Improving IEEE 802.11 to Support Quality of Service in Wireless Networks

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ABSTRACT

This paper investigates and describes the Quality of Services (QoS) provisioning techniques for IEEE 802.11 based networks, focusing on the Distributed Coordination Function (DCF). This paper propose better techniques to provide QoS by assigning new parameters to the DCF access method, involving the DCF Interframe Space (DIFS), backoff time and the maximum data packet size to high priority nodes, which will distinguish the high priority traffic from the low priority traffic to support QoS. A simulation is done using Network Simulator 2 (NS-2) and the expected output is then presented.

Keywords

Wireless LAN, 802.11, Quality of Service (QoS)

1. INTRODUCTION

Wireless LAN (WLAN) is a LAN to which mobile users (clients) can connect and communicate by means of high-frequency radio waves rather than wires. The WLAN consists of three basic elements [8]:

1. The physical medium used to carry WLAN signals between stations
2. A set of medium access control rules embedded in each WLAN interface that allow multiple computers to fairly arbitrate access to the shared wireless channel.
3. A packet (MAC service data unit, MSDU) that consists of a standardized set of bits to carry data over the system.

Technically, WLAN standard is described by IEEE 802.11.

As the network world becomes more popular, the network load has become a critical issue. The LAN, which was originally designed to carry data traffic (such as file transfer, e-mail and Internet browsing) is now being used to carry real-time traffic. Highly congested network are demanding for better enhancement to support Quality of Service (QoS) that requires fast yet reliable transmission. This includes applications such as internet banking, and multimedia across networks which require real-time traffic such as video streaming and voice over internet protocol (VoIP).

Since the emergence of LAN and multimedia technology across networks, network usage had increased exponentially and thus congestions occurs which leads to the need on providing QoS in the network itself. Over the past few years, researchers had come with various solutions to provide QoS. These include QoS provisioning on layer two such as packet based flow and the upper layer such as queuing algorithms and traffic shaping. However, most of the algorithms proposed are designed specifically for wired networks. Since the method on medium accessing for wired and wireless network are completely different, the proposed algorithm or technique may not be suitable to be implemented directly on the wireless medium.

The remainder of this paper is organized as follows. Firstly, this paper will discuss on the IEEE 802.11 channel coordination function before focusing on the Distributed Coordination Function (DCF) channel access method. Then, other proposed techniques from previous research are presented before outlining the author's proposed techniques. Finally, a brief description of simulation scenarios and expected results is given.

2. IEEE 802.11 Channel Coordination Function

WLAN devices can only hear one frequency at a time to communicate with each other, so there must be turns for them to use the channel to avoid collisions. The process for the workstations to take turns using the medium is called the coordination function.

The IEEE 802.11 standard defines two types of coordination function that are the Distributed Coordination Function (DCF) and the Point Coordination Function (PCF). DCF is used for asynchronous contention based distributed accesses to the channel while the latter is used in the centralized, contention-free accesses. Since this paper focuses on DCF, the following subsection will discuss more on DCF.

2.1 IEEE 802.11 Distributed Coordination Function (DCF)

DCF is used specifically for the contention-based channel access method. The definition of contention is that, the client nodes contend or compete with each other to use the network channel. In the contention basis, any client nodes can attempt to transmit data at any time it wanted to. However, the problem occurs when two computers start to transmit data at the same time, where a collision will definitely happen. DCF adopts the Ethernet, IEEE 802.3 Carrier Sense Multiple Access with Collision Detection (CSMA/CD) mechanism with several modifications, which is known as Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) mechanism. Whereas CSMA/CD is used to handle collisions after it occurs (by retransmitting the damaged packet), CSMA/CA avoids the collisions altogether.

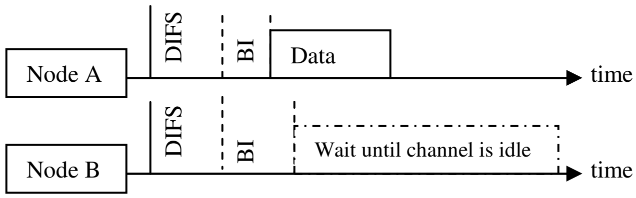


Figure 1. The operation of DCF mechanism

CSMA/CA does not wait for collisions to occur to handle collision avoidance. Figure 1 shows how the DCF mechanism operates to avoid collision before it actually occurs. Instead of having the two clients, Node A and Node B responsible for the collision to wait a random amount of time (as in CSMA/CD), CSMA/CA has all the clients to wait for a random amount of time, T_{wait}, which consists of DCF Interframe Space (DIFS) and backoff interval (BI) before attempting to do transmission, as shown in **(1)**. BI is a uniform random value, sampled exponentially from [0, CW].

$$T_{wait} = DIFS + BI \qquad \textbf{(1)}$$

Although the value of DIFS is the same for each station, the BI value is taken randomly to avoid collision. On the other hand, DIFS is derived from an equation as in **(2)** below:

$$DIFS = 2\ (Slot\ time) + SIFS \qquad \textbf{(2)}$$

It is essential to know where the DIFS is derived from, as this involves on providing QoS which will be discussed later in this paper.

3. RELATED WORKS IN WLAN QoS

WLAN had been a critical issue in the fast paced networking world. This is reflected by the number of research done. In providing service differentiation, the network traffic is divided into two categories, which are the low priority and the high priority traffic. Service differentiation is then made based on the two priority categories. Focusing on DCF, several approaches had been made by past researchers to support QoS. In this section, several ideas to provide QoS in IEEE 802.11 are described, which involves Interframe Space based, Contention Window (CW) separation based, and persistence factor based, discussed in the subsection below.

3.1 Interframe Space (IFS) Based

Realizing the weakness of bandwidth reservation to provide QoS, Deng [6] rejects reservation schemes as it leads to a major drawback, which is when the source is reserved but unused, it is simply wasted. He proposed a method to support two priorities, high priority and low priority stations. Higher priority stations will wait for a duration of PCF Interframe Space (PIFS), while lower priority stations will wait for a duration of DIFS before attempting data transmission. This is because PIFS has a shorter waiting time compared to DIFS. Several assumptions are made where there is no hidden node, no stations operates on power-saving mode and no interference from nearby Basic Service Sets (BSS). Simscript simulation of video, voice and data traffic with priorities of 3,2 and 0 with the ratio of 1:1:2 is performed. Results (IFS based, combined with CW separation) showed that there are performance improvements for high priority traffic in heavy load conditions where video traffic uses most of the bandwidth (55%) and lower priorities use the remaining bandwidth. In low load condition, lower priority traffic has the required bandwidth. Although it is illustrated that video and voice traffic has lower access delay and lower packet loss probability than in DCF, data traffic suffers access delay and higher packet loss than in DCF.

Another IFS-based research, done by Aad [2] uses almost the same scheme as Deng [6]. Higher priority stations, labeled as $j+1$ and low priority stations, j have different Interframe space (IFS) values, denoted as $DIFS_{j+1}$ and $DIFS_j$, where the value of $DIFS_{j+1}$ is lower than $DIFS_j$. The maximum random range of priority j+1, (RR_{j+1}) is defined as the maximum Backoff Interval (BI) of that priority. If the strict condition $RR_{j+1} < DIFS_j - DIFS_{j+1}$ is satisfied, then all packets of priority j+1 have been transmitted before any packet of priority j is transmitted. In lower load condition, $RR_{j+1} > DIFS_j - DIFS_{j+1}$, a packet which could not access the medium the first time may have its priority decreased in the subsequent attempts. Simulations were carried out and the results show that the method does not change the system efficiency, with data sums remains the same [10]. The method works well for both Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) flows with more significant effect on UDP flows compared to TCP flows. It also works in noisy environment and keeps the same stability of the system.

Meanwhile, the use of Urgency Arbitration Time (UAT) to differentiate service by Benveniste [5] gives another perspective on providing QoS. UAT is the time a station has to wait before a transmission attempt following a period when the medium is busy. He also introduces Arbitration Interframe Space (AIFS) and Backoff Counter Update Time (BCUT) but both are actually DIFS and SlotTime respectively. Higher priority traffic is assigned shorter AIFS and BCUT values compared to the low priorities. The AIFS value for high priority is the same as PCF Interframe Space (PIFS) and a minimum backoff time of 1 in order to prevent conflict with medium access by centralized protocol PCF. A simulation was carried out where AIFS (high_prio) = PIFS, AIFS (low_prio) = DIFS, CW (high_prio) = [1, 32] and CW (low_prio) = [0, 31]. Results showed that the delay and jitter of high-priority traffic are decreased and under moderate load condition, the performance of low priority traffic is also improved compared to DCF.

3.2 Contention Window (CW) Separation Based Approach

In the same research on IFS based differentiation, Deng [6] also proposed a scheme based on separation of CW. Originally, the random Backoff Interval (BI) is uniformly distributed between $[0, 2^{2+i} - 1]$, in which i is the number of times the station attempted transmission of the same packet. In his scheme, the high and low priorities have random BI values uniformly distributed in intervals $[0, 2^{2+i}/2 - 1]$ and $[2^{2+i}/2, 2^{2+i} - 1]$. This approach is then combined with the IFS approach, discussed earlier. Simulation results reveal some improvement only in delay and jitter for high priority traffic (voice and video).

On the other hand, Xiaohui [12] suggests the Modified DCF (M-DCF) scheme, which uses different values of CW_{min} and CW_{max} for service differentiation. Simulations of ad-hoc wireless LAN with 10 data stations and between 10 and 35 voice stations were performed. Voice service had CW_{min} of 7 and CW_{max} of 127 while data service had CW_{min} of 15 and CW_{max} of 255. The outcome illustrates that M-DCF decreases the total packet dropping probability and the dropping probability of voice packets as well as reduces the contention delay of both voice and data packets compared with DCF.

Another work done by Barry [4] and Veres [10] recommend using different values of CW_{min} and CW_{max} for different priorities, in which higher priority has lower CW_{min} and CW_{max} values than those of lower priority. Simulations of high priority traffic with CW_{min} between [8, 32] and CW_{max} = 64, and low priority traffic with CW_{min} between [32, 128 and CW_{max} = 1024] were performed. The outcomes show that the high priority and low priority traffic undergo different delay.

Meanwhile, Aad [3] introduces a differentiation mechanism based on CW_{min} separation, in which higher priority traffic has lower CW_{min} value. Simulations of a wireless LAN consisting of an access point (AP) and three stations with CW_{min} values of 31, 35, 50 and 65 were conducted with both TCP and UDP flows. The results reveal that for the same set of CW_{min} values, the differentiation effect is more significant on UDP flows than on TCP flows. The per-flow differentiation is introduced, in which the AP sends back Acknowledge (ACK) packets with priorities proportional to priorities of the destinations. In other words, the AP waits for a period of time which is proportional to delay from a destination before transmitting an ACK packet to the destination.

3.3 Priority/ Persistence Factor (PF) Based Approach

Priority or Persistence Factor (PF) is only applicable after the intended receiver did not receive the data, due to collision or data loss, where the sender did not receive any ACK packet from the intended receiver for a duration of SIFS. The sender will then attempt to transmit data and increase the CW value using an equation shown in **(3)**.

$$CW=2(CW+1)-1 \quad \textbf{(3)}$$

On the above equation, the number *2* is the PF.

The research done by Aad [1] [2] proposed a method based on backoff increase function. In the original DCF, the CW is multiplied by a Priority Factor (PF) of 2 after each collision. In Aad scheme, higher priority traffic has a lower PF, denoted as P_j. Simulations of three priorities with PF values of 2, 6 and 8 were conducted. The results demonstrate that this method works well with UDP flow, but not with TCP or in a noisy environment. The efficiency is not lost but the stability of the system is decreased.

Meanwhile, in the same research on the IFS-based approach, Benveniste [5] also recommends a technique based on Persistent Factor (PF). After each collision, the CW is multiplied by a PF. Higher priority traffic has lower value of PF. For delay sensitive applications and with capability congestion estimation, PF value should be less than 1. For delay insensitive traffic, value between 1 and 2 can be used. Simulation of traffic with two priorities and AIFS (high_prio) = PIFS, AIFS (high_prio) = DIFS, PF (high_prio) = 0.5 and PF (low_prio) = 2 was carried out. The outcomes showed that high priority traffic performance is improved without any significant effect on low priority traffic. Furthermore, the delay and jitter is less than 10ms, which could not be obtained with IFS based differentiation alone.

3.4 Discussion

Regarding the IFS-based approach, Deng's [6] idea uses only two IFS values, which are PIFS and DIFS, meaning that it can only support differentiation for only two types of priorities. Moreover, it is known that, for a station to have the privilege to transmit data, it has to wait for a total time of the sum of IFS and random backoff time. Therefore, if this technique is used for service differentiation, it is possible that even if high priority traffic has a low IFS value, it still have the probability to have a higher total waiting time compared to lower priority traffic, if the value of the random backoff time of the high priority traffic is higher than the lower priority traffic's random backoff time. In Aad's technique [2], it is possible that the idea used can support multiple priorities, provided that the value of $DIFS_j$ is properly selected. Benveniste's [5] method on using AIFS is similar to Aad [1] [2] and Deng [6], as AIFS is actually the generalization of shortened DIFS. However, the idea on BCUT for differentiation is not possible as the SlotTime used in the original DCF is the minimum possible [10].

On the CW separation wise, although Deng [6], Xiaohui [12] and Barry [4] only provide service differentiation for two priorities, it is possible to provide more priorities, by separating the CW range into more than two ranges and assign them according to the priorities.

Based on the past researches and discussion done, this paper will propose a new scheme, which is discussed in the following section.

4. PROPOSED SCHEME

Service differentiation between traffic classes (priorities) is based on differentiation of the time the traffic has to wait before transmission. Two main parameters that decide the waiting time of traffic are Interframe Space (IFS) and Contention Window (CW) [10]. The proposed scheme adopts contention parameters, which are the DIFS period and the back off interval period to bias performance in favor of high priority traffic. Besides that, another parameter being tuned is the maximum packet size.

4.1 Shorter DIFS Period

DIFS is the duration for a mobile node that wants to transmit data has to wait after sensing the channel is idle. The technique proposed in this experiment is that the high priority nodes are assigned shorter DIFS. This means high priority nodes have a shorter waiting time, which allows the higher priority node to transmit ahead of the lower priority nodes [9]. The analogy is that in a hospital, a severely injured patient is more likely to have a short waiting time before being treated by the doctor compared to the other patients. While high priority nodes will always have a shorter waiting time, it means high priority nodes are most likely to have the opportunity to always being first to transmit data after the channel is sensed idle compared to the low priority nodes.

4.2 Dynamic Contention Window

As discussed before, CW is a backoff mechanism for a mobilenode to avoid data collision, even after sensing the channel is in the idle state after the DIFS period. CW generates a random number within the range of CW_{min} and $CW_{max.}$ In the original

IEEE 802.11, there is no differentiation of CWmin and CWmax between a high priority and low priority traffic. In this experiment, the value of CWmin and CW_{max} for both of the traffic is changed to support differentiation. The CW range is divided into two parts. The first part ranges from CW_{min} to $CW_{max}/2$ while the second part ranges from $CW_{max}/2$ to CW_{max}. The idea is simplified as shown in Figure 2. The $CW_{max}/2$ is symbolized as **α**.

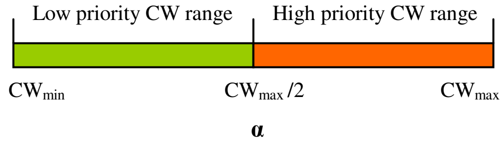


Figure 2. Contention Window Separation between high priority and low priority traffic

The CW can be configured under the MAC layer in NS-2 [7]. The dynamic feature allows the CW to be set according to the network load. If there is no high priority flow, the low priority flow can encroach into the high priority CW range to a certain extent. It goes the same way when there are lots of high priority flows in the network. The high priority flow is capable to encroach into the low priority CW range to a certain extent to support the high demand of high priority flow. Although considered as a high priority flow, it is not wise to permit high priority flows to encroach fully into the low priority CW range. This is important to protect the low priority flows in order to avoid low priority data flow starvation.

The dynamic feature checks the packet loss rate to determine the network load. If the loss is high, **α** will slide to the right, allocating more CW range for high priority flows. The higher the loss, the more **α** will slide to the right, but until it reaches to a certain extent to protect the low priority flows.

With the separation of CW range between high priority and low priority flows, logically the average delay of high priority traffic should be much lower than low priority traffic and the average throughput of high priority traffic should be much higher than low priority traffic.

4.3 Maximum Packet Size

In the original IEEE 802.11, there are no differences of packet size between a high priority and low priority traffic. No differentiation means the packets for the different priorities are treated the same. In this experiment, a high priority flow is assigned larger frame size compared to low priority traffic.

With larger frame size, a high priority node will be able to transmit more information per medium access once it has the opportunity to transmit. From the figure above, the header and preambles is the shadowed in the data box. Logically, the data transmission for a high priority node will be much faster in terms of less packet header and preambles to process.

5. SIMULATION SCENARIO

All simulation setup are configured using the TCL language in the TCL script of NS-2 [7]. In the simulation setup, the environment is set to radio links where channel type is configured as wireless channel.

Radio propagation models are used to predict the received signal power of each packet. Since IEEE 802.11 considers both the direct path and a ground reflection, the propagation model used in this simulation is the Two-Ray Ground Reflection Model.

This experiment is done as a per-based mobile communication. This means that each node only transmit one type of data, that is whether a high priority data, or a low priority data. Eight nodes are used where four nodes acts as the data source and four of the nodes as the destination. As a result, there are four pairs of nodes, where two pairs will simulate the high priority data flow and the other two pair will simulate the low priority data flow. All of the QoS parameter readings are taken at the destination nodes.

In order to simulate the real wireless network, the traffic involved includes Constant Bit Rate (CBR), Hypertext Transfer Protocol (HTTP) and File Transfer Protocol (FTP).

As discussed before, the network simulation topology can be shown as in Figure 3. This means every node is in every workstations range, where no hidden node exists.

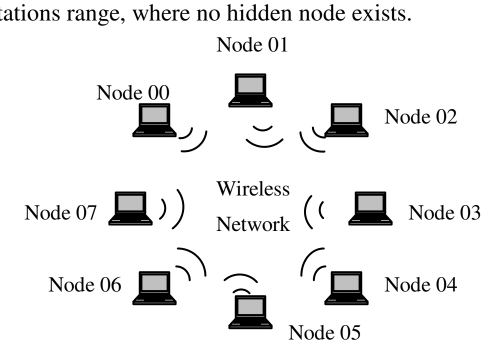


Figure 3. The topology of the network simulation

The bandwidth of the wireless channel is set to 54 Mbps, which represents the capacity of the 802.11g link.

There will be two types of network traffic that will be generated in the wireless channel, the low priority traffic and the high priority traffic. The high priority traffic includes both voice and video traffic while the low priority traffic represents the usual data traffic. Both types of traffic will be generated by the nodes, where 4 nodes will be the traffic generator while the other four nodes will be the traffic receiver. Two nodes of the traffic generator will simulate high priority traffic, while the other two nodes generate low priority traffic. The topography is set as flat, which means only the X and the Y axis are involved. The Z axis is always set as zero.

To simulate the voice and video traffic, the node will generate a CBR data flow. The only difference between voice and video traffic is the throughput, where video is set to deliver 64kbps while voice is set to 32kbps. On the other hand, data traffic is simulated through two traffic patterns, using FTP and HTTP. From Figure 3, N00 and N02 will generate a high priority flow while N04 and N06 will be the high priority packet receivers respectively. N01 and N03 will generate FTP and HTTP traffic respectively, with N05 and N07 will be their packet receivers. Otherwise, the summary of the nodes is shown as Table 1 below:

Table 1. Summary of nodes and traffic type in simulation

Nodes	Status	Traffic Type	Data Rate
N00	HP Sender	Voice	32 kbps
N01	LP Sender	FTP	default
N02	HP Sender	Video	64 kbps
N03	LP Sender	HTTP	default
N04	HP Receiver	Voice	-
N05	LP Receiver	FTP	-
N06	HP Receiver	Video	-
N07	LP Receiver	HTTP	-

Legend:

HP = High Priority

LP = Low Priority

Parameters used for the simulation is shown as below, in Table 2.

Table 2. Other parameters in simulation configuration

Station	SIFS	Slot time	DIFS	Pkt size	CW min	CW max
High Priority	5µs	9µs	23µs	1024	8	32
Low Priority	10µs	9µs	28µs	512	32	1024

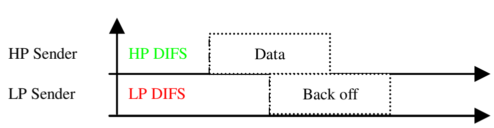
As discussed before, DIFS is derived from an equation of (2(Slot time) + SIFS), as shown in **(2)**. Thus, in order to change the DIFS period, changes can be made in two different parts, the Slot time and the SIFS. However, in this simulation, the Slot time remained as it is, while the SIFS changed, particularly for high priority nodes. No changes of parameter had been made for low priority nodes (low priority nodes will use the default settings).

In a predetermined duration, the simulation time used is 90 seconds. At time, t = 10s, the low priority traffic, FTP and HTTP will be generated into the network from the respected nodes. Then, at t=15s, voice and video traffic will be started to generate packets into the network. At t=89s, all the traffic will be stopped before the simulation ended at t=90s.

6. EXPECTED OUTPUT

The simulation results expected are for the ad-hoc network, where supposedly, high priority and low priority traffic have a significant difference in terms of throughput, delay and bit error rate.

On tuning a shorter period of DIFS on high priority nodes, compared to low priority nodes, means that high priority nodes have a shorter waiting time before attempting to transmit data. Shorter DIFS means that high priority traffic will always be transmitted first, before the low priority data as shown in Figure 4.



Legend:

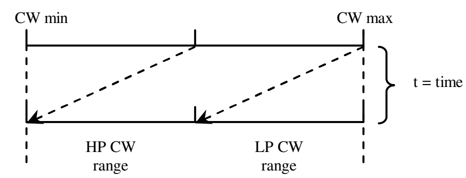
HP = High Priority

LP = Low Priority

Figure 4. High priority DIFS and low priority DIFS differences

Thus, from the results of the simulation, it is expected that high priority traffic will have a higher throughput compared to low priority traffic.

CW tuning and DIFS tuning must co-exist. This is because if the high priority node is only given the high priority DIFS without a specific CW value, it is possible that the CW for high priority nodes that will be generated randomly will have a higher value than the CW of the low priority nodes. This will result to a normalized network environment without having service differentiation. From this simulation on tuning the CW, it is expected that although high priority traffic will have a higher throughput and lower delay, the low priority traffic will not be starved. This is because, even though low priority traffic is given the second part of the CW division, eventually it will be given the opportunity to transmit data as the CW countdown enters the high priority CW range, as shown in **Figure 5**.



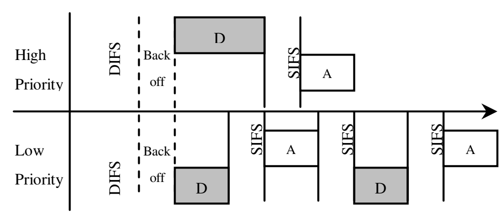
Legend:

HP = High Priority

LP = Low Priority

Figure 5. Low priority CW value will eventually be a high priority CW over time

Larger frame size gives the opportunity for the transmitting node to transfer more data when it is given the permission to use the medium. This can be illustrated in **Figure 6**.



Legend:

HP = High Priority

LP = Low Priority

D: Data

A: ACK

Figure 6. High priority and low priority frame size comparison

The figure suggests that for the same amount of data, high priority nodes sends it only once, while low priority nodes sends it twice. On sending it twice, it has taken more time for the receiver to process the data overhead, and time is also wasted to wait for the SIFS period and the ACK from the receiver.

7. CONCLUSION

The primary contribution of this paper focuses on detailed investigation on many of the DCF based access method of the wireless LAN. Of all the methods used in past researches, most of them only consider throughput guarantee but not delay and jitter requirements. These aspects of QoS are very important for video streaming and interactive video applications.

The simulation model proposed in this paper involves ideas, derived from the literature which includes tuning three parameters on the DCF access method to differentiate services between high priority and low priority traffic. This includes assigning shorter DIFS period for high priority nodes to decrease the data transmit waiting time. Besides that, dynamic differentiation of the CW ensures that, although high priority traffic is given special treatment, low priority traffic is not at all forgotten which may lead to starvation. Finally, the last technique is to use larger frame size for high priority nodes which is capable to carry more data compared to smaller frames at one time. Since the parameters had been configured to bias towards the high priority traffic, it is expected that the author's approach to provide QoS in wireless LAN is valid and applicable, thus improving the IEEE 802.11 to support Quality of Service.

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