THE VPQ SCHEDULER IN ACCESS POINT FOR VOIP OVER WLAN

Kashif Nisar[1] and Suhaidi B Hassan[2]

[1,2]InterNetWorks Research Group
School of Computing, College of Arts and Sciences
Universiti Utara Malaysia
06010 UUM Sintok, MALAYSIA
kashif@uum.edu.my

ABSTRACT

The Voice over Internet Protocol (VoIP) application has observed the fastest growth in the world of telecommunication. VoIP is seen as a short-term and long-trem transmission for voice and audio traffic. Meanwhile, VoIP is moving on Wireless Local Area Networks (WLANs) based on IEEE 802.11 standards. Currently, there are many packet scheduling algorithms for real-time transmission over network. Unfortunately, the current scheduling will not be able to handle the VoIP packets with the proper manner and they have some drawbacks over real-time applications. The objective of this research is to propose a new Voice Priority Queue (VPQ) packet scheduling and algorithm to ensure more throughput, fairness and efficient packet scheduling for VoIP performance of queues and traffics. A new scheduler flexible which is capable of satisfying the VoIP traffic flows. Experimental topologies on NS-2 network simulator wereanalyzed for voice traffic. Preliminary results show that this can achieve maximum and more accurate VoIP quality throughput and fairness index in access point for VoIP over WLANs. We verified and validated VPQ an extensive experimental simulation study under various traffic flows over WLANs.

KEYWORDS

VF, NVF, VoIP, WLAN, VPQ, WLAN

1. INTRODUCTION

This research Voice over IP over Wireless Local Area Networks (VoIPWLAN) is in the developing field of wireless broadband Internet technologies which has the great potential to provide a low-cost high-speed Internet voice calls with user mobility that can profoundly impact our lives in a positive way. VoIP over WLAN environment allows users to make IP-based calls over global networks. VoIP transmits IP-based telephony calls over packet on WLANs. In IP-based networks, analogue voice signals are digitized and move on real-time transmission over network. Then, find the efficient way to reach the proposed destination. Normally, they out of the order from original order and receiver side the packets rearranged in the proper way before convert into aging analogue voice signals for voice conversion over networks [1], [2], [3], and[4].

Currently, there are approximately 1 billion fixed telephony lines and 2 billion mobile-phone lines in the world. Now, we are moving ahead to IP network based protocols known as Voice over Internet Protocol (VoIP) [5]. VoIP over Wireless Local Area Network (WLAN) might be a leading application in collaboration with 3rd Generation (3G) mobile network [6]. A variety of new multimedia applications such as VoIP, video on demand (VoD), Internet Protocol TV (IPTV) and teleconferencing are based on network traffic scheduling algorithms. A number of research solutions have been proposed to satisfy different Quality of Service (QoS) requirements [7], [8], [9], [10], [11] and [12].

DOI : 10.5121/ijmnct.2011.1202

1.1 VoIP Protocol Architecture

Figure 1 shows the fundamental of VoIP protocol stack architecture to implement a VoIP network system over WLANs [13], [14]. VoIP is a real-time application and transmit the voice using Real-Time Transport Protocol (RTP), User Datagram Protocol (UDP) and Internet Protocol (IP) (RTP/UDP/IP) over WLANs [15], [16], [17] and [18]. Each VoIP packet has the headers, RTP (12 bytes), UDP (8 bytes), and IP (20 bytes) headers. Lastly, data-link layer Medium Access Control (MAC) has a (34 bytes) header. These all bytes headers calculate as 74 bytes of overhead in VoIP protocol.

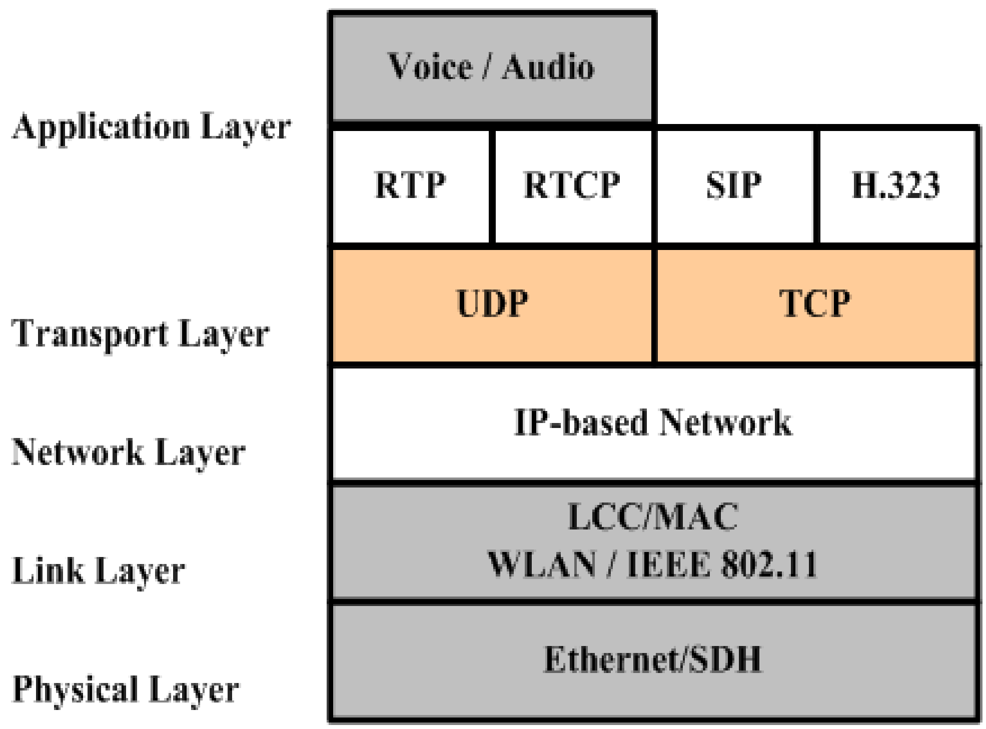

Figure 1. Voice processing Signals in a VoIP Network System

The Session Initiation Protocol (SIP) was considered for handles a multimedia call setup and H.323 is considered by ITU to allocate IP-based phones on the public telephone network to talk to PC-based phone over IP-based networks [19] and [20].ITU 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 [21], [22], [23], [24] and [25].

1.2 VoIP over WLANs Networks

VoIP over WLANs have observed a fastest growth in the World of communication. WLAN is most guaranteeing technologies among wireless networks, which have been facilitated high-rate voice services at very less cost and flexibility over IP-based networks [26], [27], [28] and [29]. Main source for such adaptation is that VoIP real-time application over WLANs is more flexible than traditional public switched telephone networks systems (PSTN) [30].Moreover, VoIP can support multiple infrastructure environments IP-based-Phones (IP-Phone, PC-based soft-Phone, IP-based Packet-Phone), Soft-Phones (PC-to-PC Phones), Traditional and mobile Phones (Telephone, Cell-Phone). Detail are as below in table 1.

Table 1 Classifications of Phone Systems

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

VoIP provides mix mode communication with PC-to-PC, PC-to-IP-Phone and PC-to-Cell-Phone commutation over WLANs. They are moving on campuses, hotels, airports, health care, commercial, education, and industries to provide voice traffic. WLAN also provide audio, voice and video conferences over IP-based networks [31], [32], [33], [34], [35] and [36].In WLANs, there are two essential kinds of the services architectures: ad-hoc architecture and infrastructure architecture. In ad-hoc architecture, station (STA, a mobile node) can be able to connect with IP-based network without the connectivity to any wired backbone network and without the need of an Access Point (AP) [37]. In infrastructure, the STA can be able to connect with IP-based network with the connectivity to any wired backbone network and with the need of an AP. In this paper we will focus on infrastructure architecture network where VoIP traffic is transmitted signals via an AP. WLANs provide number of industries standards of AP. Each AP can maintain a restricted number of parallel voice nodes [38], [39], [40], [41], [42], [43], [44], [45] and [46].

1.3 VoIP, IEEE 802.11 MAC and WLANs Standards

The IEEE 802.11 WLANs, we called as a wireless Ethernet and play an important function in the future-generation networks. WLAN based on Link Layer (LL). LL divided into Logical Link Control (LLC) and Medium Access Control (MAC) sub-layer categorizes with two functions, Distributed Coordination Function (DCF) and Point Coordination Function (PCF) [47], [48] and [49]. 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. [50], [51], [52], [53], [54], [55], [56] and [57].

1.4 Problem Statement

IP-based networks are managing voice, data, web browsing, email, and video applications on the same network flow over WLANs. They were not mainly designed for real-time transmission over WLANs and it can be a deadlock in the traffic flow over WLANs. We are showing a bottleneck topology of mix mode traffic over WLAN as below in figure 2 [58].

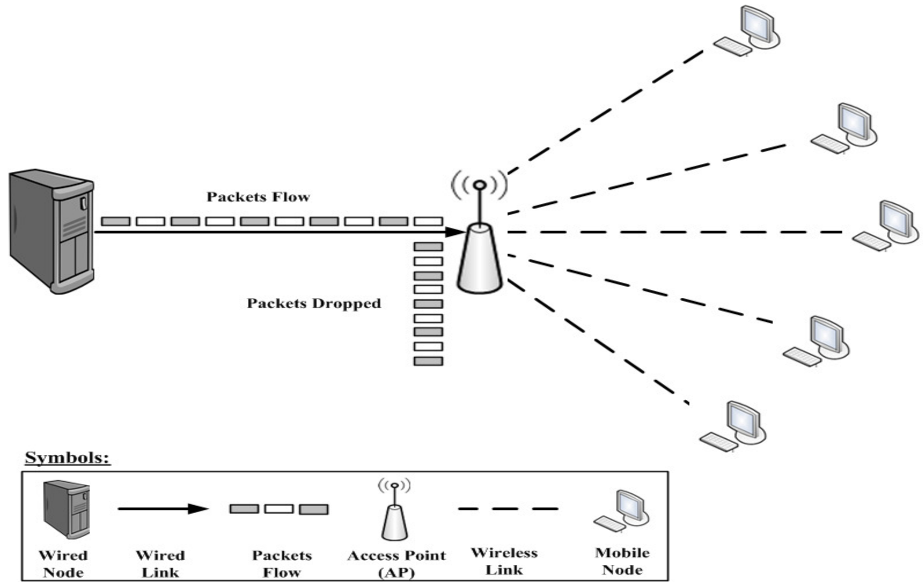

Figure 2.Bottleneck Topology of mix mode Traffic over LAN networks

Quality of Services (QoS) is considered the main issue in VoIP system. Due to its importance, following research focuses on solving the VoIP scheduling algorithm problem. This research tries to compare with some well know real-time scheduling algorithms over WLANs. The proposed method tries to achieve better acceptable results for VoIP high-speed real-time application. In figure 1.6, we implement a topology, it has bottleneck problem due to attacker on node 2 and node 5. VoIP is a real-time application that needs timely techniques to enhanced traffic over networks. This is a challenging task on VoIP networks.

Through the past decades many schedulers were introduced to solve real-time traffic application issues. These schedulers can be divided into three groups and these groups are as following, packet-based schedulers, frame based-packet schedulers, and regulative packet schedulers. These above problems degrade the QoS of VoIP over IP-based networks. We need to introduce a new voice scheduling algorithm to solve above VoIP traffic issues. New method should be an efficient, fair, high throughput, bandwidth guarantee and that will enhance performance of VoIP over WLANs

1.5 Aim and Objectives

The aim of this paper is to introduce an efficient schedulers and algorithms that support the VoIP application over WLANs. We will assume a fundamental related work to examine the available schedulers with their outcomes and their drawbacks. We will introduce new scheduler and algorithms to enhance the performance of VoIP over WLANs using IEEE 802.11 standards. We will evaluate, examine, and simulate our techniques with related algorithms for real-time applications. To improve the real-time traffic scheduler algorithm it is possible to resolve many of these problems. In this research the specific objectives are as following:

- To develop a new Voice Priority Queue (VPQ) scheduler architecture and algorithms for VoIP traffic that can be proficient to fulfil the scheduling requirements over WLANs.

- To classify VoIP-Flow (VF) traffic and Non-VoIP-Flow (NVF) traffic over WLAN using IEEE 802.11 standards.
- To compare VPQ scheduler and algorithms with more related work, to evaluate, validate and verify our scheduler with other schedulers.
- To enhance the scalability of traffic over WLAN, using our Test-Bed in VoIP.

The paper is organized as follows. In section II we discuss the related work with different scheduling algorithms and initiate their limitation for multimedia application. In section III, we proposed a new VoIP scheduling algorithm and methodology. In section IV we describe simulation experimental Setup that compares the efficiency between new VoIP scheduling algorithm and other related scheduling algorithm. Section V, we describe the results and in section VI we conclude this paper with future research work remarks.

2. RELATED WORK

The high-speed packet-switched networks are essential research areas. Voice over IP (VoIP) over Wireless Local Area Network (WLAN) network is one of the most applying technologies to utilize high-speed packet-switched networks. The IEEE 802.11 standard has expanded with an importance of researchers in the last decade.There are some scheduling algorithms to support packet scheduling in wire and wireless networks. Few of them are as following: Class Based Queue (CBQ), Faire Queue (FQ), Weight Faire Queue (WFQ), Generalized Processor Sharing (GPS), Worst-case Fair Weighted Fair Queuing (WF2Q), 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 (DCATS). We will also study a general discussion of related research work in this section for real-time applications.

2.1 Class Based Queue (CBQ)

Floyd et. al. [59] introduced the Class Based Queue (CBQ) for hierarchical bandwidth link-sharing and resource technique of packet-switch networks. CBQ is a class based algorithm and share the bandwidth for each class with well organize manner over IP-based networks. CBQ manages bandwidth link-sharing rations for all classes. It maintains each queue and provide fairly link-sharing. We have noticed that CBQ best solution for data traffic. It is implementing with gateway technique and fulfils the range of service and link-sharing. CBQ is a combination of classifier, estimator, selector, and over limit process to schedule the traffic flow classes those extended link sharing limits.

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

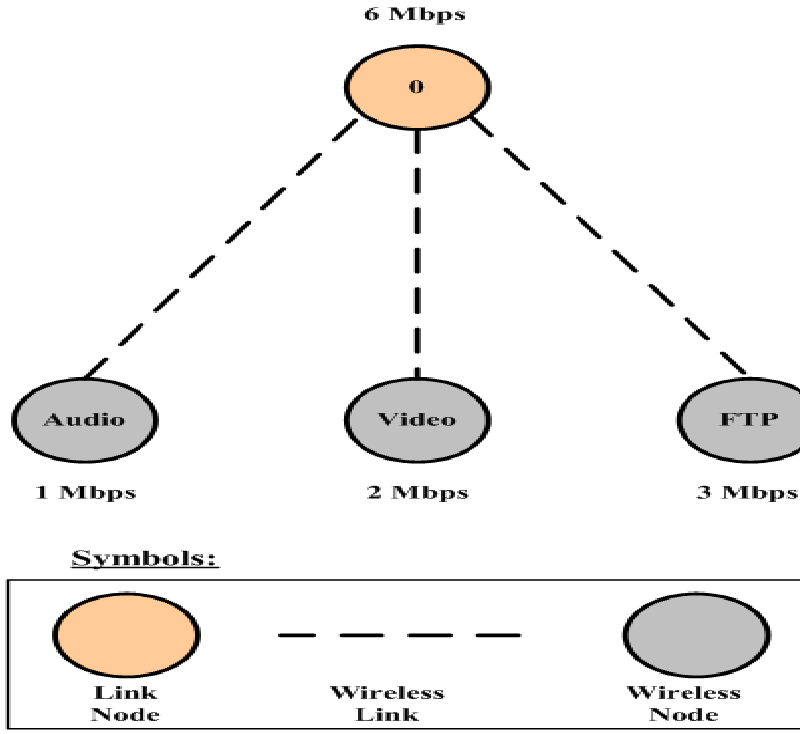

Figure 3. Priority, Structure of Link-Sharing Allocation [59]

The drawbacks of CBQ are as following: If bursty traffic on real-time application and it will become the cause of the delay. CBQ does not assume the delay of packets and the dealy of packets is needed less and scalable for VoIP application. The real-time application requires a specific buffer to control the bursty traffic on IP-based networks but the bulky buffer introduces the delay of playback longer.

2.1 Weighted Fair Queue (WFQ)

WFQ is a sort-based packet scheduling algorithm with latest updating of Generalized Processor Sharing (GPS). WFQ offers N queues same time with different bandwidth service rate by providing each queue a weight and arranges different percentage of output port bandwidth. WFQ calculates the departure time of each packet and manages multiple sizes of packets. The traffic flows of bigger packets are not assigned extra bandwidth. WFQ is inefficient to calculate the timestamps. WFQ will invite the work complexity of $O(\log n)$, where task n is the number of flows. Furthermore, due to slow process of sorting among the timestamps, WFQ is not suitable of real-time applications.

2.2 Contention-Aware Temporally fair Scheduling (CATS)

Seoket. al. introduced [61] the Contention-Aware Temporally fair Scheduling (CATS) is packet based algorithm and offered fairness traffic flow over WLANs IEEE 802.11a/b standards. CATS introduced for equal time sharing for each flow over IP-based networks. CATS decide packets scheduling order after virtual finish time. In addition, the scheduler is proficient for performance in multi-rate WLANs. CATS provides solution for Carrier Sense Multiple Access / Collision Avidness (CSMA/CA) technique. Figure 4 shows the flowchart of QP-CAT algorithm.

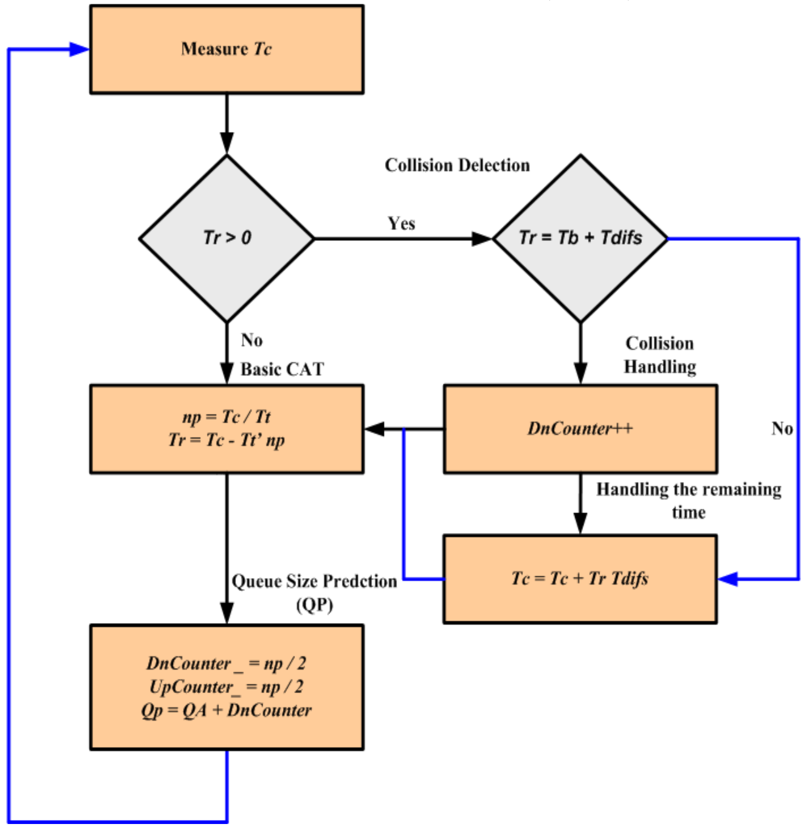

Figure 4. Flowchart of QP-CAT Algorithm [60]

The drawbacks of Contention-Aware Temporally fair Scheduling (CATS) are as following. The CATS based on Generalized Processor Sharing (GPS) that applied on wired based networks and based on fluid flow mechanism. GPS based on fixed link of capacity and cannot facilitate the pre-flow process. CATS applied T-GPS and cannot provide each flow calculation on WLAN. Meanwhile, the Temporally-Weighted Fair Queuing (T-WFQ) introduced that also based on GPS and less performance than CATS.

3. Methodology

The scheduling architectures provide major role in Wireless LAN (WLAN) networks to fulfill the essential requirements of Voice over Internet Protocol (VoIP) traffic flow over IP-based network. To guarantee the requirements of the new VoIP over WLAN applications, a Quality of Service (QoS) requires responsible scheduling architecture, algorithms, efficient traffic, and enhances voice traffic flow over WLANs. Being an emerging technology, WLAN offers VoIP, Internet Protocol TV (IPTV) and High-Performance Video-Conferencing (HP-VC) traffic over IP-based networks. The goal of this section is to propose an efficient traffic scheduler for VoIP traffic flow over WLANs. The numbers of related schedulers have been proposed to support traffic flow over IP-based networks. We noticed, these related schedulers support traffic flows with limited services, the special requirements of real-time application cannot meet with fully support over WLANs.

The scheduling architecture an important technique to achieve efficient throughput and fairness over WLANs IEEE 802.11 Standards. Scheduling technique illustrated the voice traffic over WLANs. It will be able to offer link-sharing of bandwidth, to tolerate the status of changing traffic queues and to be scalable over IP-based networks. The Weighted Round Robin (WRR) [62] scheduler idea is to increase and allocate the different quantity of traffic flow in a round for each non-empty queue over networks. The Round Robin (RR) scheduler is one that keeps a separate queue for every flow with packets waiting flow, and RR serves one packet from each queue in turn [63].

3.1 Voice Priority Queue (VPQ) Scheduler Architecture

VPQ is pre-packet delay bounds and provide both bounded delay and fairness over WLANs IEEE 802.11 Standards. VPQ is making it easy to provide both delay-guarantees and fairness concurrently over networks. VPQ provides throughput guarantees for error-free flows, long term fairness for error flows, and short term fairness for error-free flows, and graceful degradation for flows that have received excess service. VPQ scheduler architecture is as following in figure 5.

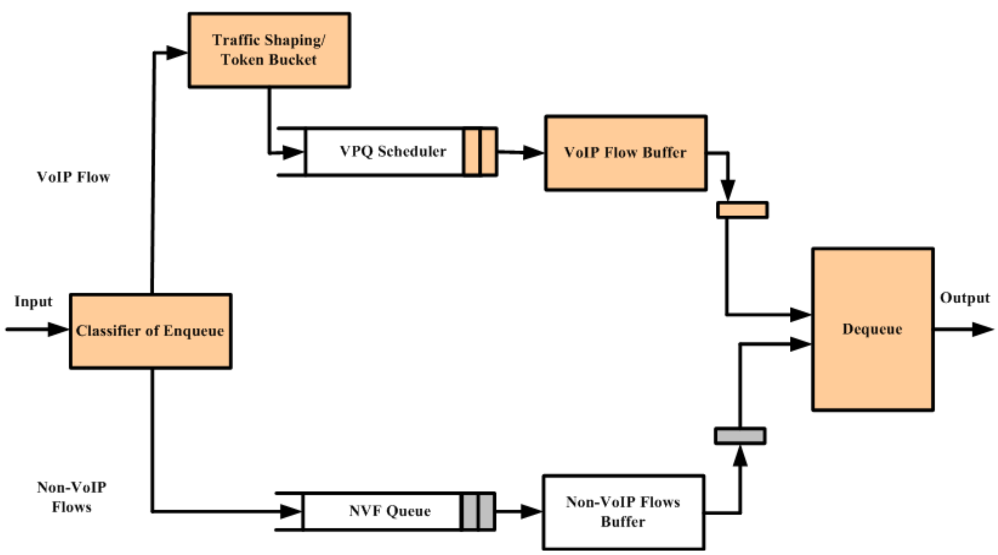

Figure 5. Voice Priority Queue (VPQ) Scheduler Architecture

VPQ scheduler architecture based on initializes traffic flows, classification of enqueue traffic flows, VoIP-Flow (VF) and Non-VoIP-Flow (NVF), traffic shaping, token bucket, Voice Priority Queue (VPQ) management system, VoIP-Flows buffer, Non-VoIP-Flow buffer and dequeue traffic flows for end user over WLANs IEEE 802.11 Standards. In Figure 5, classification of enqueue traffic flows sorted-based and checks the index of the incoming packet. The classification architecture supports a mechanism that work as like Differentiated Services (DiffServ) and dispatched into different traffic flows depending on their destination mobile stations. After that, classification send traffic flows to the VF traffic shaping and token bucket flow. The shaping controls the amount of flow and sent traffic flow to the token bucket flow. The token bucket applies on bursty traffic as regulated maximum rate of traffic over WLANs IEEE 802.11 Standards. The traffic shaping and token bucket send VF flows to main Voice Priority Queue (VPQ) scheduler component for processing over network. VPQ forward these packets into Voice- -Flow-Buffer (VFB). The VFB is temporary buffer flow for VF and NVF traffic. For easily understand we would like to inform you that Base Station (BS) has instantaneous and perfect understanding of VF, NVF channel's position due to their weight, energy and priority queue. Finally, VF and NVF flows dequeue traffics in well manner to the destination over WLANs.

4. EXPERIMENTAL SETUP & SIMULATION

In this section, we will discuss experimental setup and simulation of our methodology. We will perform validation and verification of the developed simulation modules. We will explains some of the main component that used in this research paper, this include: A Novel Voice Priority Queue (VPQ) Scheduler Architectures over WLANs IEEE802.11 Standards. Furthermore, we provide an experimental setup of the scheduler architectures considered in our research. We will explain all three stages of the VPQ and simulate VPQ scheduler components. VPQ based on the two types of traffics flow named as VoIP-Flow (VF) and Non-VoIP-Flow (NVF) as discussed in the section three . VPQ traffic will initialize from classification of enqueue traffic flows to dqueue traffic flows for end user.

4.1 NS-2 Simulations and Results Analysis Process

NS-2 is based on OTcl scripts to setup network topologies for VoIP over WLANs IEEE 802.11 Standards. Normally, NS-2 processes consists of the following steps: The Tcl simulation codes, Tcl interpreter, simulation results, pro-processing and finally, it will fall into two types of results formats trace file analysis and Network Animator (NAM). The NAM will perform graphically and interpreted into the trace file (.tr) and then analysis is shown in X-graph or graph tool. NS-2 supports the real-time, VoIP traffic schedulers over WLANs.

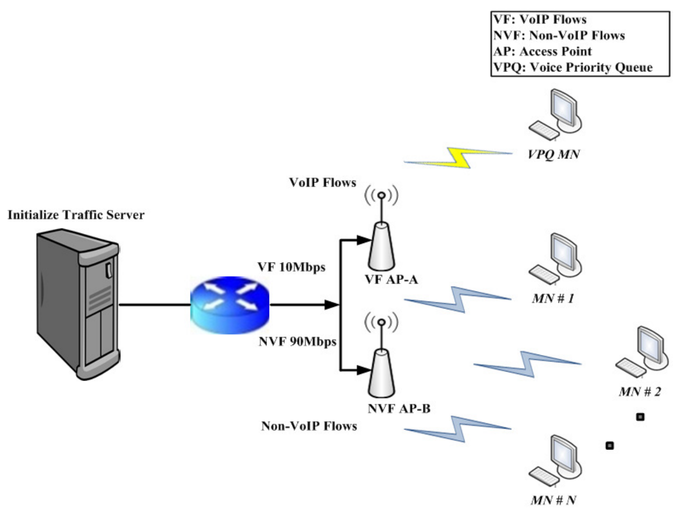

Figure 6 Simulation Setup of VPQ Scheduler Algorithm

We can create multiple topologies using nodes and packet forwarding technique. We can also connect the nodes to form links. NS-2 allows queue management and packet scheduling and queues shows the locations where packets may be held or dropped over IP-based networks. Its can supports is included for 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 we discussed in detail in the section two related work. Furthermore, NS-2 supports differentiated traffic services as like classification of traffic over WLANs. We implement our Novel Voice Priority Queue (VPQ) Scheduler Architectures over WLANs IEEE802.11 Standards. As above,Figure 6 shows the simulation setup. We have two type of traffic flow like VoIP flow and Non-VoIP flow. We have VoIP flow 1Mbps and Non-VoIP flow 10 Mbps traffic flow. The bottleneck for this set of simulation is the link between the two types of flow. The bandwidth of the links between nodes and routers is 11Mpbs and the propagation delay in 1ms.

5. Result & Discussion

In this section, includes our achieved results those based on various types of topologies simulations in Network Simulation – 2 (NS-2). We have shown stages of Voice Priority Queue (VPQ) using NS-2 simulation and test-bed experimental setup. We have described the topologies in simulation. We have compared the VPQ with Contention-Aware Temporally fair Scheduling (CATS), Temporally-Weighted Fair Queuing (T-WFQ) and controlled access phase scheduling (CAPS) traffic schedulers. Furthermore, we have also compared the VPQ Test-Bed with CATS, T-WFQ and CAPS schedulers. We will discuss in detail these results are as following. Figure 7 shows the total throughput (Mbps) of proposed VPQ, CATS and T-WFQ algorithms over WLAN network in the simulation. If we compared with previous all 4 flows the difference in the throughput measurement. In the all pervious flows we measured in (Kbps) and in the total throughput we measured in (Mbps) to evaluate the throughput results.

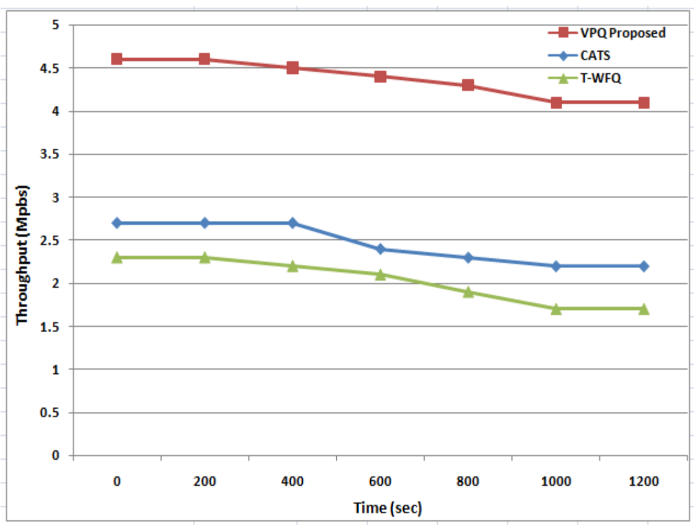

Figure 7. Total Throughput According to the Mobility of Mobile Station

The T-WFQ algorithm shows the lowest throughput among and T-WFQ throughput stated from 2.3 (Mbps) and its reached 1.7 (Mbps) on 1200 (sec). The CATS was higher than T-WFQ algorithm due to better performance over WLAN. The CATS throughput started from 2.7 (Mbps) and step by step decreased to 2.2 (Mbps). Our proposed VPQ algorithm better then both algorithms over WLANs and VPQ has higher throughput due to, classify the VF, NVF traffic, high date-rate flow and facilities more packets. The proposed VPQ started the throughput 4.6 (Mbps) and ended to 4.1 (Mbps).

Figure 8 shows the fairness index according to the mobility of mobile station. The fairness measured from 0 to 1 and above than 0 considered best. We also compared proposed VPQ, CATS and T-WFQ algorithm. We noticed in the T-WFQ algorithm start its fairness from 0.88 and ones time increased 0.9 on 600 (sec) then it's moved to down and against 1200 (sec), its fairness index 0.86.

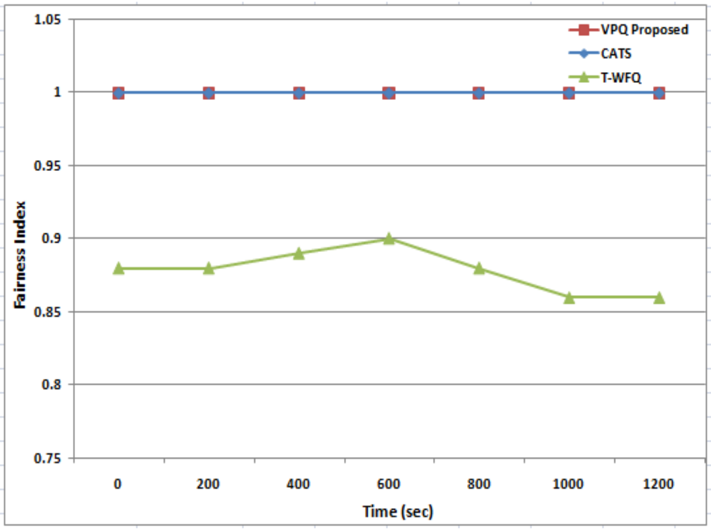

Figure 8. Fairness Index according to the Mobility of Mobile Station

As compared VPQ and CATS both shown best fairness index and they reached to 1 index from started and ended time of simulation. Figure 9 shows the throughput of the all flows over IP-based networks. In this figure, the total throughput (Mbps) of the proposed VPQ, CATS and T-WFQ algorithms over WLAN network in the simulation.

If we compared with previous all 4 flows the difference in the throughput measurement. In the all pervious flows we measured in (Kbps) and in the total throughput we measured in (Mbps) to evaluate the throughput results.

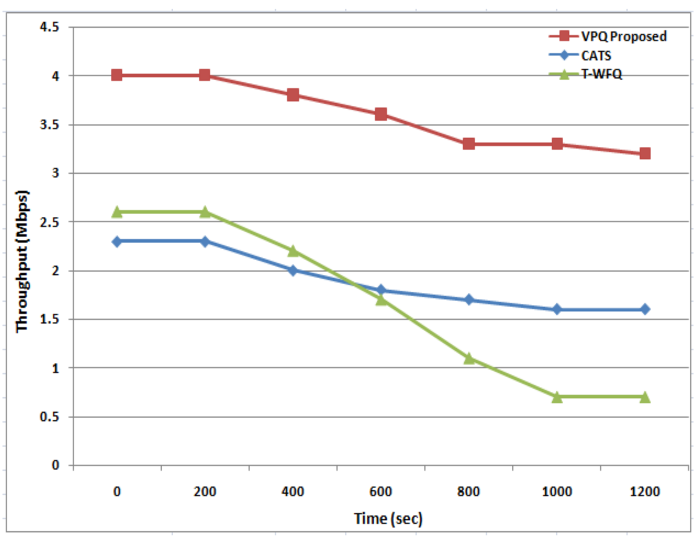

Figure 9. Total Throughput According to the Packet Size of Flow

The T-WFQ algorithm shows the lowest throughput among and T-WFQ throughput stated from 2.6 (Mbps) and its reached 0.7 (Mbps) on 1200 (sec). The CATS was higher than T-WFQ algorithm due to better performance over WLAN. The CATS throughput started from 2.3 (Mbps) and step by step decreased to 1.6 (Mbps). Our proposed VPQ algorithm better then both algorithms over WLANs and VPQ has higher throughput due to, classify the VF, NVF traffic, high date-rate flow and facilities more packets. The proposed VPQ started the throughput 4 (Mbps) and ended to 3.2 (Mbps) against 1200 (sec).

Figure 10 shows the fairness index according to the packet size of the flow. The fairness measured from 0 to 1 and above than 0 considered best. We also compared proposed VPQ, CATS and T-WFQ algorithm. We noticed in the T-WFQ algorithm start its fairness from 0.88 and it's moved to down 0.3 against 1200 (sec).

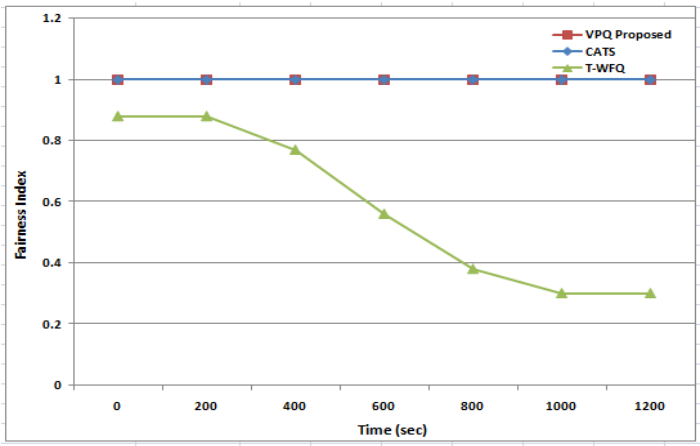

Figure 10 Fairness Index according to the Packet Size of Flow

As compared VPQ and CATS both shown best fairness index and they reached to same 1 index from started and ended time of simulation.We explained our simulation and experimental results in the graphs and tables. We also evaluated our Voice Priority Queue (VPQ) scheduler's results with most related scheduler and algorithms. We evaluated VPQ with Contention-Aware Temporally fair Scheduling (CATS), Decentralized-CATS, Decentralized-CATS+ and controlled access phase scheduling (CAPS), and Temporally-Weighted Fair Queuing (T-WFQ).

6. CONCLUSION

In this paper, we have presented a novel scheduling discipline called Voice Priority Queue (VPQ) scheduling algorithm. VPQ is simple, fair, and efficient with voice flow. In addition, it satisfies the unique requirement imposed by voice flow. In comparison to other scheduling algorithms of similar efficiency, VPQ has better throughput properties, as well as a higher efficiency. We expect that our proposed algorithm will be able to be offer fairness and delay guarantee in the future VoIP over WLANs. VPQ has satisfied these two requirements in CSMA/CA based 802.11 WLANs. The main function of VQP is that time should be fairly allocated despite the variable data rate and packet size of access point. In future work, we will investigate VPQ with Video traffic over WLANs on IEEE 802.11 Standards.

ACKNOWLEDGEMENTS

The authors would like to thank you for Universiti Utara Malaysia for support my research work.

References

[1] V. Soares, P. Neves, and J. Rodrigues, "Past, Present and Future of IP Telephony," *International Conference on Communication Theory, Reliability, and Quality of Service, Bucharest*, pp. 19–24, 05, July. 2008.

[2] R. Beuran "VoIP over Wireless LAN Survey," *Internet Research Center Japan Advanced Institute of Science and Technology (JAIST,) Research report. Asahidai, Nomi, Ishikawa, Japan,* pp. 1-40. 2006.

[3] 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, (AINA2010), Perth, Western Australia,* pp. 485-490, 20-23 April. 2010.

[4] L. Cai, Y. Xiao, X. Shen, and J. Mark, "VoIP over WLAN: Voice capacity, admission control, QoS, and MAC," *International Journal of Communication System, Published online in Wiley Inter-Science, Waterloo, Ontario, Canada,* Vol. 19, No. 4, pp. 491-508, May. 2006.

[5] V. Mockapetris, "Telephony's next act," *in IEEE Spectrum, Nominum Inc., Redwood City, CA, USA,* Vol.43, No. 4, pp. 15-29, 08, May. 2006.

[6] M. ALAkhras, "Quality of Media Traffic over Lossy Internet Protocol Networks: Measurement and Improvement," *PhD thesis, Software Technology Research Laboratory, De Montfort University, United Kingdom,* 2007.

[7] S. Oueslati, J. Roberts, "A new direction for quality of service: Flow-aware networking", *Proceedings of NGI 2995, Rome*, pp. 1-8, June 2005.

[8] A. Kortebi, S. Oueslati and J. Roberts; "Implicit Service Differentiation using Deficit Round Robin", *ITC19, Beijing,* Aug 2005.

[9] S. Kanhere, H. Sethu, and A. Parekh, "Fair and efficient packet scheduling using elastic round robin", *IEEE transactions on Parallel and Distributed Systems,* vol.13, pp. 324-336, Mar 2002.

[10] A. Striegel and G. Manimaran, "Dynamic class-based queue management for scalable media servers", *Real-Time Technology and Applications Symposium IEEE,* pp. 228 – 236, June 2000.

[11] Wu-Chang Feng, D.D. Kandlur, D. Saha, K.G. Shin, "Stochastic fair blue a queue management algorithm for enforcing fairness", *Proceeding of IEEE, INFOCOM 2001,* April 2001.

[12] Wu-Chang Feng, K.G. Shin, D.D. Kandlur, D. Saha, "The BLUE active queue management algorithms", *IEEE/ACM Transaction on Networking August*, 2002.

[13] H. Chong and H. Matthews, "Comparative analysis of traditional telephone and voice-over-Internet protocol (VoIP) systems," *Electronics and the Environment, 2004. Conference Record. 2004 IEEE International Symposium, Pittsburgh, PA, USA,* pp. 106-111, 24, May. 2004.

[14] A. Lindgren, A. Almquist, and O. Schelen, "Evaluation of quality of service schemes for IEEE 802.11 wireless LANs," *26th Annual IEEE Conference on Local Computer Networks, Tampa, FL, USA,* pp. 348-351, 06, August. 2002.

[15] L. X. Cai, X. Ling, X. Shen, J. Mark, and L. Cai, "Supporting voice and video applications over IEEE 802.11n WLANs," *Journalon Wireless Networks, Springer Science, Victoria, BC, Canada,* Vol. 15, No. 4, pp. 443-454, November. 2009.

[16] X. Ling, L. X. Cai, J.W. Mark, and X. Shen, "Performance analysis of IEEE 802.11 DCF with heterogeneous traffic," *Process in IEEE CCNC'07, Las Vegas, Nevada,* pp. 49-53, January. 2007.

[17] A. Rahman and O. Ghazali, "TCP-Friendliness of Modified Explicit Rate Adjustment," *International Conference on Network application, Protocols and Services, Malaysia,* Vol. 1, No.1, pp. 1-5, 22, November. 2008.

[18] H. Jin, R. Hsu, and J. Wang, "Performance comparison of header compression schemes for RTP/UPD/IP packets," *Proceedings of the IEEE Wireless Communications and Networking Conference, Atlanta, GA, USA,* Vol. 3, pp. 1691–1696. 2004.

[19] S. Ehlert, G. Zhang, and T. Magedanz, "Increasing SIP Firewall Performance by Ruleset Size Limitation," *IEEE PIMRC 2008, VoIP TechnologiesWorkshop, Cannes, France*, September. 2008.

[20] G. Zhang, S. Ehlert, T. Magedanz, and D. Sisalem, "Denial of Service Attack and Prevention on SIP VoIP Infrastructures Using DNS Flooding," *Principles, Systems and Applications of IP Telecommunications (IPTComm2007), New York, USA,* pp. 57-66, July. 2007.

[21] S. Shankar, D. Paven and P. Wienert, "Optimal packing of VoIP calls in an IEEE 802.11a/e WLAN in the presence of QoS Constraints and Channel Errors," *Global Telecommunications Conference, 2004. GLOBECOM '04. IEEE,* Vol. 5, pp. 2974-2980, 17, January. 2005.

[22] 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) Journal, Calgary, Canada, Advances in Computer Science and Engineering (ACSE)* 2009 Phuket, Thailand. 2009. http://www.actapress.com/Abstract.aspx?paperId=34875

[23] 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,* June, 17. 2010.

[24] 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.

[25] 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, January, 24. 2009.

[26] I. Aad, Q. Ni, C. BArakat and T. Turletti, "Enhancing IEEE 802.11MAC in congested environment," *Preprint submitted to Elsevier Science*, *IEEE ASWN'04, Bostan, Massachustts, USA,* pp. 01-19, 10, December. 2004.

[27] L. Xiong and G. Mao, "Saturated throughput analysis of IEEE 802.11e EDCA," *The International Journal of Computer and Telecommunications Networking, Elsevier, B.V. Sydney, NSW, Australia,* Vol. 51, No.11, pp. 3047-3068, 08, August. 2007.

[28] A. Zahedi and K. Pahlavan, "Capacity of a Wireless LAN with Voice and Data Services," *IEEE Transactions Communications, MA, USA*, Vol.48, No.7, pp 1160–1170, 06, August. 2002.

[29] Y. Xiao, "IEEE 802.11n: enhancements for higher throughput in wireless LANs," *IEEE Wireless Communications, TN, USA,* Vol. 12, No. 6, pp. 82-91, 19, December. 2005.

[30] A. Floros, M. Avlonitis AND P. Vlamos "Frequency-Domain Stochastic Error Concealment for Wireless Audio Applications," *Mobile Networks and Applications, SpringerLink, Corfu, Greece,* Vol. 13 No. 3, pp. 357-365, 4, August. 2008.

[31] K. Yasukawa, A. Forte and H. Schulzrinne "Distributed Delay Estimation and Call Admission Control in IEEE 802.11 WLANs," *Proceding of the 2009 IEEE International Conference on Communications, IEEE ICC 2009, Ericsson Research Japan,* pp. 5057-5062, 18, June. 2009.

[32] P. Wang, and W. Zhuang "A Token-Based Scheduling Scheme for WLANs Supporting Voice/Data Traffic and its Performance Analysis," *IEEE Transactions on Wireless Communications, Waterloo, Ontario, Canada,* Vol. 7, No 4, pp.1-11, April. 2008.

[33] P. Wang, H. Jiang, and W. Zhuang, "Capacity improvement and analysis for voice/data traffic over WLANs," *IEEE Transactions. Wireless Communications, Waterloo, Ontario, Canada,* Vol. 6, No. 4, pp. 1530-1541, April. 2007.

[34] F. Anjum, M. Elaoud, D. Famolari, A. Ghosh, R. Vaidyanathan, A. Dutta, and P. Agrawal, "Voice Performance in WLAN Networks-An Experimental Study," *In Proceedings of the IEEE Conference on Global Communications (GLOBECOM), Rio De Janeiro,* pp. 3504-3508, 2003.

[35] Hung. Liu, and J. Wu, "Packet telephony support for the IEEE 802.11 wireless LAN," *IEEE Communications Letters, Taipei, Taiwan,* Vol. 4, No. 9, pp. 286-288, September. 2000.

[36] A. Kopsel and A. Wolisz, "Voice transmission in an IEEE 802.11 WLAN based access network," *Processing in WoWMoM, Rome, Italy,* pp. 24-33, July. 2001.

[37] D. Leith, P. Clifford, D. Malone, and A. Ng, "TCP Fairness in 802.11e WLANs," *IEEE Communications Letters, Hamilton, Ireland,* Vol. 9, No. 12, pp. 1-3, December. 2005.

[38] T. Li, Qiang Ni, D. Malone, D. Leith, Y. Xiao and T. Turletti, "Aggregation with Fragment Retransmission for Very High-Speed WLANs," *IEEE/ACM Transactions on Networking (TON), Piscataway, NJ, USA,* Vol. 17, No. 2, pp. 591-604, April. 2009.

[39] N. Shetty et al. "Voice Capacity under Quality Constraints for IEEE 802.11a based WLANs," *Proceedings of the 2006 international conference on Wireless communications and mobile computing, Vancouver, British Columbia, Canada*, pp. 1345-1350, July, 6. 2006.

[40] G. Cantieni, Qiang Ni, C. Barakat and T. Turletti "Performance analysis under finite load and improvements for multirate 802.11," *Computer Communications, Science Direct, Elsevier B.V,* Newton, MA, USA, Vol. 28, No. 10, pp. 1095-1109, June. 2005.

[41] I. Aad, Q. Ni, C. Barakat and T. Turletti, "Enhancing IEEE 802.11 MAC in Congested Environments," *Elsevier Computer Communications Journal, Sophia Antipolis, France,* Vol. 28, No.14, pp. 1605-1617, September. 2005.

[42] P. Clifford, K. Duffy, D. Leith and D. Malone, "On Improving Voice Capacity in 802.11Infrastructure Networks," *2005 International Conference on Wireless Networks, Communications and Mobile Computing,* pp. 214-219, 2005.

[43] Q. Ni, L. Romdhani and T. Turletti, "A Survey of QoS Enhancements for IEEE, Wireless LAN," *Journal of Wireless Communications and Mobile Computing, John Wiley and Sons Ltd. Chichester, UK,* Vol. 4, No. 5, pp. 547-566, 2004.

[44] H. M. Chong and H. Matthews, "Comparative analysis of traditional telephone and voice-over-Internet protocol (VoIP) systems," *The IEEE International Symposium on Electronics and the Environment, IEEE Computer Society, Pittsburgh, PA, USA,* pp. 106-111, 24, May. 2004.

[45] M. Abusubaih, S. Wiethoelter, J. Gross, and A. Wolisz, "A new access point selection policy for multi-rate IEEE 802.11 WLANs," *International Journal of Parallel, Emergent and Distributed Systems, Berlin, Germany,* Vol. 23, No. 4, pp 1-20, August. 2008.

[46] K. Brian and D. Mark, “An Experimental Analysis of the Call Capacity of IEEE 802.11b Wireless Local Area Networks for VoIP Telephony,” *Irish Signals and Systems Conference, IET, Dublin, Ireland,* pp. 283 – 287, 12, March. 2007.

[47] J. Mirkovic, G. Orfanos, H. Reumerman, and D. Denteneer, “A MAC protocol for MIMO based IEEE 802.11 wireless local area networks," IEEE *Wireless Communications and Networking Conference, 2007.WCNC 2007, Kowloon,* pp. 2131-2136, 04, June. 2007.

[48] C. Wang and H. Wei, “IEEE 802.11n MAC Enhancement & Performance Evaluation,” *Journal ofMobile Networks and Applications, Springer Science, Taipei, Taiwan,* Vol. 14, No.6, pp.761-771, 09, January. 2009.

[49] L. Xiong and G. Mao, “Performance Analysis of IEEE 802.11 DCF with Data Rate Switching,” *IEEE Communications Letters, Sydney, NSW, Australia,* Vol. 11, No.9, pp. 759-761, 09, September. 2007.

[50] Q. Cao, T. Li, Tianji and D. Leith “Achieving fairness in Lossy 802.11e wireless multi-hop Mesh networks,” *Third IEEE International Workshop on Enabling Technologies and Standards for Wireless Mesh Networking MESH,* Macau *SAR, P.R. China,* pp. 1-7, 10, November. 2009.

[51] P. Dini, O. Font-Bach, and J. Mangues-Bafalluy, “Experimental analysis of VoIP call quality support in IEEE 802.11 DCF,” *Communication Systems, Networks and Digital Signal Processing, 2008. CNSDSP2008. 6th International Symposium*, pp. 443 – 44729, August. 2008.

[52] T. Li, Q. Ni, T. Turletti and Y. Xiao “Performance analysis of the IEEE 802.11e block ACK scheme in a noisy channel*”, Broadband Networks, 2005. BroadNets 2005. 2nd International Conference. Hamilton Inst. Ireland,* Vol. 1, pp. 511-517, 13, February. 2006.

[53] Q. Ni, T. Li, T. Turletti, and Y. Xiao. "Saturation Throughput Analysis of Error-Prone 802.11 Wireless Networks," *Wiley Journal of Wireless Communications and Mobile Computing (JWCMC), INRIA, France,* Vol. 5, No. 8, pp. 945-956. December. 2005.

[54] S. Garg and M. Kappes, “Can I add a VoIP Call,” *Communications, 2003. ICC '03. IEEE International Conference on, Avaya Labs, Basking Ridge, NJ, UAS,* pp. 779-783, Vol. 2, 20, June. 2003.

[55] D. Hole and F. Tobagi, “Capacity of an IEEE 802.11b Wireless LAN Supporting VoIP,” *IEEE ICCCalifornia, USA,* pp. 196-201, 2004.

[56] L. Cai, X. Shen, J. Mark, L. Cai, and Y. Xiao, “Voice Capacity Analysis of Wlan with Unbalanced Traffic,” *IEEE Transactions On Vehicular Technology,Ontario, Canada,* Vol. 55, No. 3, pp. 752-761, 05, June. 2006.

[57] J. Deng and H. Yen, “Quality-of-service provisioning system for multimedia transmission in IEEE 802.11 wireless LANs,” *IEEE Journalon Selected Areas in Communications, Taipei, Taiwan,* Vol. 23, No. 6, pp. 1-6, 2005.

[58] Y. Xiao and J. Rosdahl, “Throughput and Delay Limits of IEEE 802.11,” *IEEE Communication Letters, USA,* Vol. 6, No. 8, pp. 355-357, 07, November. 2002.

[59] S. Floyd and M. Speer, “Experimental Results for Class-Based Queueing,” *IEEE/ACM Transactions on Networking, USA,* pp. 1-13, 11, November. 1998.

[60] D. Demarch and L. Becker, “An integrated scheduling and retransmission proposal for firm real-time traffic in IEEE 802.11e. *Proceeding in the 19th Euromicro Conference on Real-Time System,*

Brazil, pp. 146.158, 2007.

[61] Y. Seok, T. Kwon, Y. Choi, and J. Bonnin, "Temporal fair-ness guarantee in multi-rate wireless LANs for per-flow protection," *Wireless Networks, Springer Science, business Media,* Vol. 13, No. 2, pp. 237-258, April. 2007.

[62] S. Keshav, *An Engineering Approach to Computer Networking: ATM networks, the Internet, and the Telephone Network*, Addison Wesley, 1997.

[63] J. Nagle, "On Packet Switches with Infinite Storage," *IEEE Transactions on Communications, IEEE Communications Society, Stanford, CA, USA,* Vol. 35, No. 4, pp. 435-438, April. 1987.

Authors

Dr.KashifNisar received his first degree Bachelor in Computer Science from Karachi, Pakistan and his MS degree in Information Technology from the Universiti Utara Malaysia (UUM). He received his PhD degree in Computer Networks (specializing in WLANs, VoIP, IPv6 and he proposed a novel Voice Priority Queue (VPQ) scheduling system model) from the UniversitiTeknologi PETRONAS (UTP) in the Malaysia. Dr.Kashif's research interests are in multi-disciplinary areas, including Real-time Multimedia Technologies, Network Performance, Network Traffic Engineering, Wireless and Mobile Networking, Local Area Networking, Telecommunications/Network Management, Wireless Sensor Networks (WSNs), Mobile Ad Hoc Networks, Audio/Video Transport Extensions, Call Control UUI Service for SIP, Audio/Video Transport Payloads, VoIP, IPv6, and evaluation of wireless networks. He is currently working as a Senior Lecturer at School of Computing, Collage of Arts and Science, Universiti Utara Malaysia, Malaysia. He is also serving as a Senior Member of InterNetWorks Research Group.

AssocProfDr.Suhaidi B Hassan received his BSc degree in Computer Science from Binghamton University, New York (USA) and his MS degree in Information Science (with concentration in Telecommunications and Networks) from the University of Pittsburgh, Pennsylvania (USA). He received his PhD degree in Computing (specializing in Networks Performance Engineering) from the University of Leeds in the United Kingdom.Dr.Suhaidi's research interests are in multi-disciplinary areas, including Computer Networks: Protocol Engineering, Network Quality of Service (QoS), Internet Architecture and Technologies, Network Performances, Network Traffic Engineering, Network Security, Wireless and Mobile Networking, Local Area Networking Telecommunications/Network Management.Computer Systems: {Wireless, Mobile, Distributed, Secure, Ubiquitous, High Performance, Grid, Cloud} Computing.He is currently anAssociate Professor & Assistant Vice Chancellor at College of Arts and Science, Universiti Utara Malaysia, Malaysia. He is also serving as a Chairman of InterNetWorks Research Group.