Adaptive DCF of MAC for VoIP services using IEEE 802.11 networks

[1] Mr. Praveen S Patil , [2]Mr. Rabinarayan Panda ,[3] Mr. Sunil Kumar R D
[1,2,3] Asst. Professor, Department of MCA , The Oxford College of Engineering,
Bommanhalli , Hosur Road, Bangalore ,Karnataka, India-560100
Email: [1] sit.praveen@gmail.com, [2] rabi.mtech2011@gmail.com
[3] sonyvhm@gmail.com

Abstract:-Voice over Internet Protocol is an important service with strict quality of requirements in Wireless Local Area Network (WLANs). The IEEE 802.11e Standard has been introduced recently for providing Quality of Service (QoS) capabilities in the emerging wireless local area networks. This 802.11e introduces a contention window based that is Enhanced Distribution Channel Access (EDCA) technique that provides a prioritized traffic to guarantee minimum bandwidth needed for time critical applications. However this EDCA technique resets statistically the contention window of the mobile station after each successful transmission. This static behavior does not adapt to the network state hence reduces the network usage and results in bad performance and poor link utilization whenever the demand for link utilization increases. For that purpose a new adaptive differentiation technique has been proposed for IEEE 802.11e wireless local area networks that take into account the network state before resetting the contention window.

To improve the QoS of Voice over Internet Protocol services we proposed a new traffic for VoIP. The performance of the proposed technique and proposed traffic is evaluated and compared with the original IEEE802.11a technique. Preliminary results show that the proposed adaptive technique enhances the channel utilization and increases throughput

Keywords: Voice over Internet Protocol (VoIP), QoS, WLANs, DCF, enhanced DCF.

1. Introduction

The rapid increase in the number of Personal Digital Assistants PDA devices, palmtops and compact laptops has made wireless networks popular in practice. Such networks provide a flexible data communication system that can either replace or extend a wired LAN to provide location independent network access between computation and communication devices using waves rather than a cable infrastructure. Wireless networks are becoming more widely recognized as a general-purpose connectivity alternative for a wide range of business organizations owing to its simplicity, scalability, relative ease of integrating wireless access and ability for wireless stations to roam throughout the business organizations with the remaining connected to other existing network resources such as servers, printers, and Internet connections. One of the major types of wireless networks is the infrastructure Wireless Local Area Networks (WLANS) which are distinguished by the use of an access point which are distinguished by the use of an access point (AP). All communication in the infrastructure WLANs come through the AP.

In order to allow the stations to share the wireless medium efficiently, many practical WLANs have widely been deployed following the Institute of Electrical and Electronics Engineers (IEEE) 802.11 standards ratified in 1997 that operate at data rates up to 2 Mbps in the 2.4-GHz Industrial, Scientific and Medical (ISM) band. But the most general business requirements cannot be well supported by the slow data rate of the original IEEE 802.11 standard. Recognizing the critical need to support higher data-transmission rates, the IEEE ratified both 802.11a and 802.11b standards with the rates up to 54 and 11

Mbps in the 5 and 2.4-GHz ISM band, respectively. Moreover, both standards specify the operation of the Medium Access Control (MAC) protocol, which is responsible for controlled access to the transmission medium. The most important purpose of this protocol is to enable the capacity of transmission media to be utilized in an efficient manner by wireless network devices.

2. Legacy 802.11

In this section the legacy 802.11 MAC protocol defined. Its limitations to support QoS are highlighted. We consider an infrastructure basic service set (BSS), which is composed of an access point (AP) and number of stations associated with the AP. The AP connects its stations the infrastructure such as the Internet, and each associated station communicates only via the AP in legacy 802.11

2.1 Carrier Sense Multiple Access

The listen-before talk scheme of the DCF is based on carrier sense multiple access (CSMA). Applying the DCF, a station determines individually when to access the medium. Hence, the decision making process about medium access is distributed among all stations. The station service responsible for information exchange is referred to as MAC service data unit delivery (MSDU delivery). MSDUs are transmitted with the help of a MAC protocol data unit (MPDU) or, if a station decides to fragment a long MSDU into a number of MPDUs, with the help of several MPDUs. Stations deliver MSDUs1 of arbitrary lengths (up to 2304 bytes) ("talk"), after detecting at there are no other transmissions in progress on the wireless medium ("listen"). If two or more stations detect the medium as being idle at the same time, they may initiate their transmissions at the same time, and inevitably a collision occurs.

2.2 Collision Avoidance

To reduce the probability of collisions, the DCF applies a collision avoidance (CA) mechanism, where stations perform a so called back off procedure before initiating a transmission. After detecting the medium as idle for a minimum duration called DCF interface space (DIFS). Stations keep sensing the medium (listening) for an additional random time called back off time. A station initiates its transmission only if the medium remains idle for this additional random time. Each station determines the duration of this random time individually, as a multiple of a slot time. A new independent random value is selected for each new transmission attempt. See Fig.1 for an illustration of the back off procedure. Since stations select the number of slots at random out of an interval between (, CW), which is initially set to the
minimum value CWmin (15 in 802.11a), it is less likely that collisions occur. All stations use the same value for CWmin, but select their random backoff time individually. Since all stations operate with the same CWmin, all stations have the same medium access priority in the DCF.This results in no mechanism to differentiate between stations and their traffic, and therefore no QoS support in the DCF.

At the very first transmission attempt, CW value is equal to the initial back off window size CWmin. For every unsuccessful transmission the value of CW is doubled until CWmax is reached. After transmitting a frame, the station expects to receive an acknowledgment (ACK) from the destination station following a Short Inter-Frame Space (SIFS) time. If the acknowledgment is not received, the sender assumes that the transmitted frame was subject to a collision, so it schedules a retransmission and enters the back off process again. After every successful transmission the
CW is reset to CWmin.

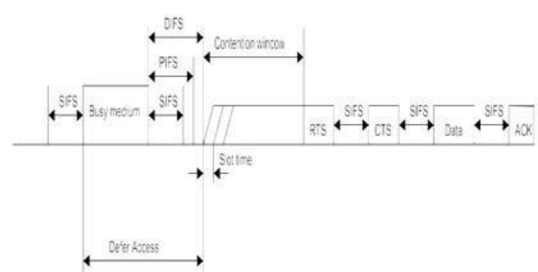

FIG 2.1 Mac Layer

If the medium gets busy due to interference or other transmissions while a station is down counting its back off counter (i.e., waiting until the random time has elapsed), the station stops down counting and defers from medium access until the medium becomes idle for a DIFS again. This occurs, for example, when the random back off time of a station is longer than the random back off time of at least one other station. Stations that deferred medium access because of detecting the medium as busy do not select a new random back off time, but continue down counting the time of the deferred back off after sensing the medium as idle again.

Since collisions may occur due to the nature of the CSMA/CA protocol, a station that has transmitted an MPDU needs to be informed about the success of its transmission. Therefore, in legacy 802.11 using the DCF, each transmitted MPDU requires an acknowledgment (ACK). For each successful reception of an MPDU, a receiving station immediately acknowledges the frame reception by transmitting an ACK frame back to the transmitting station. If the transmitting station does not receive this ACK frame right after the MPDU transmission, this transmitting station concludes that the MPDU was not delivered successfully and may repeat the transmission.

The CW of a transmitting station increases when a transmission fails (i.e., the transmitted data frame is not acknowledged). After any unsuccessful transmission attempt, a new backoff procedure is performed with double-sized CW, up to a maximum value defined by CW_{max} (equals 1023 in 802.11a). The CW is now larger than before to reduce the probability of repeated collisions if there are multiple stations attempting to access the medium. The larger the Cw, the collision probability is low.

2.4 Optional Fragmentation and RTS/CTS

Additionally, to alleviate the hidden station problem, DCF uses optional Request-to-send /Clear-to-Send (RTS/CTS) frames before packet transmission. This process is illustrated in Figure1

3. Enhanced DCF

EDCA offers differentiation depending on the traffic access category. Each category is given a priority depending on resource requirements of each application. This differentiation technique offers adaptation depending on traffic category and does not take into account network state or network contention level since EDCA technique resets the CW of each station after each successful transmission to CW_{min}

In case of highly congested channels there is a high probability of collisions. These collisions entail cw into acquiring higher values. Consequently it is more favorable that cw values are accustomed to the channel state therefore getting values distant from CW_{min} whenever the channel is congested and values close to CW_{min} whenever the channel is free. Besides, this adaptation scheme should also be relative to cwmax therefore cw should acquire values close to CW_{max} whenever the channel is congested and values distant from CW_{max} whenever the channel is free Enhancing the EDCA technique to adapt CW values to the channel state has many advantages over the original EDCA technique of IEEE 802.11e. Such enhancement will avoid the waste of time (backoff time) since the original EDCA keeps increasing CW to a value that would allow transmission. Instead the proposed adaptive approach would set CW directly to a value close to the required one for transmission therefore eliminating the time spent for the try, fail, and wait of transmissions.

The proposed approach is based on adapting the values of CW depending on the channel congestion level. In IEEE 802.11e the value of CW is incremented whenever a station fails to transmit due to a collision. This would imply that when the channel is highly congested CW would acquire values distant from CWmin and close to CWmax. Similarly, when the channel is free, CW values would be close to CWmin and distant from CW_{max}. Hence, it is feasible to estimate the channel congestion level by taking into consideration the current value of CW. We use a very simple approach to estimate this level. In this approach, we start from the fact that CW value ranges in the interval [CW_{min}, CW_{max}], then we compute its relative distance ($CW_{current}$ – CW_{min}) compared to the maximum distance ($CW_{current}$ – CW_{min}) as an indication for channel congestion level. It follows that the estimated link congestion ratio in the proposed Adaptive scheme can be written as:

$$Ratio = \frac{(cw_{current} - cw_{min})}{(cw_{max} - cw_{min})}$$

To optimize further this ratio, we use a weight that reflects the certainty of the channel estimation. This certainty is decreased as time elapses between consecutive transmissions. Hence our current estimation of the channel has a greater value if the time difference between successive transmissions is negligible. In the proposed scheme, the ratio is weighted as follows.

$$ratio = weight \times \frac{(cw_{current} - cw_{min})}{(cw_{max} - cw_{min})}$$

For instance, the weight of the ratio would be very small if current channel estimate is used in a transmission that occurred several minutes ago. However the ratio would be highly weighted if the difference in time between estimation and transmission is of the order of milliseconds. To obtain some preliminary simulation results, the weight was fixed in this paper to a value of 0.9. Indeed, the weight converged to this value after several tests. This is due to fact that video streaming is characterized by transmission occurring at very small time intervals

The CW value of the proposed adaptive scheme, CW_{new} can be given then as follows

$$cw_{new} = ratio \times (cw_{current} - cw_{\min})^2 + cw_{\min}$$

$$cw_{new} = weight \times \frac{(cw_{current} - cw_{\min})^2}{(cw_{\max} - cw_{\min})} + cw_{\min}$$

The ratio is a normalized value ranging from 0 to 1 that reflects the weighted degree of channel contention. This ratio would take a value close to 0 whenever the channel is free. Therefore CWcurrent would have a value close to CW_{min} and distant from CW_{max}. The value of this ratio would be close to 1 whenever the channel is congested. Therefore CWcurrent would have a value distant from CWmin and close to CW_{max}. Multiplying this ratio by the factor ($CW_{current} - CW_{min}$) and adding the result to CW_{min} would result in a value bounded by [CW_{min}, CW_{max}]. This value of CWnew would be a good representation of the backoff timer value needed for transmission for the current traffic priority taking into account the current network conditions.

4. Simulation Scenario

The simulation topology of this scenario is simple. It consists of 8 mobile nodes: 4 source nodes and 4 destination nodes. Each node is transmitting with a different priority. Node 1 is given a higher priority than Node 2, which is given also a higher priority than Node 3. Node 3, in its turn, is given a higher priority than Node 4. Each source is a Constant Bit Rate source over UDP (User Datagram Protocol). The size of a transmitted packet is 512 bytes. Transmission rate of a node is 600Kbps. We assumed that the nodes are in transmission range at a constant distance of 195 m. The simulation time lasted for 80 sec.

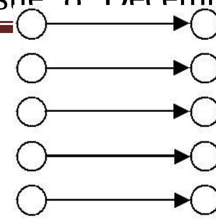

FIG 4.1 Scenario for the proposed Problem

To model voice traffic the simulations use ITU-T G.729 standard. G.729 is supported widely in VoIP products. In G.729, the voice is encoded at the rate of 8 kbps and with 20 or 40 bytes payload size in a packet. The voice quality can be degraded compared to another widely used standard, G.711, because the compression in G.729 can be lossy. However G.729 requires less bandwidth. The payload size is 20 bytes. With packet overhead, the rate required is 26.4 kbits/s. In order to simplify the situation in the network, there are only two kinds of traffic, voice and data

5. Simulation results

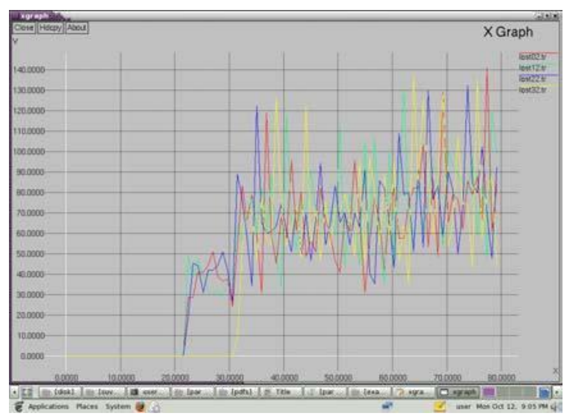

Fig 5.1 VoIP Traffic using exponential in 802.11a

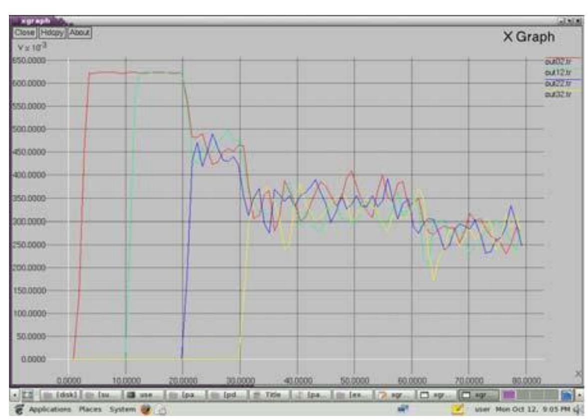

Fig 5.2 Bit rate for 8 nodes

In the above Fig, Node1 starts transmitting at time T = 1.4 sec while Node 2 starts transmitting at time T = 10sec Node 1 is the only transmitting N node using the entire available bandwidth. The bit rate plot experiences heavier oscillations and reduction as the number of transmitting nodes increases. Oscillations are reflected in Heavy disorders in network performance

The system will identify the data that exist the all the Packet loos rate for 8 nodes to identify the system approach to control the identity of the Packet

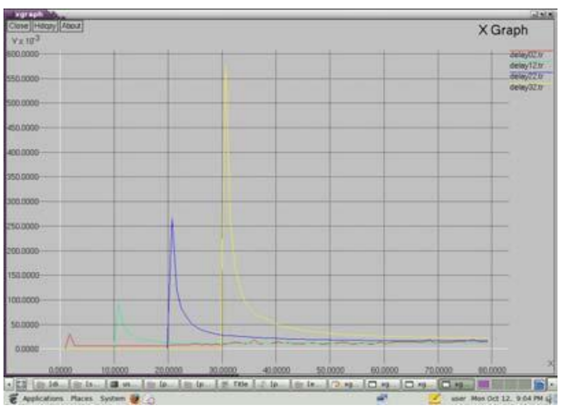

Fig 5.3 Packet loss rate for 8 nodes

These results reveal the bad behavior of IEEE 802.11a networks when many nodes are transmitting offering no protection for streaming traffic. QoS is not guaranteed in IEEE 802.11a MAC layer

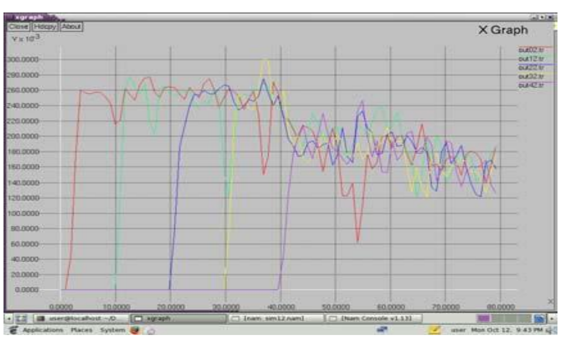

Fig 5.4 Packet Delay for 8 nodes

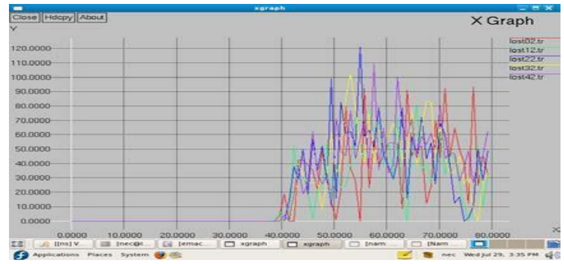

Fig 5.5 Packet loss for 10 nodes

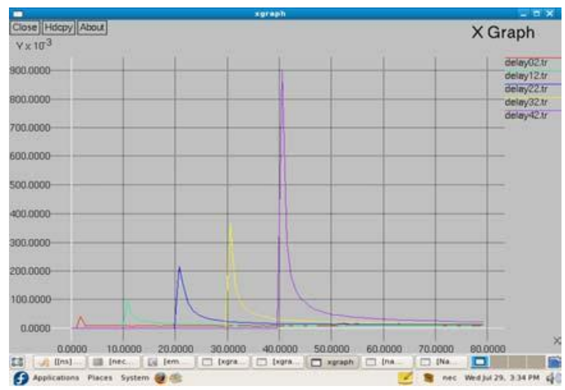

Fig 5.6 Packet Delay for 10 nodes

The simulation experiments aim to investigate the performance of the enhanced DCF scheme in the IEEE 802.11and compared with the IEEE802.11e Wlans in terms of throughput, access delay, medium utilization and packet loss probability in the presence of voice traffic. The average throughput is calculated as the amount of data actually delivered to the destination during each time unit. Many factors affect the throughput, including the efficiency of collision avoidance, medium utilization, latency, and control overhead. The access delay is defined as the time elapsed from the arrival of a packet from the higher layer to the MAC layer until the start of the successful transmission on wireless medium. Access delay to find out how well the enhanced DCF scheme accommodates the VoIP services. Medium utilization is referred to the percentage of time that is used for successful transmission. Moreover, the packet loss probability is calculated as the ratio of the number of lost packets over the number of generated packets.

Wireless Local Area Networks (Wlans) offer the flexible data communication systems to provide location independent network access between computation and communication devices using waves rather than a cable infrastructure. Many MAC protocols have been reported to manage and control the shared wireless medium. This paper has proposed an enhanced MAC scheme in order to enhance the performance of the IEEE 802.11 Wlans by using well-known Network Simulator (NS-2). By comparing the Results we can notice that VoIP in IEEE802.11e works better than in IEEE802.11a.

6. Conclusion

In this project the performance of the IEEE 802.11a and IEEE 802.11e systems have been evaluated. We have also proposed a new adaptive differentiation technique for resetting the value of the contention window after each successful transmission. The proposed adaptive technique takes into account the current level of link utilization when resetting such value. We have performed several simulations, for different Scenarios, using NS -2, to evaluate the proposed technique compared to IEEE 802.11a and IEEE 802.11e. During the evaluation, we have focused on three parameters: bit rate, end-to- end packet delay, and packet drop rate. And the results reveal that the proposed traffic gives better results than IEEE802.11a.

References

[1]. P. Chatzimisions, V. Vitas as, A. Boucouvalas and M. Tsoulfa, Achieving performance enhancement in IEEE 802.11 Weans by using DIDD back off mechanism, International Journal of Communication Systems, vol. 20, no. 1, pp. 23-41, 2007.

[2]. A. Bacioccola, C.Cicconetti, G.Stea User level performance evaluation of VoIP using NS-2, Computer Networks Group, 2007

[3] I. -H. Lin and J.-Y. Pan, Throughput and delay analysis of a novel back off algorithm for IEEE 802.11 Weans, Proc. Wireless Telecommunications Symposium, pp. 85- 90, 2005.

[4]. L. Autoscore, and M. L.Lobina. Play out buffering in ip telephony: a survey discussing problems and approaches .IEEE Commun surveys Tut, vol. 8, 2006

[5] Mamun I. Abu-Tair, Geyong Min, Qiang Ni, Hong Liu “Adaptive Medium Access control for VoIP services using IEEE 802.11 Networks”, IEEE 2008

[6]Mamun abu Tair.Geyong Min,Qiang Ni,Hong Liu “Adaptive medium access control for VoIP services using IEEE 802.11 Networks.