A Voice Priority Queue (VPQ) Fair Scheduler for the VoIP over WLANs

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Abstract—Transmission of VoIP over packet switching networks is one of the rapidly emerging real-time Internet Protocol. The real-time application of the Voice over Internet Protocol (VoIP) is growing rapidly for it is more flexible than the traditional Public Switched Telephone Networks systems (PSTN). Meanwhile, the VoIP deployment on Wireless Local Area Networks (WLANs), which is based on IEEE 802.11 standards, is increasing. Currently, many schedulers have been introduced such as Weighted Fair Queueing (WFQ), Strict Priority (SP) General processor sharing (GPS), Deficit Round Robin (DRR), and Contention-Aware Temporally fair Scheduling (CATS). Unfortunately, the current scheduling techniques have some drawbacks on real-time applications and therefore will not be able to handle the VoIP packets in a proper way. The objective of this research is to propose a new scheduler system model for the VoIP application named Voice Priority Queue (VPQ) scheduler. The scheduler system model is to ensure efficiency by producing a higher throughput and fairness for VoIP packets. In this paper, only the First Stage of the VPQ packet scheduler and its algorithm are presented. Simulation topologies for VoIP traffic were implemented and analyzed using the Network Simulator (NS-2). The results show that this method can achieve a better and more accurate VoIP quality throughput and fairness index.

***Keywords*- Scheduler, VPQ, VoIP; WLANs**

I. INTRODUCTION

The VoIP allows users to make IP-based calls over the global networks. In IP-based networks, analogue voice signals are digitized before being transmitted over the network. While traveling through the network, the digitized packets are directed in an efficient way to reach their destination. Normally, voice packets are received out of the order they have been sent in. Therefore, at the receiver side the packets are rearranged into the correct order before they are converted back into the analogue voice signals to be played out [1-4].

Currently, there are approximately 1 billion fixed telephony lines and 2 billion mobile-phone lines in the world. This increasing number shows that the next generation technology is moving ahead to IP network based protocols such as the VoIP [5]. Furthermore, the VoIP over Wireless Local Area Network (WLAN) which transmits IP-based telephone calls over WLANs might be a leading application in collaboration with the 3rd Generation (3G) mobile network [6]. On the other hand, 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. That leads to a number of research solutions to be proposed in order to satisfy different Quality of Service (QoS) requirements [7-12].

A. *VoIP Protocol Architecture*

Figure. 1 demonstrates the fundamental stack architecture needed to implement a VoIP network system over WLANs [13-14]. The VoIP is a real-time application that transmits voice using different protocols of the WLAN such as the Real-Time Transport Protocol (RTP), User Datagram Protocol (UDP) and Internet Protocol (IP) (RTP/UDP/IP) [15-18]. Each VoIP packet includes headers of RTP (12 bytes), UDP (8 bytes) and IP (20 bytes). In addition, a data-link layer adds a Medium Access Control (MAC) header (34 bytes). All these added headers accumulate 74 bytes of overhead in the voice packet.

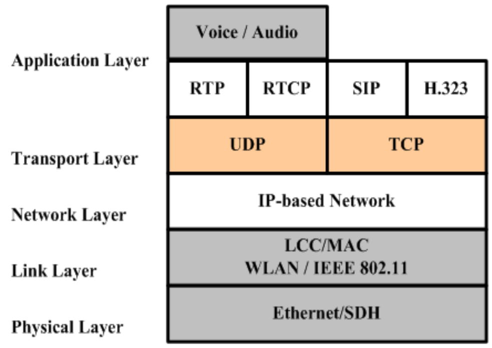

Figure 1. VoIP Protocol Arcihitecture

Besides this, the Session Initiation Protocol (SIP) is designed to handle multimedia call setups while H.323 is designed to allocate IP-based phones on the public telephone network to talk to PC-based phones over IP-based networks [19-20]. H.323 is defined by the International Telecommunications Union (ITU). 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 networks [21-25].

B. *VoIP over WLANs*

Basically, WLAN is the most promising technology among the wireless networks as it facilitates high-rate voice services at much less cost and more flexibility [26-29]. The VoIP over WLANs has witnessed a fast growth in the world of communication [6]. This adaptation is growing due to the fact that the VoIP over WLAN is more flexible than traditional Public Switched Telephone Network systems (PSTN) [30]. In addition, WLANs is rapidly expanding [31] in its deployment in campuses, hotels, airports, health care facilitates, commercial use, education and various other industries. WLANs also support the transmission of audio, voice and video conferencing packets [32-36].

In WLANs, there are two essential kinds of service architectures: ad-hoc architecture and infrastructure architecture. In ad-hoc architecture, a station (STA, a mobile node) can connect directly to any of the other stations in the same network without a need for an Access Point (AP) and without any required connectivity to a wired backbone network [37]. However, in the infrastructure structure of WLAN, the STA communicates with another STA via an AP and the network can have a connection to a wired backbone network. The infrastructure architecture will be studied in this paper, in which the VoIP traffic is transmitted between stations via an AP. Additionally, there are number of industrial standards of AP defined for WLANs where each AP can maintain a restricted number of parallel voice nodes [38-46].

C. *VoIP, IEEE 802.11 MAC and WLANs Standards*

The IEEE 802.11 WLAN, also known as the wireless Ethernet, plays an important function in the future-generation networks. The Link layer (LL) of the network stack as defined by IEEE 802.11 WLAN standards is divided into a Logical Link Control (LLC) sub-layer and a Medium Access Control (MAC) sub-layer. The MAC sub-layer is categorized into two core functions, the Distributed Coordination Function (DCF) and the Point Coordination Function (PCF) [47-49]. The IEEE 802.11 WLANs support both contention-based DCF and contention-free PCF functions. Furthermore, the DCF function applies the Carrier Sensing Multiple Access/Collision Avoidance CSMA/CA as an access method [50].

D. *Problem Statement*

QoS is considered a main issue in VoIP systems. The VoIP is a delay sensitive application as it requires packets to arrive at their destination on time with the least delay. The VoIP also requires a higher throughput, less packet loss and a higher fair index on the network. Furthermore, IP based networks were initially designed for transmitting data packets and it was not considered a real time application. However, with the growing increase of internet users and their demands, today the network shares the bandwidth among web browsing, email, voice data, and video applications on the same network. Carrying different types of data which increases the load on the network may cause a bottleneck in the network. A bottleneck topology of mix-mode traffic over a WLAN is shown in Figure. 2 [51]. Due to the bottleneck and the competition among different kinds of traffic flow on the network, packets of voice streaming may experience dropping in the scheduling buffer. Thus, the quality of the VoIP application cannot be guaranteed. Therefore, many researches focus on improving QoS by addressing the problem with proposing new VoIP scheduling algorithms that overcome the drawbacks of the available algorithms which were designed mainly for data transmission.

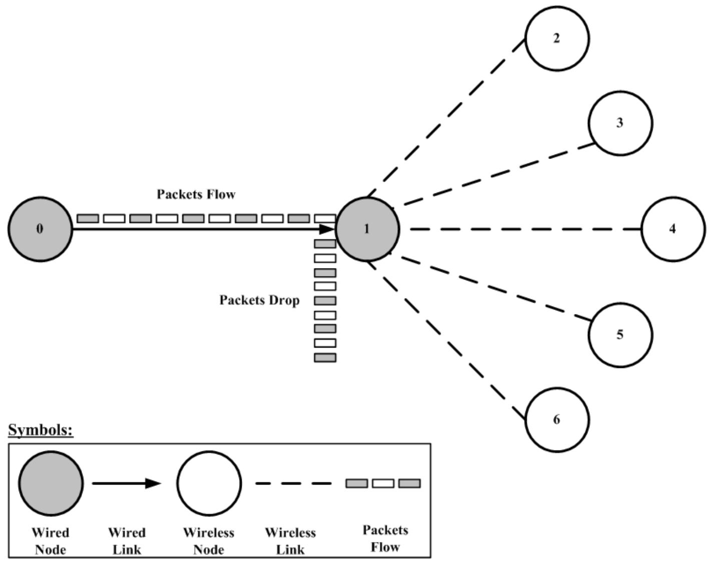

Figure 2 Bottleneck topology of mix-mode traffic

Through the past decades many schedulers were introduced to improve the performance of real-time applications. These schedulers can be classified into three groups; packet-based schedulers, frame-based packet schedulers, and regulative packet schedulers. Yet, there is still a need for a new voice scheduling algorithm to solve the bottleneck issue in order to improve the QoS of the VoIP. The new method should be efficient and fair, give a high throughput, be bandwidth-guaranteed and enhance the performance of the VoIP over WLAN. Thus, a new scheduling algorithm is proposed in this research paper to meet these requirements. The proposed method aims to achieve better acceptable results for VoIP applications.

E. *Aim and Objectives*

The aim of this research paper is to propose an efficient scheduler and algorithm that supports the VoIP application on WLANs. A fundamental related work will be studied to examine the available schedulers with their outcomes and their drawbacks. A new scheduler and algorithm will be then introduced to improve the real-time traffic scheduling by resolving the previously stated issue, thereby, enhancing the performance of VoIP applications on WLANs. The proposed techniques will be simulated, evaluated and compared with some related algorithms of real-time applications. In this research the specific objectives are:

- To classify VoIP-Flow (VF) traffic and Non-VoIP-Flow (NVF) traffic on a WLAN.
- To develop a new Voice Priority Queue (VPQ) s and algorithm for VoIP traffic that can fulfill VoIP scheduling requirements.
- To compare, evaluate, validate and verify the VPQ scheduler and algorithm with other schedulers of related work

The rest of the paper is organized as the following: in section II, related work of different scheduling algorithms is discussed initiating their limitation when applying multimedia applications; the methodology and the new VoIP scheduling algorithm are explained in section III. In section IV, the simulation experimental setup is demonstrated in which the new VoIP scheduling algorithm is compared with other related scheduling algorithms. Finally, section V discusses the simulation results and section VI concludes this paper with remarking on some future research work.

II. RELATED WORK

The high-speed packet-switched networks are essential research areas. The VoIP over WLAN is one of the most application technologies to utilize high-speed packet-switched networks. Furthermore, there are some scheduling algorithms designed to support packet scheduling in wired and wireless networks. A few of them are listed as the following: Class Based Queue (CBQ), Faire Queue (FQ), Weight Faire Queue (WFQ), Generalized Processor Sharing (GPS), Worst-case Fair Weighted Fair Queueing (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) [52]. Some of these related schedulers will be briefly discussed in the following sections.

A. *Class Based Queue (CBQ)*

Floyd et al. [53] introduced the Class Based Queue (CBQ) scheduling which is a class based algorithm. The CBQ scheduler is a combination of classifier, estimator, selector and over limit processor. CBQ shares the bandwidth among different traffic classes. This technique also provides a functionality of link-sharing resources as illustrated in Figure 3. Furthermore, CBQ manages the link-sharing bandwidth rations for all classes, maintains each queue and provides fair link-sharing. CBQ is implemented with a gateway technique and fulfils the range of service and link-sharing. Hence, CBQ provides the solution of having multiple types of traffic flow on a network.

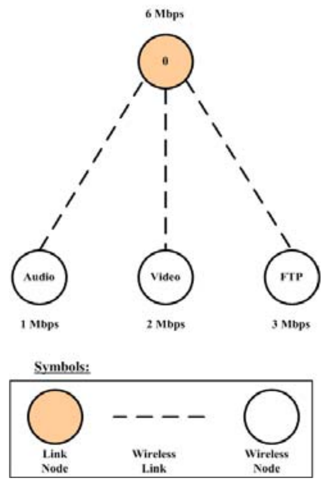

Figure 3 Priority and structure of link-sharing allocation [54]

CBQ divides the bandwidth or allocates the link-sharing according to the traffic requirement. In Fig. 3 above, CBQ has 6 Mbps bandwidth shared among three different kinds of traffic flows like audio 1Mbps, video 2Mbps and File Transfer Protocol (FTP) 3Mbps over wireless nodes. It has been observed that CBQ is the best solution for data traffic.

However, there are drawbacks of CBQ such as when there is bursty traffic on a real-time application, it will cause more delay. CBQ does not consider the delay of packets whereas the delay of packets should be reduced and scalable for the VoIP application. The real-time application requires a specific buffer to control the bursty traffic on IP-based networks as the bulky buffer introduces a longer delay for playback.

B. *Weighted Fair Queue (WFQ)*

The Weighted Fair Queue (WFQ) scheduling algorithm was introduced in 1989 by A. Demers et al. [54]. WFQ is a sort-based packet scheduling algorithm with the latest updating of Generalized Processor Sharing (GPS). WFQ offers N queues simultaneously with different bandwidth service rates as it assigns each queue a weight and a 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 in calculating the timestamps and it adds a complexity of $O(\log n)$, where n is the number of flows. Nonetheless, due to the slow process of sorting among the timestamps, WFQ is not a suitable scheduler for real-time applications.

C. *Contention-Aware Temporally fair Scheduling (CATS)*

Seok et al. [55] introduced the Contention-Aware Temporally fair Scheduling (CATS) which is a packet-based algorithm that offers fairness among traffic flows on WLANs. CATS is introduced for equal time sharing for each flow on the network. CATS arranges the packets scheduling order after a virtual finish time. In addition, the scheduler is capable of performing in a multi-rate WLAN environment and provides a solution for the Carrier Sense Multiple Access / Collision Avidness (CSMA/CA) technique.

The drawbacks of Contention-Aware Temporally fair Scheduling (CATS) are as follows; CATS is based on the Generalized Processor Sharing (GPS) that is applied on wired based networks and based on the fluid flow mechanism. GPS is based on a fixed link of capacity and cannot facilitate the pre-flow process. CATS applies T-GPS and cannot perform the required calculation for each flow on the WLAN. Meanwhile, the Temporally-Weighted Fair Queueing (T-WFQ) that is also based on GPS has less performance than CATS.

Being an emerging technology, the WLAN supports several applications simultaneously such as the VoIP, IPTV and High-Performance Video-Conferencing (HP-VC). Therefore, fairness among these applications is required. On the other hand, the scheduling technique illustrates the flow of traffic on WLANs. Furthermore, the system model of scheduling techniques play a major role in WLANs to fulfill the essential requirements of VoIP traffic flow on the network. Therefore, to guarantee the requirements of the VoIP over WLAN, the Quality of Service (QoS) requires a responsible scheduling system model, algorithms, efficient traffic and enhanced voice traffic flow on WLANs.

III. METHODOLGY

As it has been discussed in related work, a number of related schedulers have been proposed to support voice traffic flow over IP-based networks. However, these related schedulers support traffic flows with limited services and the requirements of real-time application are still not fully met in particular on WLAN environments. The goal of this research is to propose an efficient traffic scheduler for VoIP traffic flow over WLAN named as the VPQ scheduler. The development of the VPQ scheduler is divided into three stages which are:

- The First stage of the Voice Priority Queue (VPQ) Scheduler
- The Second stage of the Voice Priority Queue (VPQ) Scheduler
- The Third stage of the Voice Priority Queue (VPQ) Scheduler

Furthermore, the VPQ scheduler aims to achieve high throughput and fairness for VoIP applications over WLANs. It is scalable, able to offer link-sharing of bandwidth and able to tolerate the status of changing traffic queues. It mainly classifies the traffic into VoIP-Flow (VF) and Non-VoIP-Flow (NVF). In the second stage of the VPQ scheduler bursty Virtual VoIP Flow (Virtual-VF) component will be designed to handle the bursty traffic, while the third stage of the scheduler is designing a Switch Movement (SM) technique. In this paper, the focus is only on the first stage of VPQ scheduler. The first stage is explained in the following paragraphs.

As illustrated in Figure 4, VPQ scheduler system model is based on initializing 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.

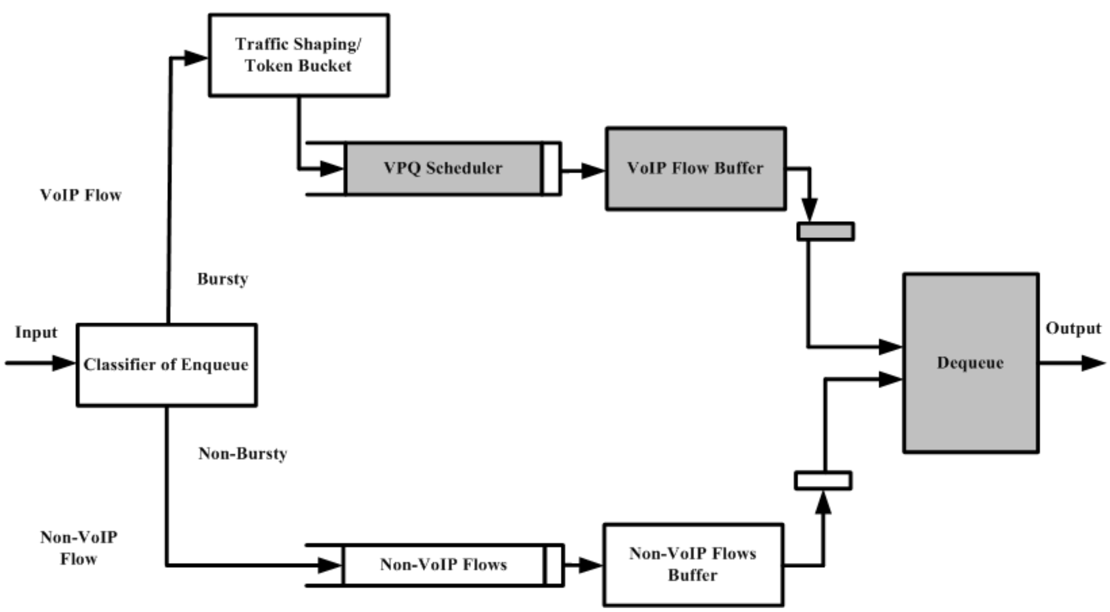

Figure 4 The Voice Priority Queue System Model

The classification of enqueue traffic flows is sorted-based which checks the index of each incoming packet. The classification architecture supports a mechanism that works similar to Differentiated Services (DiffServ) as it classifies packets into different traffic flows depending on their destination packet types. After classification, voice traffic flows are sent to the VF traffic shaping and token bucket flow. The shaping controls the amount of the flow and sends the traffic flow to the token bucket flow. The token bucket is applied to bursty traffic to regulate the maximum rate of traffic on WLANs. The traffic shaping and token bucket send the VF flows to the Voice Priority Queue (VPQ), the main component of the VPQ scheduler, for processing.

The VPQ forwards these packets to the VoIP Flow Buffer (VFB). The VFB is a temporary buffer for VF traffic while the Non-VoIP Flow Buffer is for NVF traffic. Moreover, the Base Station (BS) has instantaneous and perfect understanding of the VF and NVF channel's position due to their weight, energy and priority queue. Finally, VF and NVF flows are dequeued to the destination. In addition, the designed algorithm of the VPQ scheduler is demonstrated in Figure. 5.

```
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 No- 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 Flows (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 Flows*

Figure 5 The Voice Priority Queue Algorithm

IV SIMULATION SETUP

In this section, the simulation of our scheduler algorithm will be discussed. The VPQ is based on two types of traffic flows named as the VoIP-Flow (VF) and the Non-VoIP-Flow (NVF) as discussed in the previous section. In the simulation, the VPQ traffic will be initiated from classification of enqueue traffic flows up to the phase of dequeue traffic flows to be sent to the end user. The simulation of the Voice Priority Queue (VPQ) scheduler on a WLAN is implemented using the NS-2 [57] and validation and verification of the developed simulation modules will be performed.

The Network Simulation-2 (NS-2) is based on OTcl scripts to setup network topologies such as the VoIP over IEEE 802.11 WLANs. Generally, a NS-2 simulation consists of the following steps: The Tcl simulation codes, Tcl interpreter, simulation results and pre-processing. The obtained results in the NS-2 are generated in two formats; trace file analysis and Network Animator (NAM). The NAM results format displays simulation graphically and interprets results into the trace file (.tr) and then the analysis is shown in X-graph or graph tool.

In the simulation of the Voice Priority Queue (VPQ) scheduler, the scenario consists of two wired nodes connected with two Access Point (APs) nodes or base-stations (BS). The two APs are classified for traffic of the VoIP-Flow (VF) and Non VoIP Flow (NVF). Next, each AP is connected respectively to a VF mobile node and NVF mobile nodes numbered from queue one to a number of queues. In this section, the simulations have been performed in both types of traffic, the VF and NVF modes. However, only simulation analysis for the VF flows is discussed. There are two analysis parameters that have been focused on, fairness and maximum achievable throughput. A description of these two parameters is given in Table 1.

Table 1 Simulation Parameters and metrics [56]

Metrics Name	Description
Throughput	Throughput is a term that defines the amount of traffic correctly received at the receiver end. In order to calculate the average throughput of the network: Add the number of Packets, multiply it by the packet size and then divide by the simulation time.
Fairness Index	Fairness can be calculated based on the fairness of time shared which is rated between the 0 and 1 digits. The higher value of the fairness index means the better fairness between traffic. The equation is as below: $fairness\ index = \frac{(sum\ (Xi))**2}{number\ of\ flows sum\ (Xi**2)}$ [8] Where is the relative allocation.

Furthermore, the simulation topology is shown in Figure 6 below. It is based on a backend node which is connected with a wired network and two frontend nodes which are APs. The AP nodes are similar to gateways between wired and wireless nodes which permit packets to be exchanged between the two kinds of networks. Similarly, in the simulation of VPQ scheduler, the VPQ topology includes two wired nodes named node-0 and node-1. The node-0 provides the initialization of traffic flow and node-1 provides the classification of traffic. There are two more nodes added as gateway nodes named node-2 and node-3. Node-2 provides a VoIP Flow (VF) while node-3 provides a Non VoIP Flow (NVF).

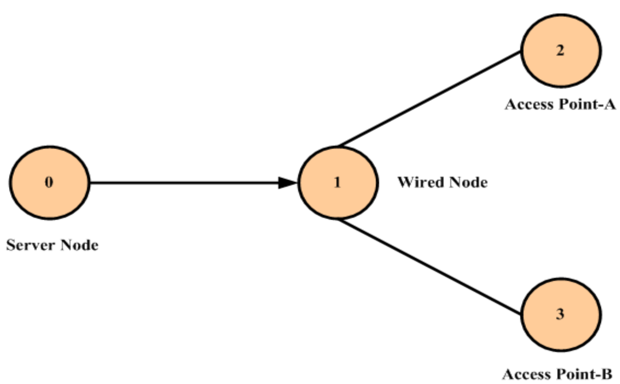

Figure 6 Simulation topology of the NS-2

In Figure 7, the VPQ scenario has been extended to include wireless nodes. Meanwhile, the wireless nodes named as node-4, node-5 and node-6 are the number of nodes on the wireless networks. Node-4 is a mobile node that provides traffic flow between the Wireless-VF (WVF) and the VF-Wired (VFW).

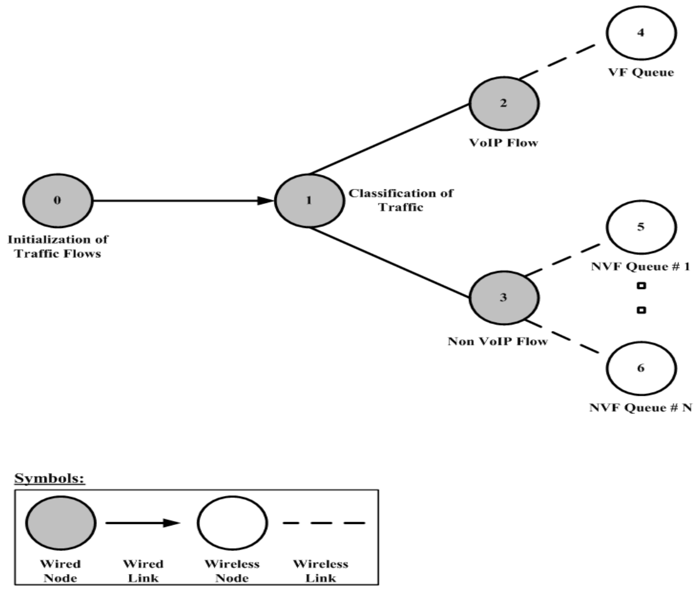

Figure 7 The VPQ topology in the NS-2

In the simulation configuration, the IEEE 802.11 based AP is Omni-directional where transmission ranges is based on data rate and distance. The AP is also implemented based on IEEE 802.11b mix-mode technology. Consequently, node-5 which is initially located at a distance of around 50 Meters, far from the AP, starts sending the VF flow to the AP at a data rate of 11Mbps. Then, node-5 gradually moves away from the AP with changing the data rate to a lower value; as the distance increases between node-5 and the AP, the data rate changes to 5.5 Mbps at a distance of 70 Meters, to 2 Mbps at a distance of 90 Meters and to 1 Mbps at distance of around 115 Meters. The VoIP connection is made between the wired node-2 and the wireless node-4. After that, the Wired-VF and the Wireless-VF start bidirectional communication with each other and packets are exchanged between node-2 and node-4 as they reach within the range mentioned above.

At the same time, node-5 provides a link between the NVF-Wired (NVF-W) and the Wireless-NVF (W-NVF). Node-5 and node-6 are numbered as NVF queue # 1 to NVF queue #, number of flows on the network. These nodes are also communicate with each other in a bidirectional way. Packets are exchanged among node-3 such as Wired-NVF and node-5 to node-N, number of nodes as they reach within the range of Access Point-B (AP-B).

IV. Result & Discussion

This section includes the achieved outcomes of the various topologies simulated in the NS-2. The VPQ has been compared with the Contention-Aware Temporally fair Scheduling (CATS) and the Temporally-Weighted Fair Queuing (T-WFQ) traffic schedulers. Results have shown that the VPQ scheduler has per-packet delay bounds that provide both bounded delay and fairness on IEEE 802.11 WLANs. The VPQ scheduler has the advantage of providing both less delay-guarantees and fairness, concurrently. It also 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. A brief discussion on the results and the performed comparison is given in the following paragraphs.

Figure 8 shows the total throughput (Mbps) of VPQ, CATS and T-WFQ algorithms in the simulation. The T-WFQ algorithm shows the lowest throughput as the T-WFQ throughput started from 2.3 Mbps and it reached 1.7 Mbps at 1200 (sec). The CATS is higher than the T-WFQ algorithm due to its better performance. The CATS throughput started from 2.7 Mbps and gradually decreased to 2.2 Mbps. Our proposed VPQ algorithm performs better than both of these algorithms. The VPQ has a higher throughput due to its classification of VF and NVF traffic, high data-rate flow and facilitating of more packets. The proposed VPQ throughput started from 4.6 Mbps and ended at 4.1 Mbps.

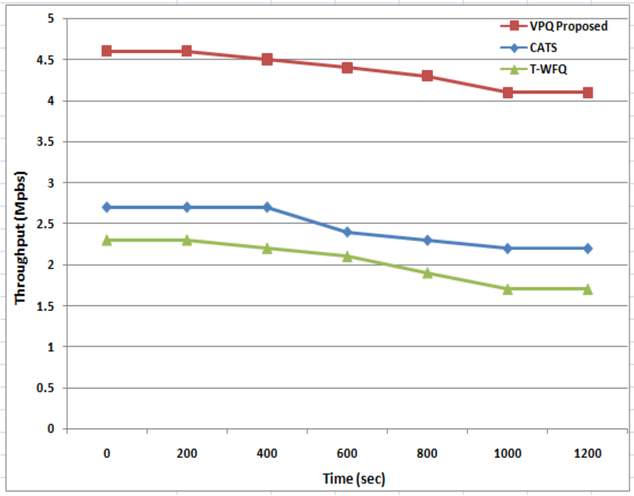

Figure 8 Total Throughput according to the mobility of mobile the station

Figure 9 shows the fairness index according to the mobility of the mobile station. The fairness measured from 0 to 1 where larger than 0, is the best.

The figure shows the comparison performed among the proposed VPQ, and the CATS and T-WFQ algorithm in terms of fairness. It has been found that the T-WFQ algorithm started its fairness from 0.88 and increased to 0.9 at 600 (sec) then it decreased gradually to reach 0.86 at 1200 (sec). 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.

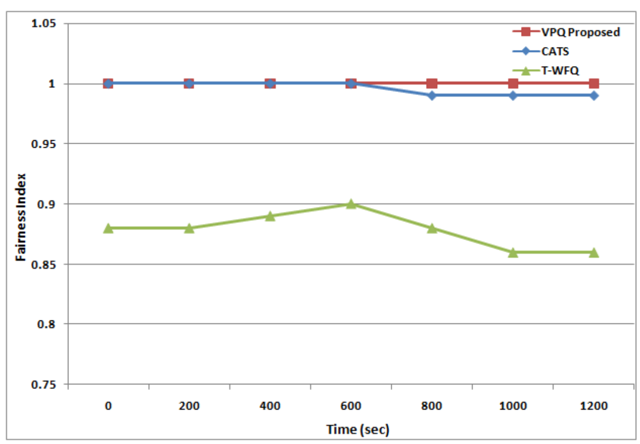

Figure 9 Fairness index according to the mobility of the mobile station

Figure 10 shows the throughput of all flows over IP-based networks based on packet size. This figure illustrates the total throughput in Mbps of the proposed VPQ as well as the CATS and T-WFQ algorithms on the WLAN. The T-WFQ algorithm shows the lowest throughput and it started from 2.6 Mbps and reached 0.7 Mbps at 1200 (sec). The CATS algorithm throughput was higher than the T-WFQ algorithm due to its better performance on the WLAN. The CATS throughput started from 2.3 Mbps and gradually decreased to reach 1.6 Mbps. However, our proposed VPQ algorithm showed better results than both of the other algorithms as the VPQ has a higher throughput due to the classification of the VF and NVF traffic, a high data-rate flow and the facilitating of more packets. The proposed VPQ started the throughput from 4 Mbps and reached 3.2 Mbps at 1200 (sec).

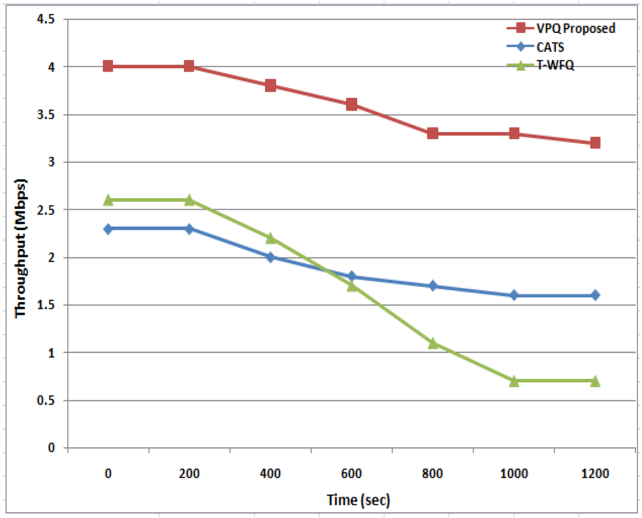

Figure 10 Total Throughput according to the packet size of flow

Figure 11 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. We also compare the proposed VPQ with the CATS and T-WFQ algorithms. We notice that the T-WFQ algorithm starts its fairness from 0.88 and reduces it to 0.3 at 1200 sec.

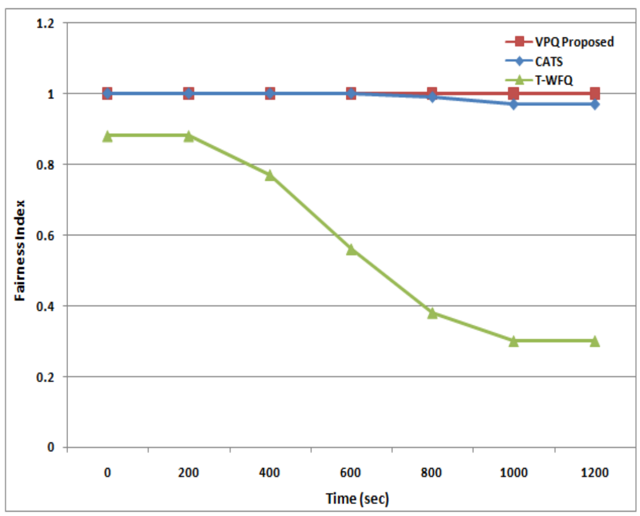

Figure 11. Fairness Index according to the Packet Size of Flow

Figure 11 clearly indicates that the performance of the VPQ is better than CATS in the simulation and both the algorithms outperform the T-WFQ algorithm.

V. Conclusion

This paper presented the a new scheduler system model named the Voice Priority Queue (VPQ) algorithm. This algorithm aims to address the scheduling issues of real-time applications on WLANs. The main function of the VQP is assuring that time is fairly allocated despite the variable data rate and packet size of the mobile station. The VPQ is an efficient scheduler as it performs better in term of fairness and throughput. In addition, it satisfies the unique requirement imposed by the VoIP. In comparison with related scheduling algorithms of similar efficiency, the VPQ has shown a higher throughput, as well as a higher fairness index. It is expected that the proposed scheduling algorithm will be able to offer fairness and high throughput in the future of the VoIP over WLANs. As a future work, the second and third stages of the VPQ will be further studied and designed.

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