATS during the simulation. Flow 2 is based on the data rate of 5.5 Mbps and a distance of around 70 meters as shown in chapter 4. Like flow 1, the time period of the algorithms start form 1 sec and end at 1200 sec. Also, the throughput of the algorithms starts form 1 Kbps and ends at 1400 Kbps.

It can see in Figure 5.2, CATS and T-WFQ algorithms' throughput remains between 500 Kbps to 600 Kbps. The T-WFQ throughput falls and rises with CATS and it reaches to 600 Kbps at 1200 sec. The CATS throughput remains consistent throughout the experiments.

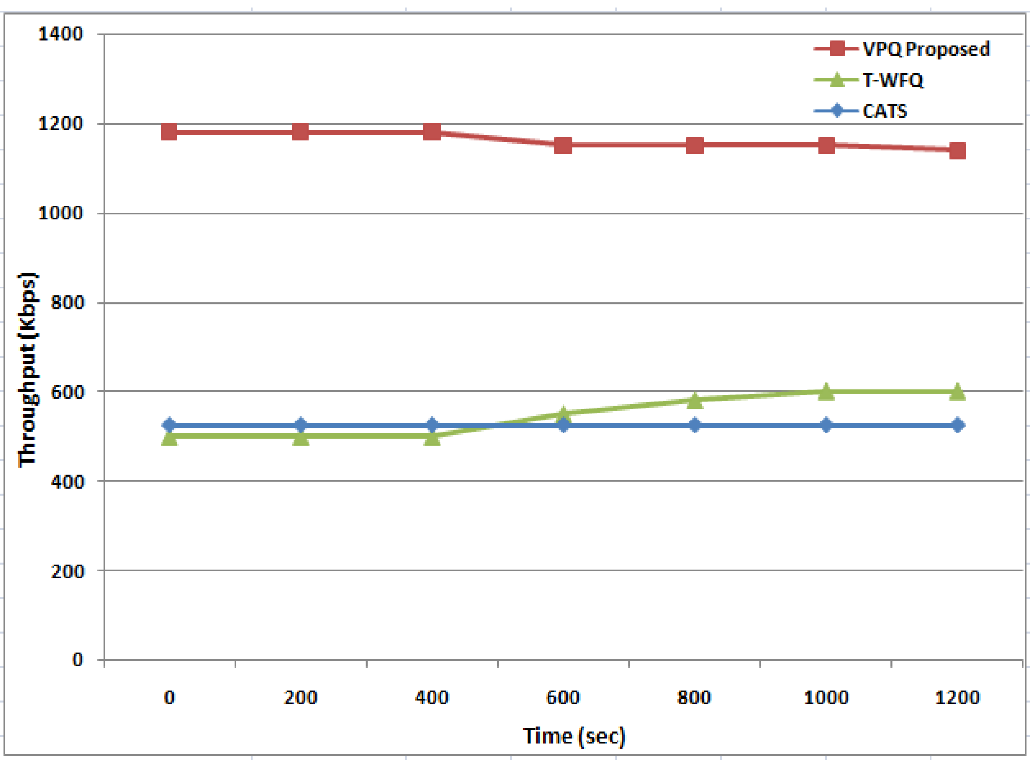


Figure 5.2 Throughput of Flow 2 per Flow when using the VPQ in Access Point

As shown in Figure 5.2, the proposed VPQ scheduler's throughput starts from 1180 Kbps until 400 sec. Then the throughput slightly decreases to 1150 Kbps until 1000 sec and goes down to 1140 Kbps until it ends. It can see the throughput of the VPQ almost doubles compared to CATS and T-WFQ.

Figure 5.3 shows the throughput Kbps of flow 3 of all algorithms over IP-based networks. Flow 3 is based on the data rate of 2 Mbps and a distance of around 90 meters. It notice that the throughput of CATS is better than T-WFQ. The CATS throughput is 280 Kbps and T-WFQ starts from 190 Kbps and it remains consistent until 600 Kbps. Gradually, T-WFQ throughput increased and its' reached to 220 Kbps.

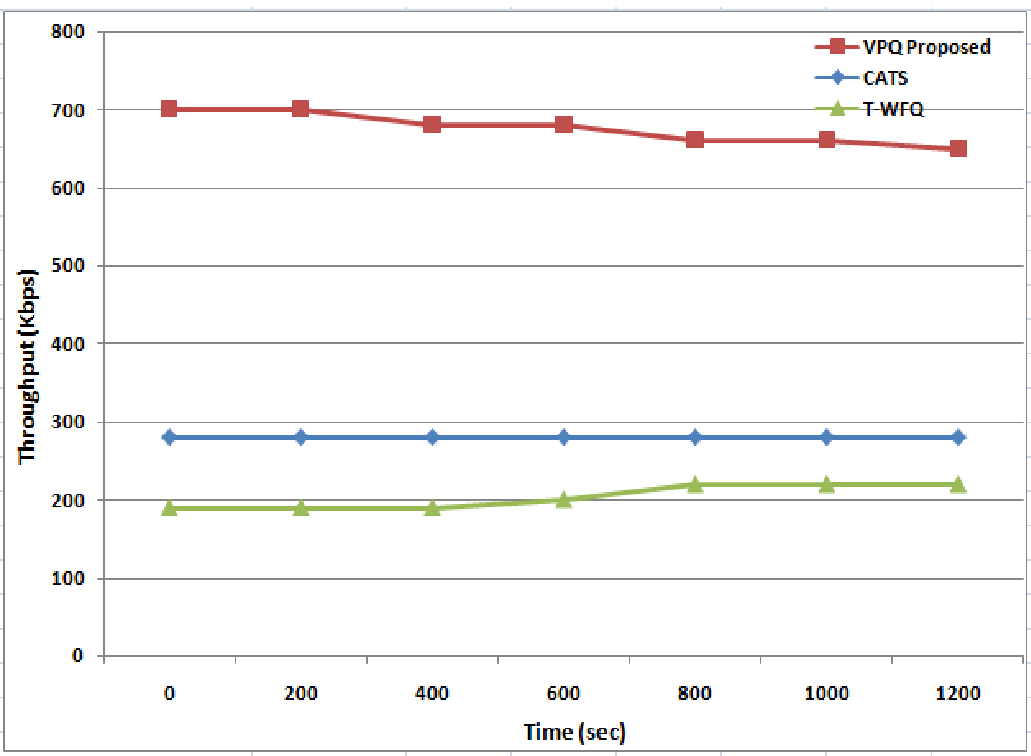


Figure 5.3 Throughput of Flow 3 per Flow when using the VPQ in Access Point

As shown in Figure 5.3, the proposed VPQ scheduler's throughput starts from 700 Kbps and ends at 650 Kbps. The graph clearly indicates that the throughput of the VPQ is better than CATS and T-WFQ.

Figure 5.4 shows the throughput Kbps of flow 4 of all the algorithms over WLANs. Flow 4 is based on the data rate of 1 Mbps and a distance of around 115 meters. It notices that the throughput of CATS has remained stable throughout the simulation. The CATS throughput has 180 Kbps which is better than the T-WFQ algorithm. The throughput of the T-WFQ algorithm starts from 100 Kbps and suddenly reaches to 130 Kbps in the simulation.

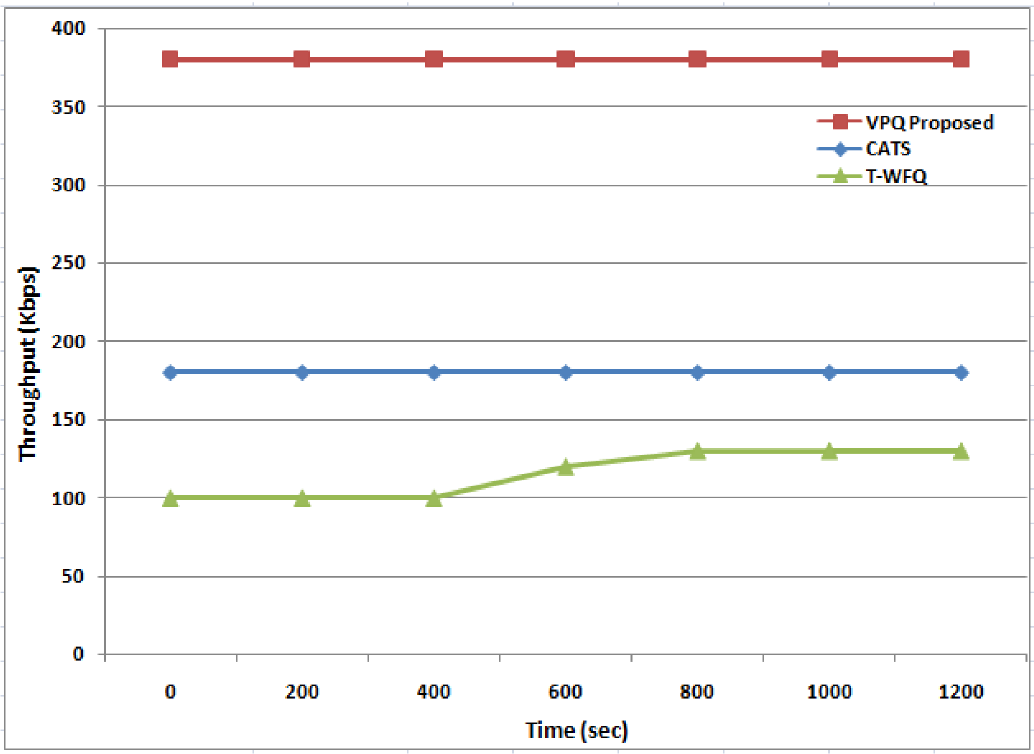


Figure 5.4 Throughput of Flow 4 per Flow when using the VPQ in Access Point

As shown in Figure 5.4, the proposed VPQ scheduler's throughput starts from 380 Kbps and ends with the same throughput. It can see from the above graph that in flow 4, the VPQ provided the best results like in previous graphs.

Figure 5.5 shows the total throughput (Mbps) of the proposed VPQ as well as the CATS and T-WFQ algorithms over the WLAN in the simulation. It compares all 4 previous flows to find the difference in the throughput measurement. In all of the pervious flows, it measured in Kbps and in the total throughput; it measured in Mbps to evaluate the throughput results.

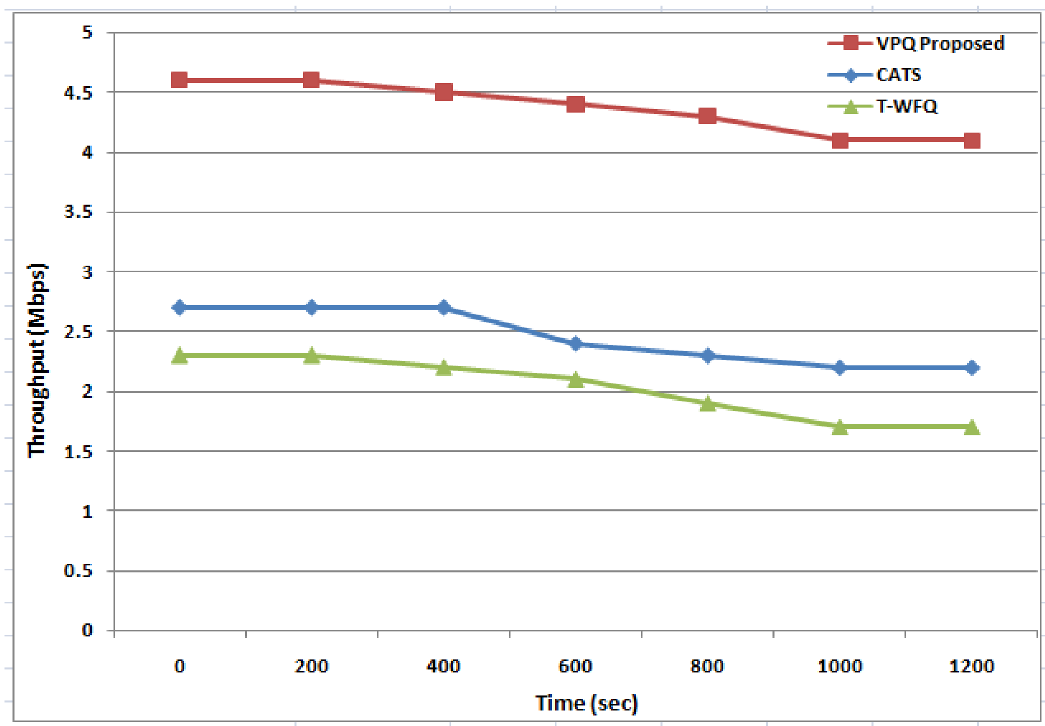


Figure 5.5 Total Throughput According to the Mobility of Mobile Station

The T-WFQ algorithm shows the lowest throughput among all the algorithms and the T-WFQ throughput starts from 2.3 Mbps and it reaches to 1.7 Mbps at 1200 sec. CATS has a higher throughput than the T-WFQ algorithm due to a better performance over the WLAN. The CATS throughput starts from 2.7 Mbps and gradually decreases to 2.2 Mbps.

Proposed VPQ algorithm is better than both algorithms over WLANs and the VPQ has a higher throughput due to classifying the traffic into the VF and NVF traffic, a high data rate flow and transmitting more packets. The proposed VPQ starts the throughput at 4.6 Mbps and ends at 4.1 Mbps.

Figure 5.6 shows the fairness index according to the mobility of the mobile station. The fairness is measured from 0 to 1 and above 0 is considered better fair indexing over the network. It also compares the proposed VPQ with the CATS and T-WFQ algorithms. It notices that the T-WFQ algorithm starts its fairness from 0.88 and increases to 0.9 at 600 sec then it decreases and at 1200 sec, its fairness index is 0.86.

The performance graph indicates that the VPQ and CATS maintain a better fairness index in the simulation. The CATS fairness index gradually decreases from 600 sec to a .99 fairness index.

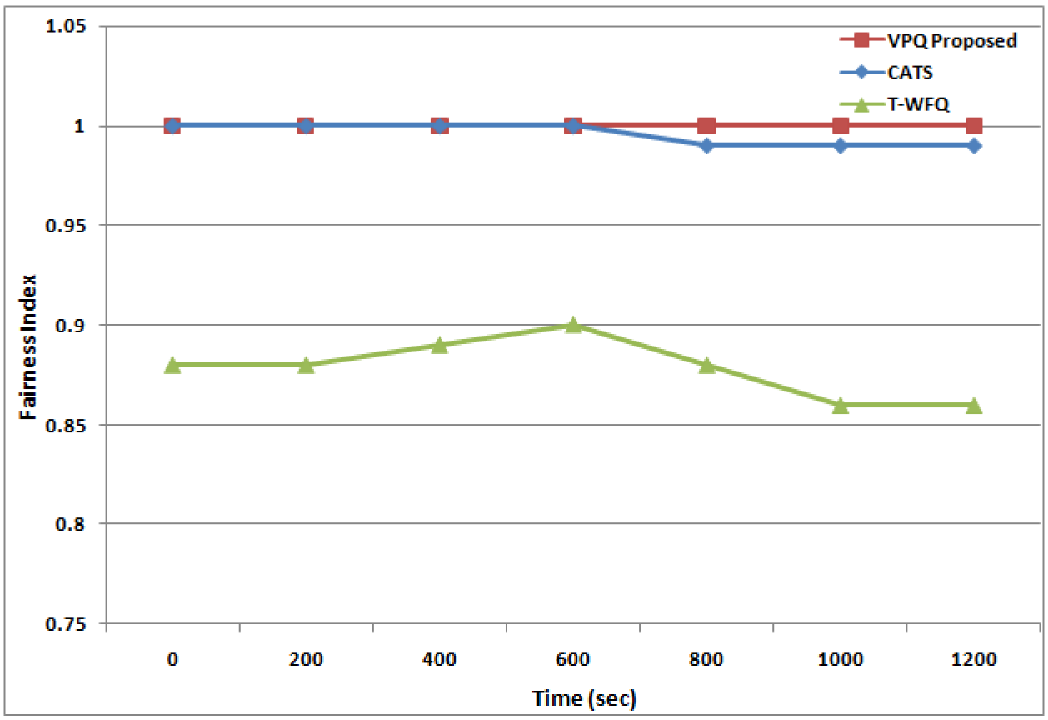


Figure 5.6 Fairness Index according to the mobility of the mobile station

5.4 Simulation Results based on Packet Size

In the Voice-Flow (VF), the packet size is an important metric of IP-based networks. The VoIP traffic needs small sized packets due to the low weight of traffic after text messages. Another, technical matter related to the small size of packets is that if the flow needs to retransmit the VF flow then it only loses less packets as compared with the big sized packets and frame flow over IP-based networks.

Figure 5.7 shows the throughput in Kbps for flow 1 of all algorithms over WLANs. All these measurements are based on the traffic flow of packet size. The T-WFQ throughput starts from 575 Kbps and it continues until 400 sec. After that, T-WFQ significantly decreases until 120 Kbps at 1200 sec. Furthermore, it notices that T-WFQ has the lowest throughput as compared with CATS and proposed VPQ algorithms.

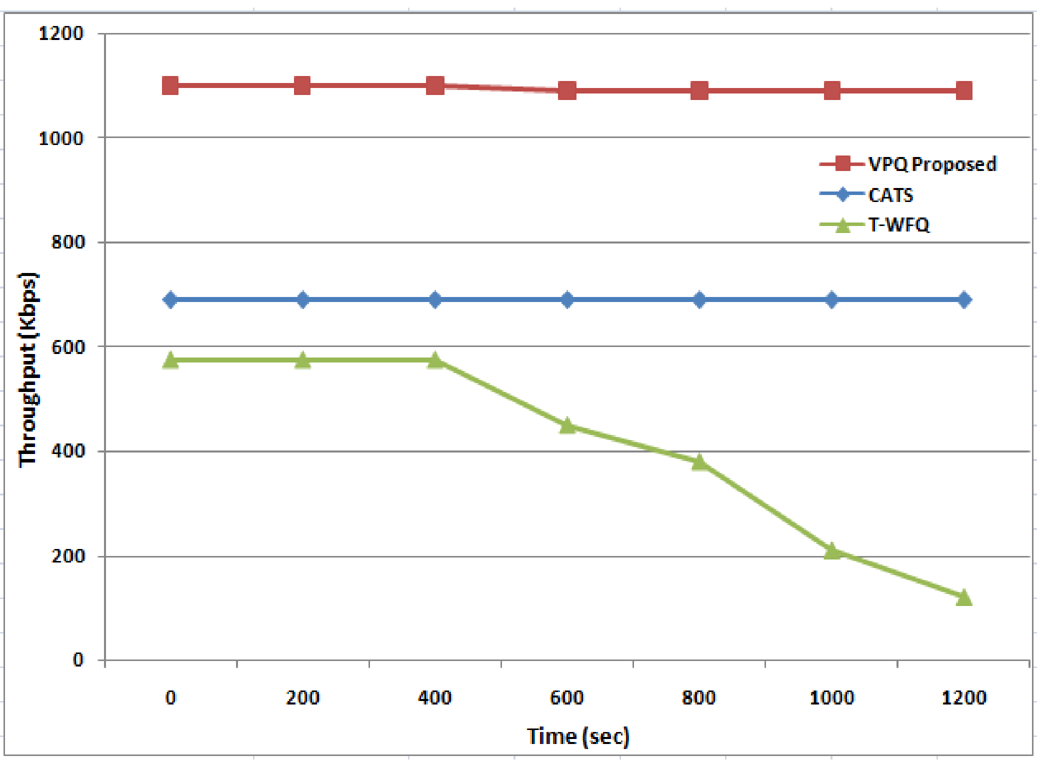


Figure 5.7 Throughput of per Flow 1 when using an Access Point

Meanwhile, the CATS algorithm shows a better throughput as compared to the T-WFQ throughput in the simulation. Proposed VPQ provides the best results due to an efficient technique over WLANs. The VPQ provides a higher throughput than the other algorithms at 1100 Kbps at 1200 sec in the simulation.

Figure 5.8 shows the throughput of flow 2 of all the algorithms over IP-based networks. It notices that the VPQ has the best throughput between 1100 Kbps and 1080 Kbps. Whereas, CATS maintains the throughput of 520 Kbps throughout the simulation over the network. The T-WFQ has the worst case throughput that starts from 490 Kbps and decreases to 130 Kbps at 1200 sec over the topology of the WLAN.

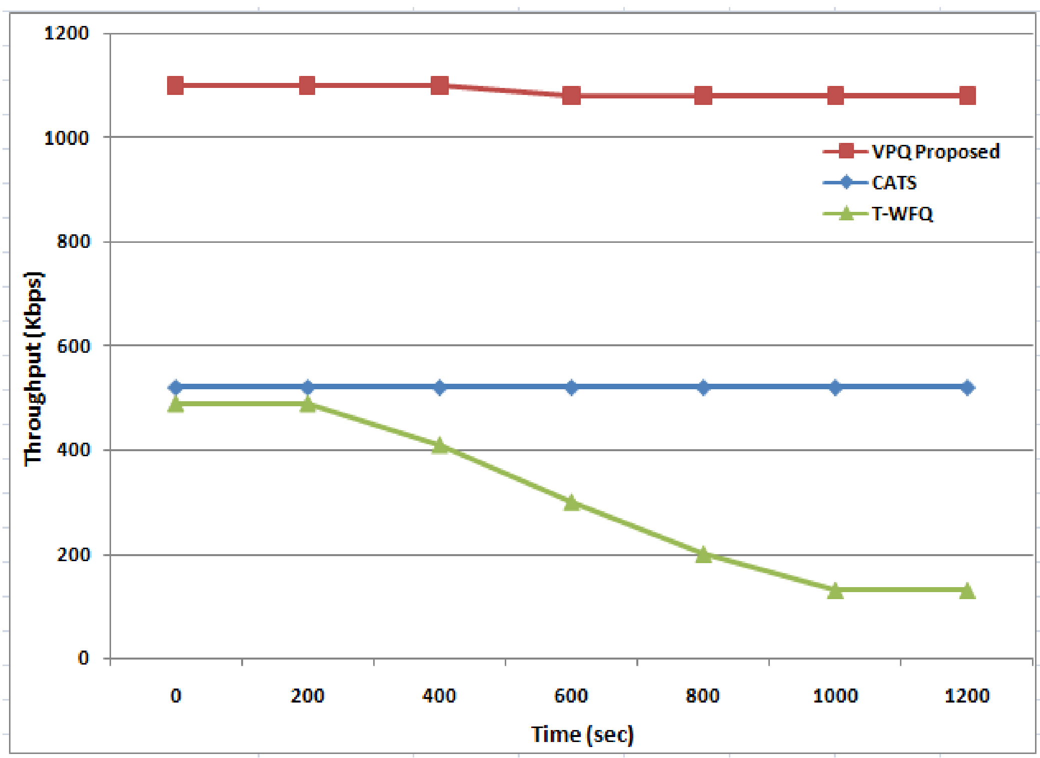


Figure 5.8 Throughput of per Flow 2 when using an Access Point

Figure 5.9 also shows the throughput of flow 3 over IP-based networks. Proposed VPQ scheduler and algorithms has the best-case throughput that consistently remains around 550 Kbps and the throughput is stable in the simulation.

CATS is also stable at 280 Kbps at 1 to 1200 sec over the network. Furthermore, T-WFQ has the lowest throughput and it starts from 180 Kbps and step by step decreases to 50 Kbps.

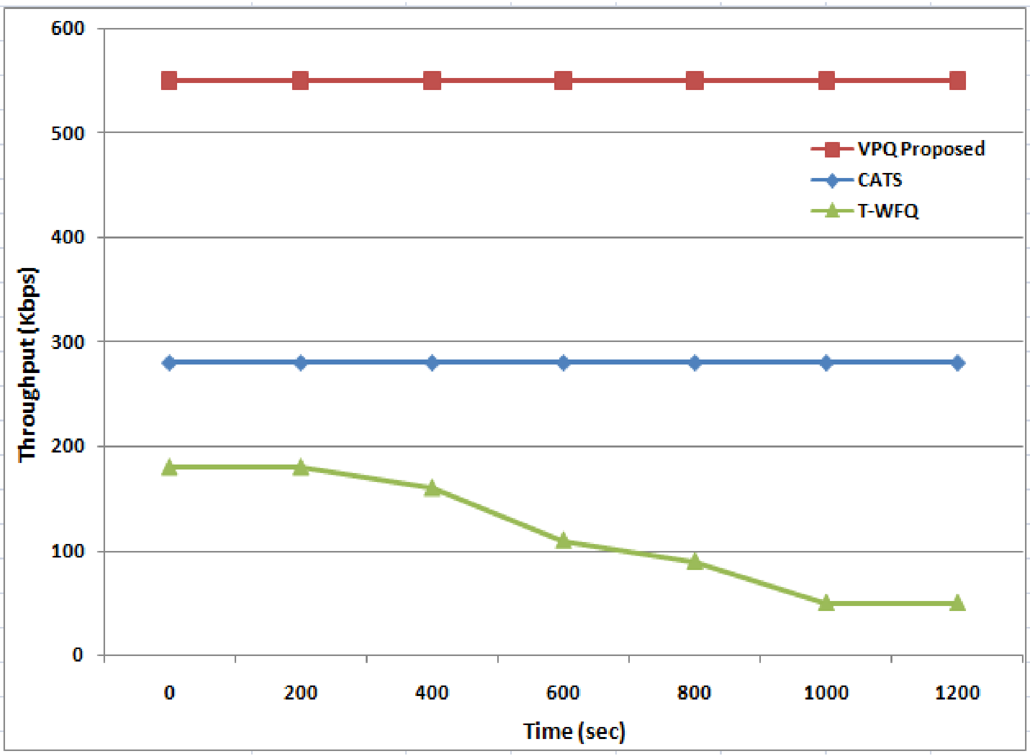


Figure 5.9 Throughput of per Flow 3 when using an Access Point

It notices that the T-WFQ throughput is the very worst in the limited packet sizes flow simulation. It starts from 180 Kbps while the VPQ throughout provides the throughput of 550 Kbps.

Figure 5.10 shows the throughput of flow 4 of all algorithms over IP-based networks. It notices that the VPQ has the best throughput of 380 Kbps and CATS has the throughput of around 180 Kbps throughout the simulation. T-WFQ has the worst case throughput that starts from 100 Kbps and gradually decreases to 30 Kbps at 1200 sec over topology of the WLAN.

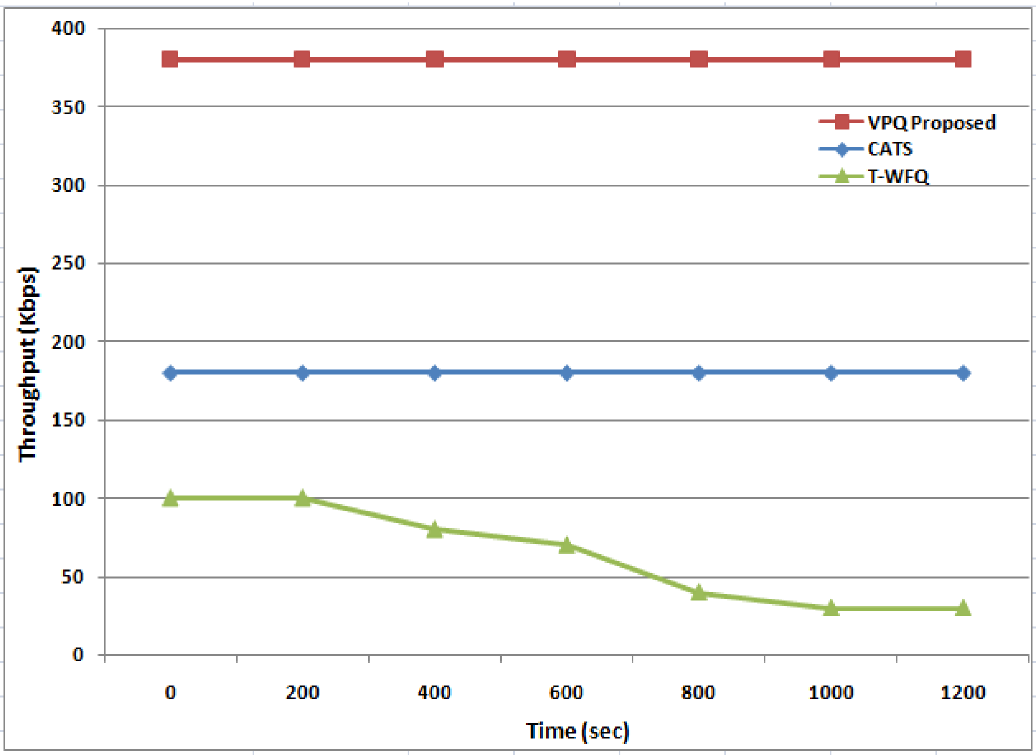


Figure 5.10 Throughput of per Flow 4 when using an Access Point

Figure 5.11 shows the throughput of all the flows over IP-based networks. In this Figure, it notices the total throughput Mbps of the proposed VPQ as well as the CATS and T-WFQ algorithms over the WLAN in the simulation. If they compare all of the 4 previous flows they can see the difference in the throughput measurement. Figure 5.11shows the throughput measured in Mbps used to evaluate the results.

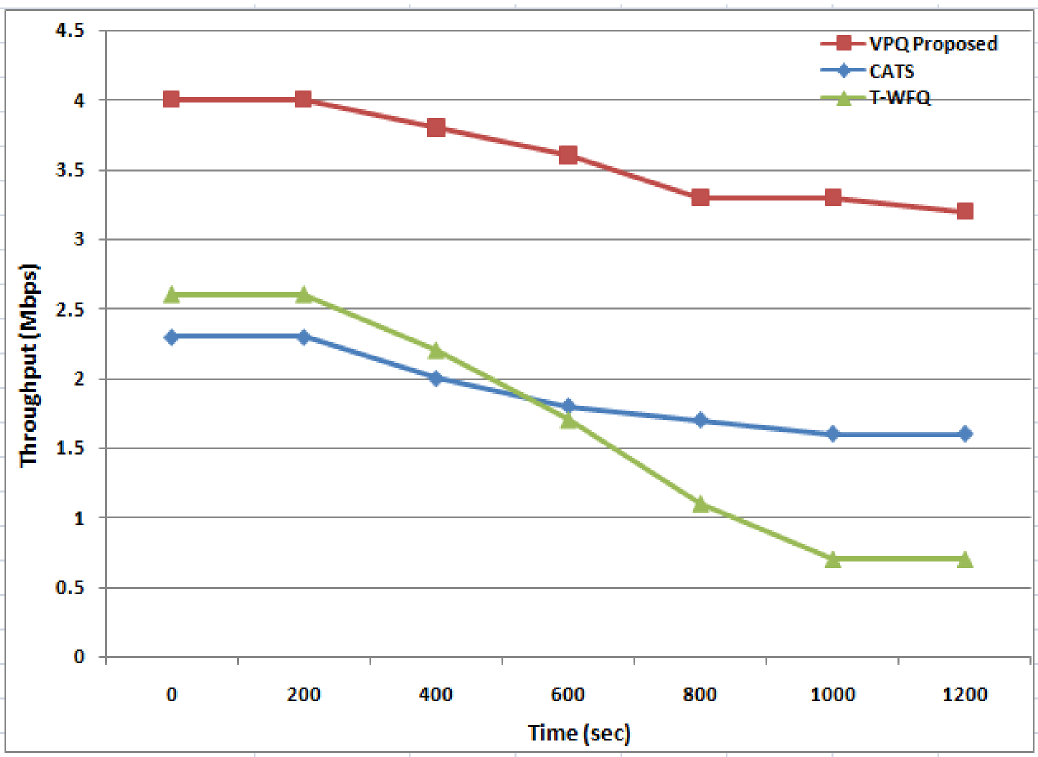


Figure 5.11 Total Throughput According to the Packet Size of Flow

The T-WFQ algorithm shows the lowest throughput that starts from 2.6 Mbps and is reduced to 0.7 Mbps at 1200 sec. CATS has a better performance than the T-WFQ algorithm due to a better performance over the WLAN.

The CATS throughput starts from 2.3 Mbps and gradually decreases to 1.6 Mbps. Proposed VPQ algorithm performs better than both the other algorithms over WLANs. The obvious reason is that the VPQ classifies the traffic into the VF and NVF. Moreover, it provides a high data rate flow that carries more packets. The proposed VPQ throughput performance starts from 4 Mbps and decreases to 3.2 Mbps at 1200 sec.

Figure 5.12 shows the fairness index according to the packet size of the flow. The fairness measured from 0 to 1 and above 0 is considered best. They also compare the proposed VPQ with the CATS and T-WFQ algorithms. They notice that the T-WFQ algorithm starts its fairness from 0.88 and reduces it to 0.3 at 1200 sec.

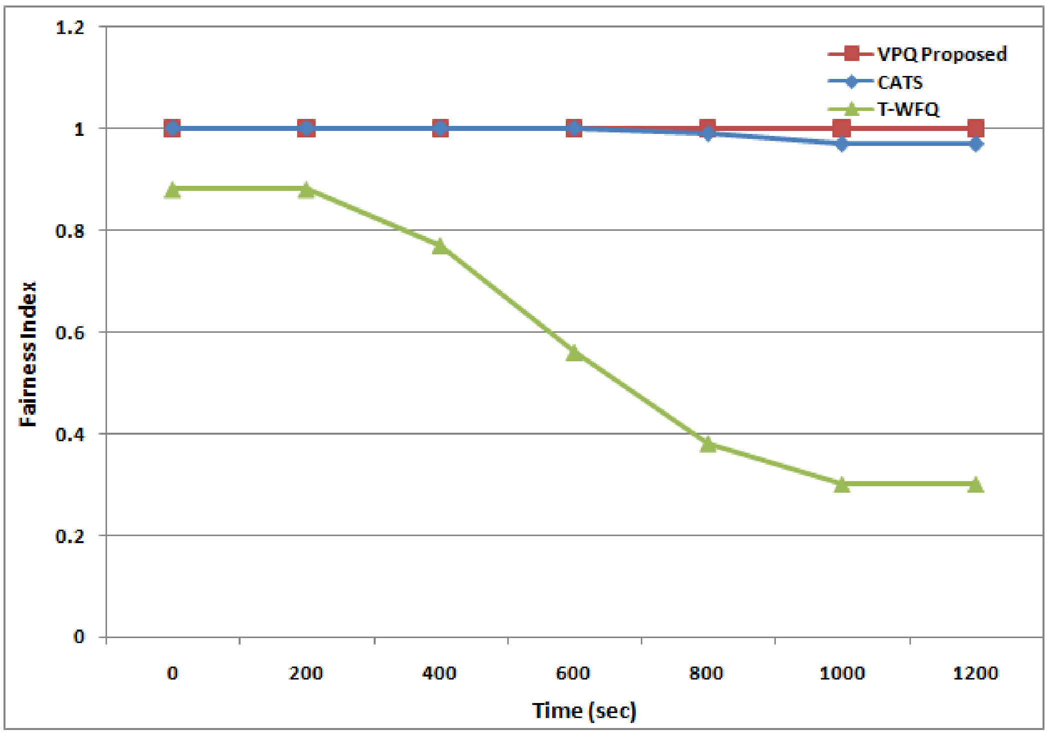


Figure 5.12 Fairness Index According to the Packet Size of Flow

Figure 5.12 clearly indicates that the performance of the VPQ is better than CATS in the simulation and both the algorithms outperform the T-WFQ algorithm.

5.5 Simulation Results based on Packets Delay

Figure 5.13 shows the end-to-end packet delay ms over the VF flow on IP-based networks. In this Figure, the flow 1 packet delay of the proposed VPQ as well as the CATS and T-WFQ algorithms are simulated over the WLAN. All these algorithms meet the basic criteria of end-to-end delay because normally, the VoIP needs less than 150 ms [8] packet delay over a WLAN. Furthermore, the VoIP traffic is delay sensitive which is why it requires the best performance in the Real-Time Transmission Protocol (RTP).

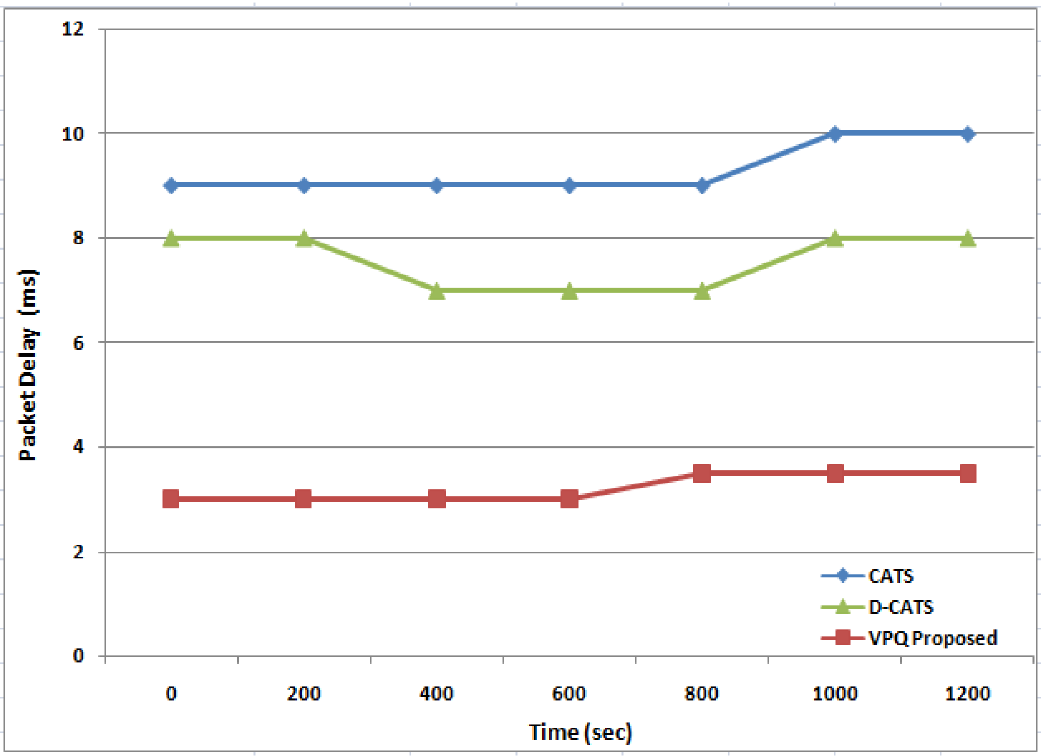


Figure 5.13 Per Flow 1 Packet delay

In Figure 5.13, they compared the performance of the VPQ with the upgraded CATS algorithm i.e. D-CATS. The performance of CATS in terms of delay starts from 9 ms and increases to 10 ms; whereas, the packet delay of D-CATS, starts from 8 ms and remains the same at 8 ms. It noticed that D-CATS performed better than the CATS algorithms.

The Figure shows that the VPQ algorithm performs better than the CATS and D-CATS algorithms in terms of packet delay. The VPQ packet delay starts from 3 ms and slightly increases to 3.5 ms over WLANs.

In Figure 5.14, it illustrated the flow 2 packet delay of the proposed VPQ as well as the CATS and D-CATS algorithms over the WLAN. The packet delay of the CATS algorithm starts from 11 ms and ended with 10.5 ms; whereas, the packet delay of D-CATS starts from 10 ms and slightly reduces to 9.5 ms. It notice that D-CATS performs better than the CATS algorithms.

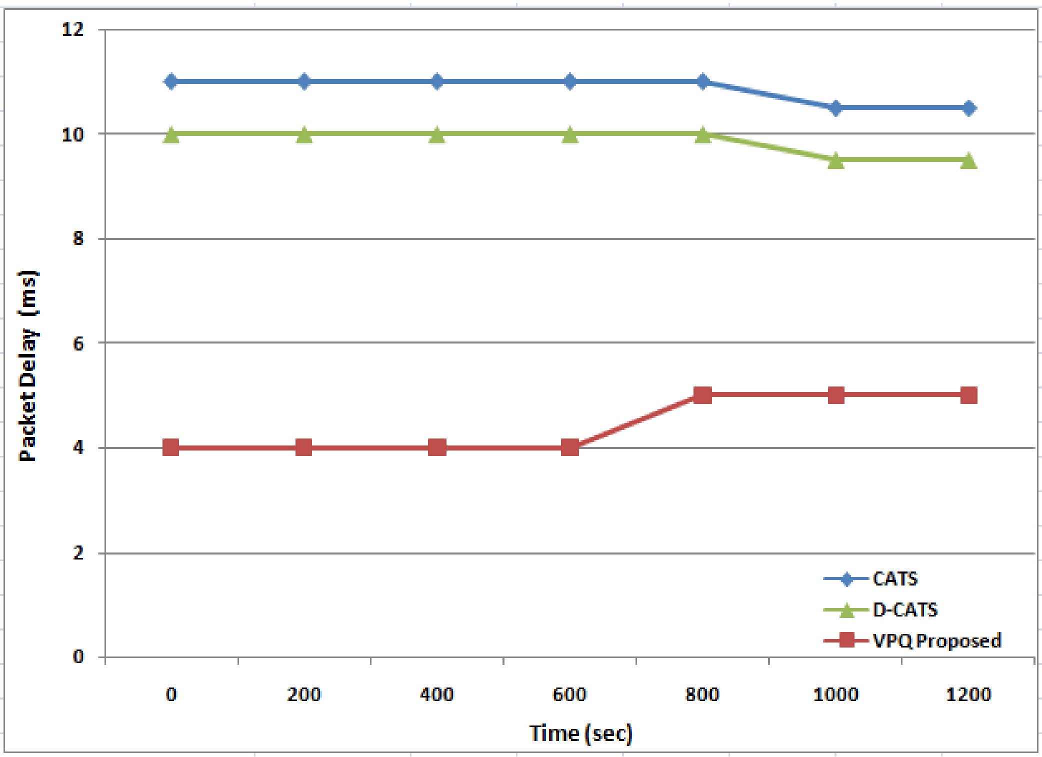


Figure 5.14 Per Flow 2 Packet delay

As they can see, the packet delay in the VPQ algorithm performed very well as compared with the CATS and D-CATS algorithms. the VPQ packet delay starts from 4 ms and slightly increases to 5 ms over WLANs.

Figure 5.15 depicts the flow 3 packet delay of the proposed VPQ as well as the CATS and D-CATS algorithms over the WLAN. The packet delay of the CATS algorithm starts from 15 ms and ends up at 17 ms; whereas, the packet delay of D-CATS starts from 14 ms and slightly decreases to 13 ms. They notice that D-CATS perform better than the CATS algorithm.

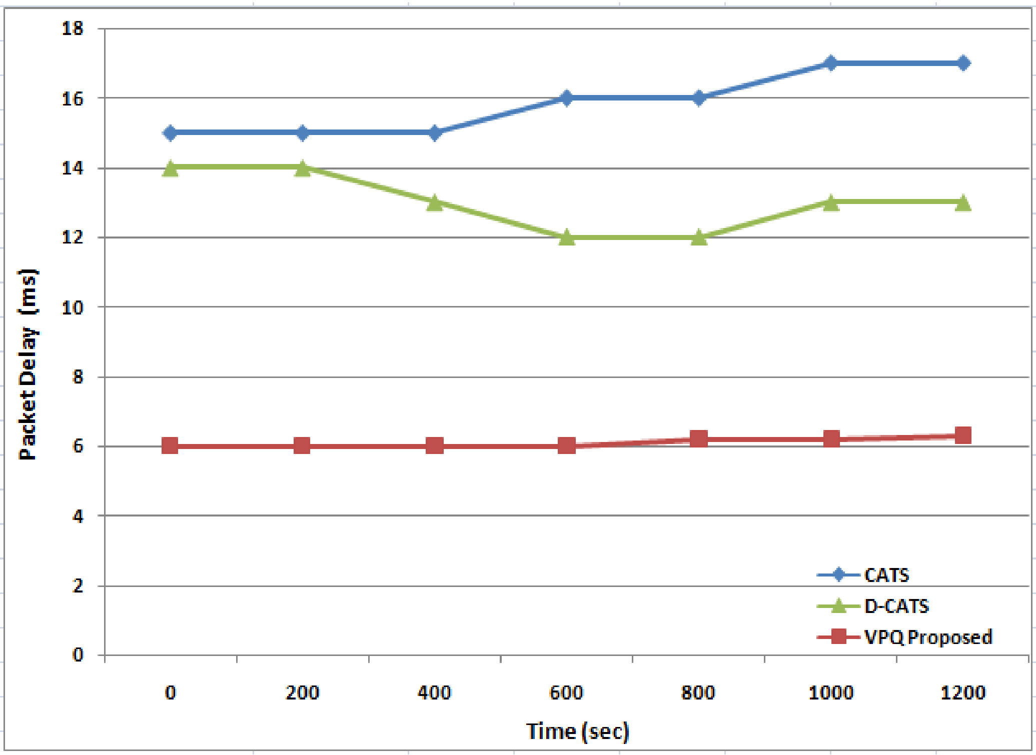


Figure 5.15 Per Flow 3 Packet delay

The Figure indicates that the VPQ algorithm performs very well as compared to the CATS and D-CATS algorithms in terms of packet delay. The VPQ packet delay starts from 6 ms and slightly increases to 6.3 ms over WLANs.

In Figure 5.16, it illustrates the flow 4 packet delay of the proposed VPQ along with the CATS and D-CATS algorithms over WLAN. The CATS algorithm and its packet delay starts from 20 ms and slightly increases to 24 ms. As well, the D-CATS packet delay starts from 20 ms and ends up at 18 ms. They notice that D-CATS performs better than the CATS algorithms.

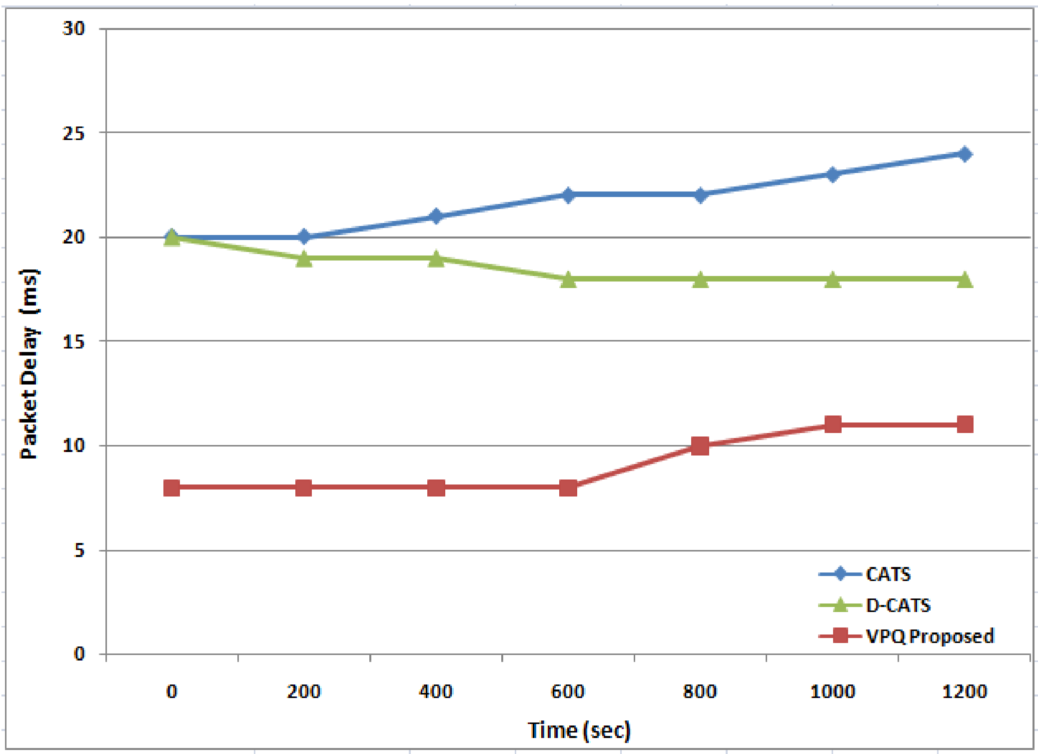


Figure 5.16 Per Flow 4 Packet delay

As they can see the packet delay in the VPQ algorithm performs very well as compared with the CATS and D-CATS algorithms. The VPQ packet delay starts from 8 ms and slightly increases to 11 ms over WLANs. We performed simulation that why the delay observed for all algorithms.

In Figure 5.17, it illustrates the total per flow packet delay of the proposed VPQ along with the CATS and D-CATS algorithms over the WLAN. The CATS algorithm and its packet delay starts from 17 ms and ends with an increase to 22 ms.

As well, the D-CATS packet delay starts from 16 ms and ends with an increase to 18 ms. They notice that D-CATS performed better than the CATS algorithms.

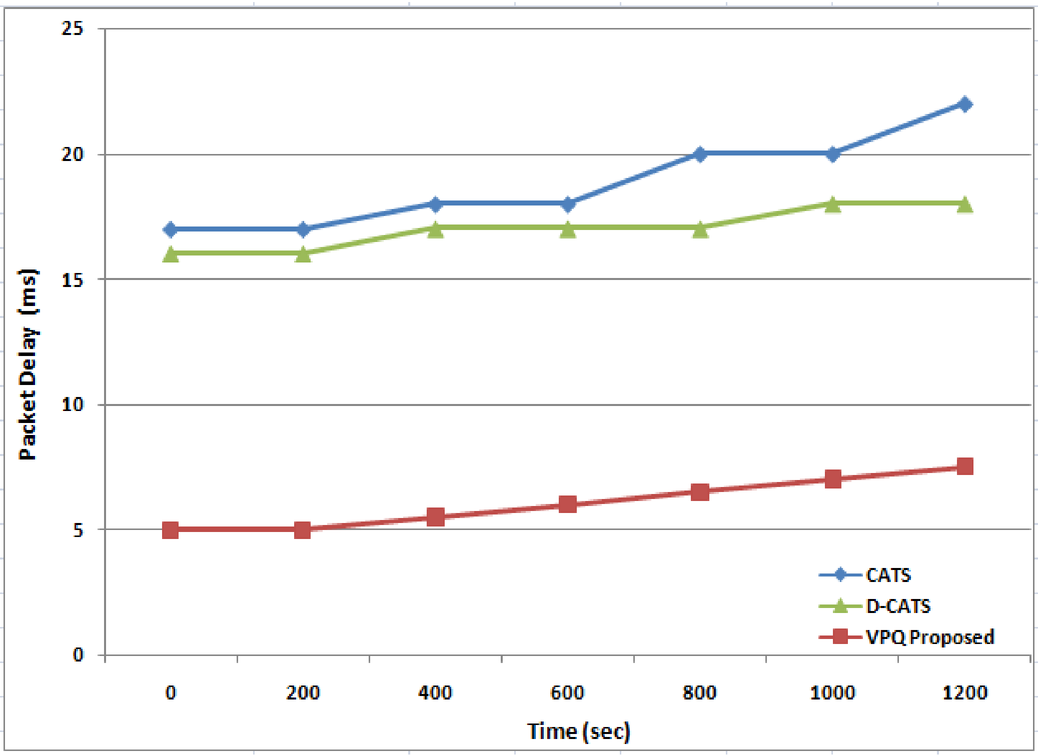


Figure 5.17 Total Per Flow Packet delay

As it can see, the packet delay in the VPQ algorithm performs very well as compared with the CATS and D-CATS algorithms. The VPQ packet delay starts from 5 ms and slightly increases to 7.5 ms over WLANs.

5.6 Simulation Results based on the Number of Stations

Figure 5.18 shows the number of mobile station as 14 to support against throughput Mbps ms over the VF flow on IP-based networks. In this Figure, the results evaluated on the proposed VPQ along with the CAPS, D-CATS and CATS algorithms are simulated over the WLAN. The details are in sections 2.5.14. Furthermore, the WLAN traffic is based on the Codec technique to improve the number of stations on the Real-Time Transmission Protocol (RTP).

In this Figure, they compare the performance of the proposed algorithm with the upgraded CATS algorithm i.e. D-CATS algorithm. Firstly, they illustrate the CATS algorithm; its throughput starts from 3 Mbps and ends at 2.8 Mbps. As well, the D-CATS throughput starts from 3.6 Mbps and ends with the same throughput of 3.6 Mbps against the number of active stations over WLANs.

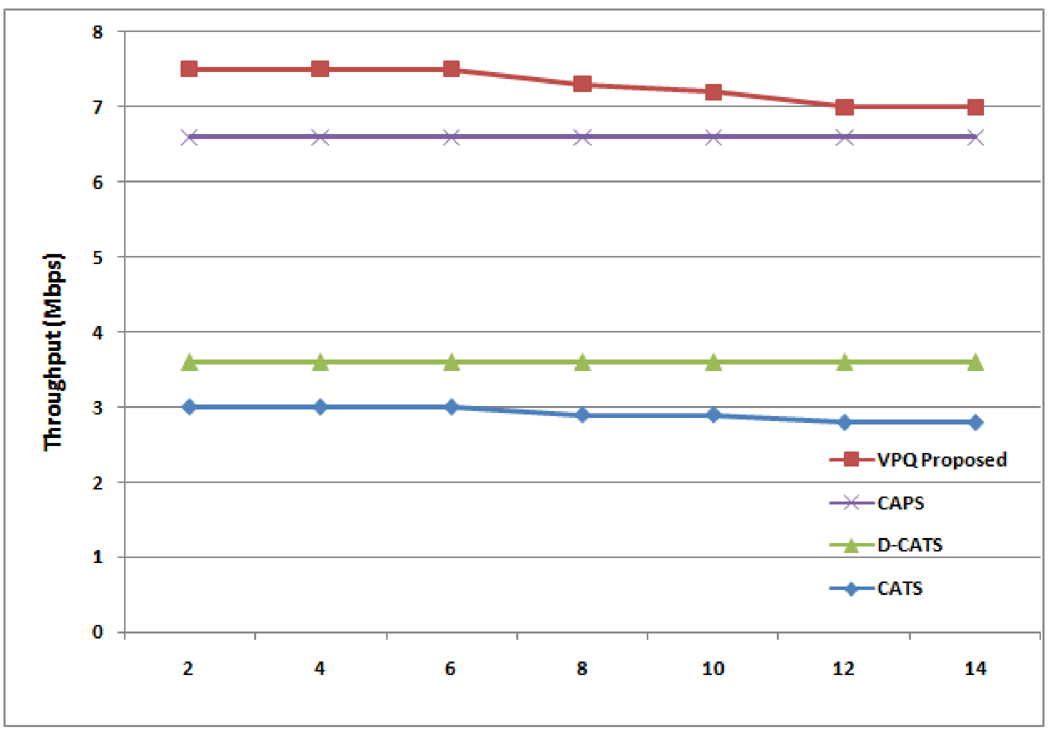


Figure 5.18 Throughput According to the Number of Active Stations

CATS starts from 6.6 and ends at 6.6 but it's better than the D-CATS and CATS algorithms. As it can see, the throughput of the VPQ algorithm performs very well as compared with the

CAPS, CATS and D-CATS algorithms. The VPQ throughput starts from 7.5 Mbps and slightly decreases to 7 ms over WLANs.

5.7 Simulation Results based on Packet Error Rate of Traffic Flow

Figure 5.19 shows the Packet Error Rate (PER) from 0 to 0.9 against time in ms over IP-based networks. If contention overhead is less than the Packet Error Rate, PER provides better per-flow protection and guarantees temporal fairness even in the presence of transmission errors. The details are in sections 2.5.14.

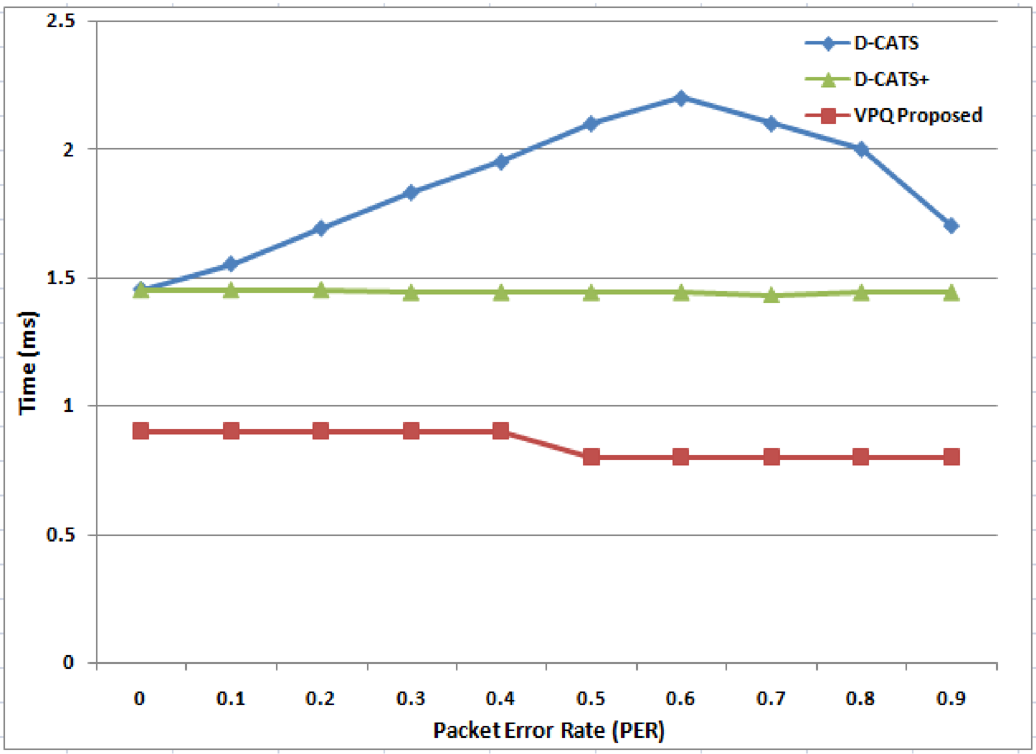


Figure 5.19 Contention overhead time according to the Packet Error Rate

It notice that the VPQ starts PER 0.9 ms and ends at 0.8 ms. D-CATS is around 1.45 ms the starting time and slightly increases to 2.2 ms at 0.6 PER.

After that it again decreases to end at 1.7 ms. They also evaluate with another upgraded D-CATS named D-CATS+. It starts from 1.45 ms and ends at 1.44 ms. Furthermore, they note that D-CATS+ is better than D-CATS over WLANs.

5.8 Test-Bed Results

Figure 5.20 shows the throughput of all the flows in the test-bed over IP-based networks. In this Figure, it can see the total throughput Mbps of the proposed VPQ along with the CATS and T-WFQ algorithms over the WLAN in the simulation. Figure 5.20 shows the throughput measured in Mbps and time in sec to evaluate the results.

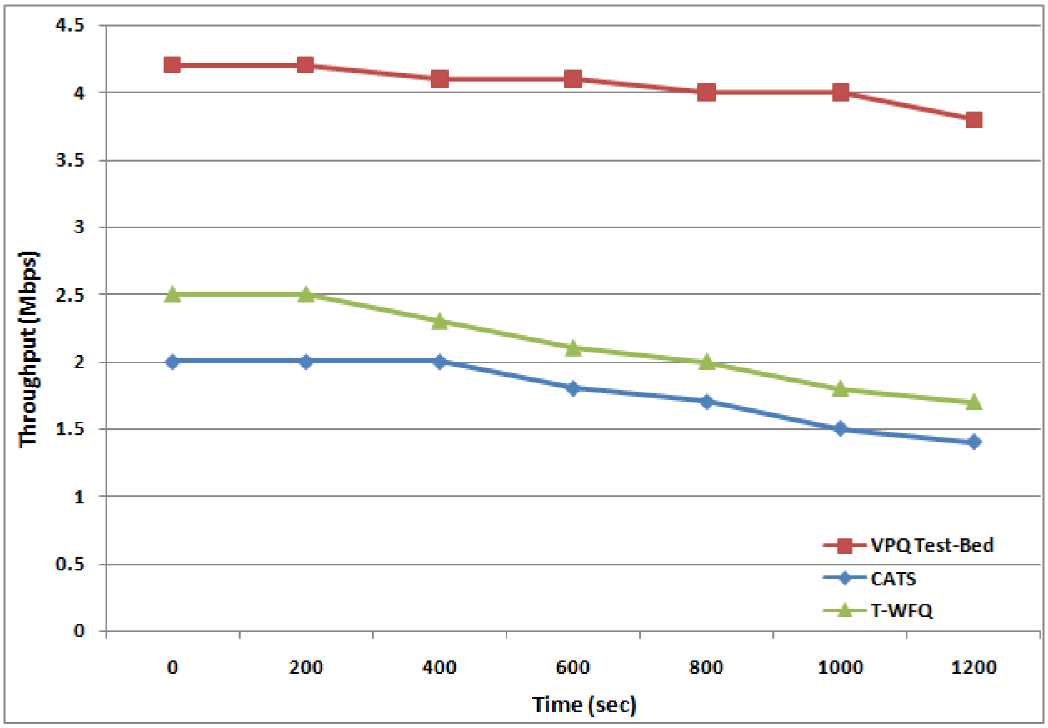


Figure 5.20 Total Throughput of Test-Bed

The CATS throughput starts from 2 Mbps and gradually decreases to 1.4 Mbps. The T-WFQ algorithm shows the higher throughput as compared with CATS that starts from 2.5 Mbps and reduces to 1.7 Mbps at 1200 sec.

The VPQ test-bed throughput performance starts from 4.2 Mbps and gradually decreases to 3.8 Mbps at 1200 sec. Proposed VPQ algorithm performs better than both the other algorithms over WLANs. The obvious reason is that the VPQ classifies the traffic into the VF and NVF. Moreover, it provides a high data rate flow that carries more packets.

5.9 Findings of the VPQ

The details are as shown in Table 5.3.

Notations:

TP = Throughput

VVF = Virtual VoIP Flow

PER = Packet Error Rate

VF = VoIP Flow

NVF = Non VoIP Flow

FI = Fairness Index

CE = Classify Enqueue

SM = Simulation

TB = Test-Bed

Symbol:

Support = √

Not Support = X

Partial Support = P √

Table 5.3 Findings of the VPQ

Scheduler Name & Research by	TP	VVF	PER	VF	NVF	FI	CE	SM	TB
NDRR S. Kanhere 2001	√	X	X	X	X	√	X	X	X
DO-WF^{2}Q X. Fei 2002	√	X	X	X	√	√	X	√	X
CB-SCFQ E. Palacios et al. 2004	√	X	X	√	√	√	X	X	√
LLEPS H. Wu et al. 2006	√	X	X	√	P √	√	X	√	X
CATS Y. Seok et al. 2007	√	X	X	P √	√	√	X	√	X
D-CATS Y. Seok et al. 2007	√	X	√	P √	√	√	X	√	X
T-WFQ Y. Seok et al. 2007	√	X	X	P √	√	√	X	√	X
CAPS Y. Fallah, 2007	√	√	X	P √	√	X	√	√	X
DTT R. Garroppo 2007	√	√	X	√	√	√	√	√	X
QP-CAT S. Shin et al. 2008	X	√	X	√	√	X	√	√	√
App-Aware N. Bayer et al. 2010	√	X	X	√	√	X	X	√	X
Proposed VPQ K. Nisar et al. 2011	√	√	√	√	√	√	√	√	P √

Table 5.4 Findings of the VPQ Results with Related Schedulers

Research Work	Mobility of Mobile Station	Packet Size Flow	Packet Delay	Number of Stations Throughput	Test-Bed Throughput
T-WFQ	50% Performance	60% Throughput	X	X	2.5Mbps to 1.7Mbps
CATS	70% Performance	50% Throughput	17ms to 23ms	3Mbps	2Mbps to 1.4Mbps
D-CATS	X	X	16ms to 18ms	3.8Mbps	X
D-CATS+	X	X	X	X	X
CAPS	X	X	X	6.8Mbps	X
VPQ Proposed	99% Performance	90% Throughput	5ms to 7ms	7.5Mbps	4.2Mbps to 3.8Mbps

5.10 Summary

In this chapter, it explained simulation and experimental results in the graphs. They also evaluated Voice Priority Queue (VPQ) scheduler's results with the most related scheduler and algorithms. They evaluated the VPQ with Contention-Aware Temporally fair Scheduling (CATS), Temporally-Weighted Fair Queuieng (T-WFQ), Decentralized-CATS (D-CATS), Decentralized-CATS+ (D-CATS+) and controlled access phase scheduling (CAPS). They have also presented the results obtained from the test-bed environment over WLANs. In the next and final chapter, they will conclude research work. They will also explain the achieved objectives, scopes and limitations of the current research work, contributions, conclusion, and they will discuss the future works.

CHAPTER 6

CONCLUSION & FUTURE WORK

6.1 Introduction

In this thesis, they have addressed the Voice over IP (VoIP) scheduling matters which noticeably affect the performance of the VoIP Flow (VF) over WLANs using IEEE 802.11 standards. The VoIP is a delay sensitive traffic due to real-time applications on networks. The assessment of VF quality in the VoIP is an essential requirement for technical and commercial motivation. In this thesis, they proposed a new Voice Priority Queue (VPQ) scheduling system model and algorithms for the VoIP over WLANs to solve scheduling issues over IP-based networks. They presented new contributions, through the three stages of the VPQ. They have verified and validated the VPQ results in simulation. In addition, they evaluated the VPQ on a test-bed environment in the VoIP Research and Development (R&D) lab at Universiti Teknologi Petronas. In this chapter, they discuss the achieved objectives, scope and limitations, contributions, conclusion and future work.

6.2 Achieved Objectives

In this section, they discuss the achieved objectives, those obtained from the VPQ scheduling system model and algorithms over networks. They did a comparison with related schedulers and algorithms to evaluate the VPQ to enhance the performance of the VoIP over WLANs using IEEE 802.11 standards. Furthermore, they examined, simulated and did test-bed techniques to verify the VPQ scheduler. They achieved these objectives to classify the VoIP Flow (VF) and Non-VoIP Flow (NVF) traffic over WLANs. In this research, the achieved objectives are as follows:

The VPQ is based on three stages of scheduling system model over WLANs. The novelty of the VPQ is providing classification of the VoIP Flow (VF) and Non-VoIP Flow (NVF) traffic and they proposed three stages of the VPQ for a VoIP application.

The VPQ scheduling system model is challenged by offering different levels of QoS, a higher throughput, higher fairness index and performance complications. The first stage of VPQ model provided better performance as compared with most related work on a normal stage over IP-based networks. The second stage of the VPQ introduced Virtual-VoIP Flow (Virtual-VF) flows for bursty traffic over WLANs and the third stage provided the Switch Movement (SM) that moves to the VF when the NVF is in the empty condition.

Another, novelty of the VPQ is classifying the VoIP Flow (VF) and Non-VoIP Flow (NVF) traffic over WLANs. They applied VPQ algorithms to classify the VF and NVF on the enqueue stage. The VF and NVF traffic performance technique gave priority to the VoIP application without disturbing the NVF traffic flow over the network. Due to VPQ scheduling system model, NVF also performed well over IP-based networks.

They compared the VPQ with most related scheduling algorithms. They performed a simulation in an NS-2 environment to evaluate, validate and verify the VPQ with other schedulers. They observed that the VPQ provides better results as compared with Contention-Aware Temporally fair Scheduling (CATS) [120], Decentralized-CATS (D-CATS) [120], Decentralized-CATS+ (D-CATS+) [120], Temporally-Weight Fair Queue (T-WFQ) [120] and Controlled Access Phase Scheduling (CAPS) algorithms [112], [113].

In addition, they evaluated the VPQ scheduler on a test-bed environment in the VoIP Research and Development (R&D) lab at Universiti Teknologi Petronas. The test-bed environment also provided better results of the VPQ scheduler over WLANs.

6.3 Scope and Limitations of the Current work

This thesis is based on essential ideas and concepts that provide efficient traffic flow for the VoIP application. The scope of this thesis may lead to the proposal of a new VPQ scheduling system model and algorithms for the VoIP over WLANs. The VoIP is known as a delay sensitive application, they can apply this application on an emergency call center, internal VoIP exchange and a WLAN environment. One of the limitations of the current work was the VoIP Flow (VF) with the highest priority queue over IP-based networks. The other limitation of research is that the VPQ can only be applied for VoIP applications. This research is based on the VF and NVF traffic flow on downlink and uplink traffic over WLANs.

6.4 Contributions

The main contributions of this thesis are as follows:

One of the contributions is based on the new Voice Priority Queue (VPQ) scheduling model that classifies the VoIP Flow (VF) and Non-VoIP Flow (NVF). In chapter three, they discussed the details of the VPQ model over IP-based networks. The main mechanism of the VPQ system model is to initialize the traffic into the VF and NVF traffic flows with regards to packets, frame, energy and weight of traffic flow over a WLAN using IEEE 802.11 standards.

The main method of the first stage of the VPQ model to provide both traffic VF and NVF flows over the WLAN. In the first stage, the VPQ provides throughput with promises of error-free flows and long term fairness with efficient techniques.

The main method is based on the development of the second stage of the VPQ system model to provide the highest priority to the VF without any delay. In the second stage, the VPQ introduced the Virtual VoIP Flow (Virtual-VF) for the bursty traffic over WLANs. The details have already been discussed in chapter three.

In the third stage of the VPQ, they introduced the Switch Movement (SM) that moves to the VoIP Flow (VF) and Non-VoIP Flow (NVF) when the NVF flows are empty due to non-real-time traffic over IP-based networks. This switch provided better throughput and fairness for the VF traffic and they applied the switch only to the bursty traffic flow at the peak time of the VoIP communication and network.

In chapter three, they proposed the VPQ system model in a comprehensive way for the reader to better understand of VoIP over WLANs. It initialized the traffic classification of Enqueue, Dequeue, VF, NVF, Traffic shaping, Token Bucket (TB), VPQ Component, Buffer Component, Virtual Flow Component and VF Switch component over WLANs. This thesis provides the detailed understanding of the Quality of Services, Scheduling algorithms, VF and NVF and the WLAN using IEEE 802.11 standards.

6.5 Conclusions

The VoIP is applied on VoIP Conferencing, Fax over IP, Directory Services over Telephones, and VoIP radio over WLANs. IP-based networks were firstly considered to transmit data traffic and they are managing this task adequately. They are not mainly designed to transmit real-time applications such as VoIP traffic in addition to data traffic. This thesis proposes an efficient scheduler and algorithms for a VoIP application over WLANs. This thesis proposes a new Voice Priority Queue (VPQ) scheduler and algorithms to provide an efficient traffic flow over WLANs. It proposed the three stages of the VPQ scheduler system model to fulfill the scheduling requirements over IP-networks.

The VPQ classified the VoIP Flow (VF) and Non VoIP Flow (NVF) traffic flow and sent it forward to the end user without any delay over the WLANs. Furthermore, it compared the VPQ scheduler with well known scheduler and algorithms like, CATS, D-CATS, D-CATS+, T-WFQ and CAPS algorithms. They observed in simulation and experimental environment that the VPQ provides better results for VoIP traffic over IP-based networks.

6.6 Future Works

There are three main research areas for future works.

Although the thesis has focused on the Voice over IP (VoIP), the approach of the new Voice Priority Queue (VPQ) considered can be applied to Video Conferencing. They need only a small technical change and the VPQ could easily be implemented in Video Conferencing. The main technical metric is conversion VoIP packet into Video Conference frame because video conferencing is a combination of voice and video. It needs more bandwidth as compared with VoIP.

The VPQ algorithm can be applied to the VoIP Radio system due to very similar technical matrices. They only need frequency bands to implement the VPQ over WLANs and LANs. The VPQ algorithm can be applied to Internet Protocol TV (IPTV). They only need frequency bands to implement the VPQ over WLANs and LANs.

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Appendix A:

Research Projects Medal / Award / Recognition

S. No	Research Title / Authors Name	Medal / Awards / Recognition
[1]	**K. Nisar,** A. Said, and H. Hasbullah, "Voice Priority Queue (VPQ) Scheduler Architecture and Algorithms for VoIP over WLAN Networks,"	**Bronze Medal** in recognition of my contribution in the 26th Engineering Design Exhibition (EDX), held from 25-26, October. 2010, Universiti Teknologi PETRONAS
[2]	**K. Nisar,** A. Said, and H. Hasbullah, "Voice Priority Queue (VPQ) Scheduler Architecture and Algorithms for VoIP over WLAN Networks,"	**Best Poster Presentation Award** in recognition of my contribution in the 26th Engineering Design Exhibition (EDX), held from 25-26, October. 2010, Universiti Teknologi PETRONAS
[3]	**K. Nisar,** S. Ahmed, A. Said, and H. Hasbullah, "First Stage of Voice Priority Queue (VPQ) Scheduler for VoIP over WLAN Networks,"	**Best Paper** Category, International Journal of Computer Science and Information Security, Pittsburgh, PA, USA, 23, December. 2010. (Indexing in Google Scholar).
[4]	**K. Nisar,** A. Said, and H. Hasbullah, "Second Stage of Voice Priority Queue (VPQ) Scheduler for VoIP over WLAN Networks,"	**Best Paper** Category, International Journal of Computer Science and Information Security, Pittsburgh, PA, USA, 20, December. 2010. (Indexing in Google Scholar).

Appendix B:

International Journal Publications

S. No	Authors Name	Title & Recognition of Journal Publications
[1]	**K. Nisar,** S. Ahmed, H. Hasbullah, and A. Said	"First Stage of Voice Priority Queue (VPQ) Scheduler for VoIP over WLAN Networks," *International Journal of Computer Science and Information Security, Pittsburgh, PA, USA, Vol. 9, No.1,* 23, December. 2010. (Accepted and Indexing in Google Scholar).
[2]	**K. Nisar,** A. Said, and H. Hasbullah,	"Second Stage of Voice Priority Queue (VPQ) Scheduler for VoIP over WLAN Networks," *International Journal of Computer Science and Information Security, Pittsburgh, PA, USA, Vol. 9, No.1,* 20, December. 2010. (Accepted and Indexing in Google Scholar).
[3]	**K. Nisar,** A. Said, and H. Hasbullah,	"Voice Priority Queue (VPQ) Fair Scheduler for VoIP over WLAN Networks," *International Journal on Computer Science and Engineering, Chennai, India, Vol. 3, No.2,* 15, February. 2011. (Indexing in Scopus).
[4]	**K. Nisar,** A. Said, and H. Hasbullah,	"Multimedia Performance System Model for VoIP Networks," *Journal of Communication and Computer, David publishing Company, USA. (*Accepted and Indexing in David Publishing).

S. No	Authors Name	Title & Recognition of Journal Publications
[5]	**K. Nisar,** A. Said, and H. Hasbullah,	"An Efficient Voice Priority Queue (VPQ) Scheduler Architectures and Algorithm for VoIP over WLAN Networks," *International Journal of Computer Science Letters (IJCSL), ISSR Journals, Vol. 2, No. 2,* pp. 1-13, 4, September. 2010.
[6]	H. Kazemitabar, S. Ahmed, **K. Nisar,** A. Said, and H. Hasbullah,	"A comprehensive review on VoIP over Wireless LAN networks," *International Journal of Computer Science Letters (IJCSL), ISSR Journals, Vol. 2, No. 2,* pp. 22-38, 4, September. 2010.

Appendix C:

International & Local Conference Publications

S. No	Authors Name	Title & Recognition of Conference Publications
[7]	**K. Nisar,** A. Said, and H. Hasbullah,	"Third Stage of Voice Priority Queue (VPQ) Scheduler for VoIP over WLAN Networks," *2011 IEEE, International Conference on Information and Computer Networks (ICICN 2011), Guiyang, China,* 20, December. 2010. (Accepted).
[8]	**K. Nisar,** A. Said, and H. Hasbullah,	"Second Stage of Voice Priority Queue (VPQ) Scheduler and Algorithms for VoIP over WLAN Networks," *2011 IEEE International Conference on Communication and Broadband Networking (ICCBN 2011), Kuala Lumpur, Malaysia*, 23, December. 2010. (Accepted).
[9]	H. Kazemitabar, S. Ahmed, **K. Nisar,** A. Said, and H. Hasbullah,	"A Survey on Voice over IP over Wireless LANs," *World Academy of Science, Engineering and Technology, Venice, Italy, 2010, Indexing in Scopus,* Article 63, No. 71, pp. 252-258, 24,November. 2010.
[10]	**K. Nisar,** A. Said, and H. Hasbullah,	"Enhanced Performance of Packet Transmission Using System Model Over VoIP Network" *International Symposium on Information Technology 2010 (ITSim 2010), IEEE, KLCC, Kuala Lumpur, Malaysia*, pp. 1005-1008, 15, June. 2010.

S. No	Authors Name	Title & Recognition of Conference Publications
[11]	**K. Nisar,** A. Said, and H. Hasbullah,	"Enhanced Performance of WLANs Packet Transmission over VoIP Network," *2010 IEEE 24th International Conference on Advanced Information Networking and Applications, Workshops, (AINA 2010), supported by IEEE Computer Society, Perth, Western Australia,* pp. 485-490, 20-23 April. 2010.
[12]	**K. Nisar,** A. Said, and H. Hasbullah,	"Enhanced Performance of IPv6 Packet Transmission over VoIP Network," *2nd IEEE International Conference on Computer Science and Information Technology, 2009, ICCSIT, Beijing, China,* pp.500-504, August, 11. 2009.
[13]	**K. Nisar,** A. Said and H. Hasbullah,	"Internet Call Delay on Peer to Peer and Phone to Phone VoIP Network," *International Conference on Computer Engineering and Technology 2009 (ICCET 2009). IEEE, Singapore,* Vol. 2, pp. 517-520, 24, January. 2009.
[14]	H. Hasbullah, **K. Nisar,** and A. Said,	"The Effect of Echo on Voice Quality in VoIP Network," *International Association for Science and Technology Development (IASTED) Calgary, Canada. Advances in Computer Science and Engineering (ACSE) 2009 Phuket, Thailand.* 2009.

S. No	Authors Name	Title & Recognition of Conference Publications
[15]	**K. Nisar,** A. Said and H. Hasbullah,	"Next Generation Network's Quality of Service (QoS) for VoIP in Telecommunication," *Seminar on Science & Technology 2008, School of Science and Technology, Universiti Malaysia Sabah, Malaysia,* 29-30 October, 2008.