# A NOVEL VOICE PRIORITY QUEUE (VPQ) SCHEDULER AND ALGORITHM FOR VOIP OVER WLAN NETWORK

**Kashif Nisar, Suhaidi Hassan, Mohammed M. Kadhum**

InterNetWorks Research Group
School of Computing, College of Arts and Sciences
Universiti Utara Malaysia
06010 UUM Sintok, MALAYSIA

Email: kashif@uum.edu.my

***Abstract***

*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 final stage of 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 final 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 over WLANs.*

***Keywords:*** *VoIP, WLANs, VPQ Scheduler*.

## I. INTRODUCTION

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 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].

### A. VoIP Network Systems

Figure 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.

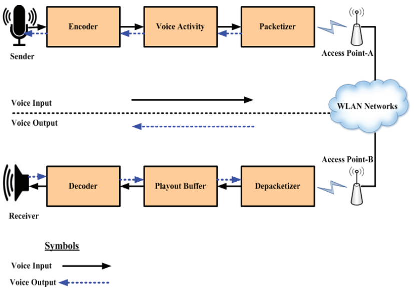


Figure 1. VoIP Network Systems

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].

Table 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 [13].

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 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 paper, we 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 [14].

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.

An AP can support (10) to (16) Mobile Nodes (MN) over 802.11b on the G.711 codec technique over an infrastructure architecture network. 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.

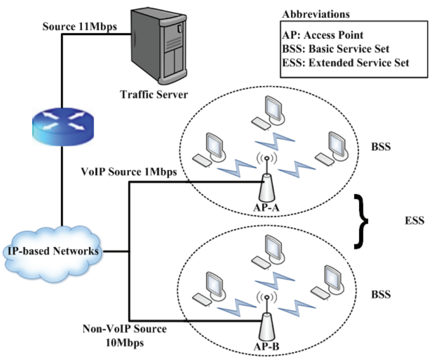


Figure 2. VoIP over a WLAN Network

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). 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 [15]. 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 2.

Table 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 |

## B. Problem Statement

Quality of Services (QoS) is considered as the main issue in VoIP systems. 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 3 shows a bottleneck topology of mixed-mode traffic over a WLAN.

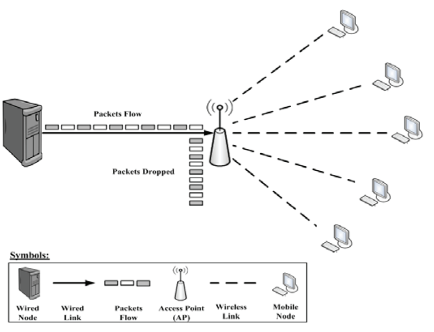


Figure 3. Bottleneck Topology of mixed-mode Traffic over WLANs

The above problems degrade the QoS of the VoIP over IP-based networks. Figure 3 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 wireless 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, our 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 real-time 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. 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.

### C. Research Aim and Objectives

We will study essential related work to examine the available scheduling algorithms outcome and drawbacks. We will propose new scheduling system model and algorithms to enhance the performance of the VoIP over WLANs using IEEE 802.11 standards. We 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 novel 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.

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 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.

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 (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 (D-CATS) [16].

### A. 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, we can classify scheduling system model and algorithms due to their nature of behavior over IP-based networks. We can classify schedulers as a packet-based scheduler, frame based-packet scheduler, bit-by-bit scheduler 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 [16].

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.

### B. Deficit Round Robin Scheduler Scheme

Round Robin (RR) has many variations and modifications of scheduling algorithms. Deficit Round Robin (DRR) was introduced by M. Shreedhar et al. [17] for active queue flows in the priority Round Robin (RR) formation [18], [19]. The DRR scheduler is classified as a frame-based packet scheduling algorithm for high-speed networks. DRR fulfils 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.

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.

### C. Deficit Transmission Time

R. Garroppo *et.al.* [20] 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. DTT supports Basic Service Set (BSS) infrastructure. Details are in Figure 4.

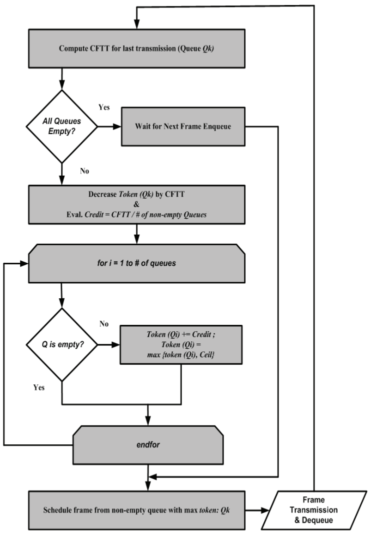


Figure 4 Flowchart Diagram of DTT Scheduler [20]

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. DTT is implemented on two-way calculations; one way with nodes and another way with AP.

The first way obtains the Signal-to-Noise Ratio (SNR) values from the Wireless Network Interface Cards (NIC's) to evaluate 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.

### D. Deficit Transmission Time

H. Wu *et.al*. [21] 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.

- 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). The history of traffic flow indicates the deadlock in real-time traffic flow. LLEPS queue process is demonstrated in Figure 5.

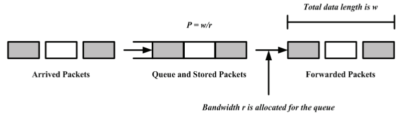


Figure 5 LLEPS queue process

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 as can be seen in the figure above. Let be the queue priority of task . Every time the value will be bigger than the other queues. LLEPS provides the value for each queue the same as for streaming transmissions on IP-based networks.

- 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 and occupy bandwidth for each task as described. LLEPS architecture and components are based on the description in Figure 6.

- Queues
- Min Heap
- System Timer
- Scheduler

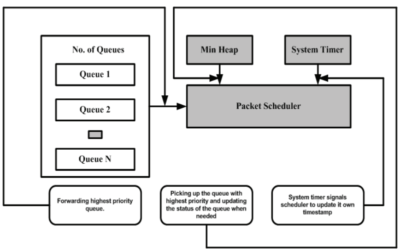


Figure 6. Architecture of LLEPS

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).

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.

## III. FIRST STAGE OF VPQ

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). A novel 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. 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.

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. 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.

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. 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 differentiates the VF packets based on packet size.

In Figure 7, 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.

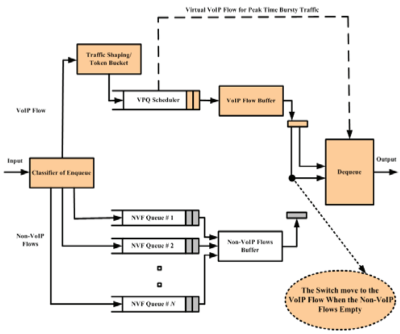


Figure 7 The Final Stage of Voice Priority Queue System Model

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 shown in Figure 7. 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 8.

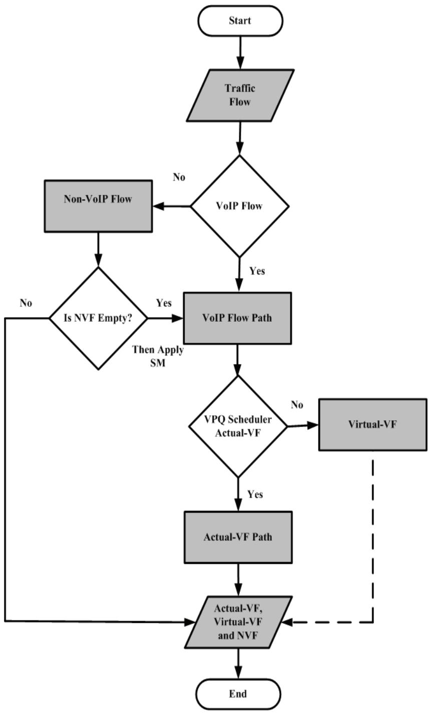


Figure 8 Final Stage of the VPQ Flow Chart

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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;
```

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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 9 Final Stage of the VPQ Scheduling Algorithm

## IV. SIMULATION SETUP

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.
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 3.

Table 3: Comparisons Of Techniques

Symbols:
License = L
License Free = LF
Support = √
Not Support = X
Partial Support = P √

| 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 |

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.

### A. 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.
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). 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.

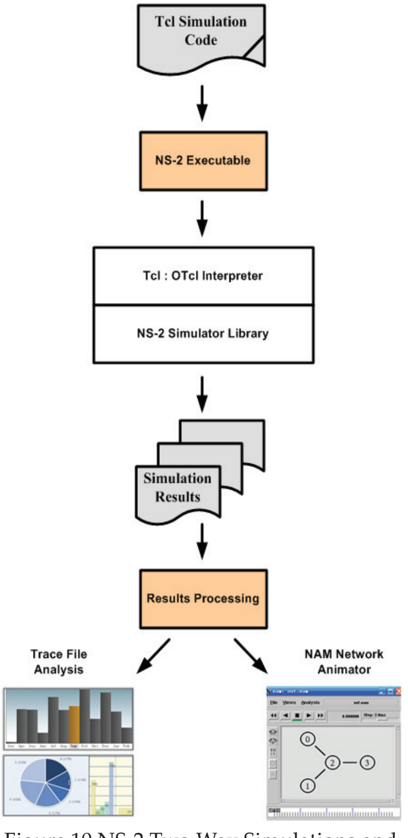



Figure 10 NS-2 Two-Way Simulations and Results Analysis Process

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. The limitations of a measurement test-bed can be difficult to configure and reconfigure. Test-beds can be very expensive.

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 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.

Furthermore, the simulation topology is shown in Figure 11 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).

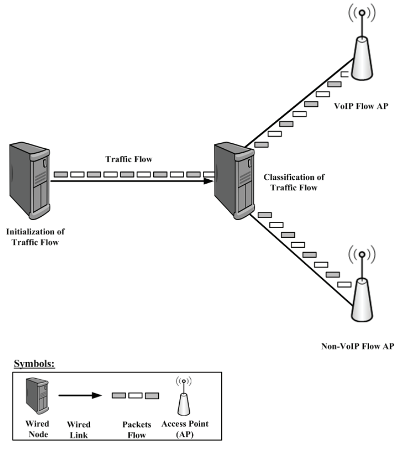



Figure 11 Simulation topology of the NS-2

In Figure 12, 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).

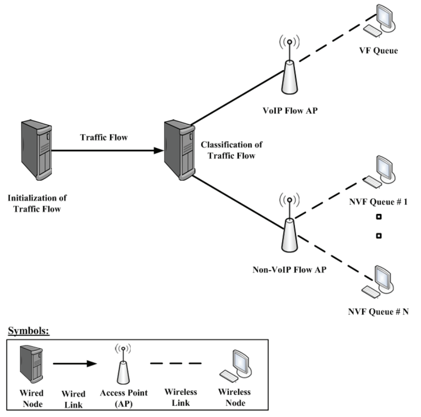


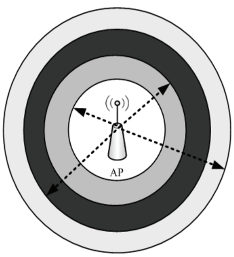

Figure 12 The VPQ topology in the NS-2

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).

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 13.




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 13 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.

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.

## V. RESULTS & DISCUSSIONS

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 14 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.

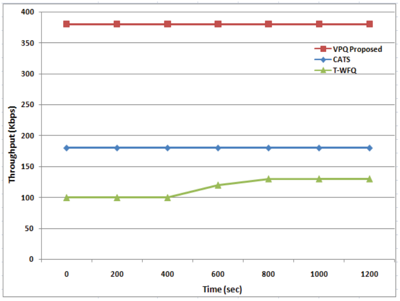



Figure 14 Throughput of Flow when using the VPQ in Access Point

As shown in Figure 15, 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 15 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.

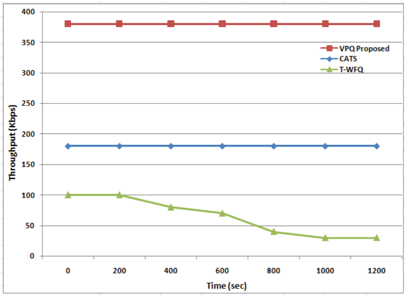



Figure 15 Throughput of when using an Access Point

In Figure 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.

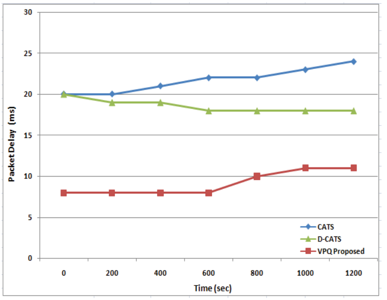



Figure 16 Packet delay

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

## VI. CONCLUSION

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 paper proposes an efficient scheduler and algorithms for a VoIP application over WLANs. This paper 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.

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