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ADAPTIVE QUALITY OF SERVICE MEDIUM ACCESS CONTROL PROTOCOL FOR IEEE 802.11 BASED MOBILE AD HOC NETWORK

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Abstract: Mobile ad hoc network is an infrastructure less wireless multi hop network with heterogeneous mobile nodes dispersed in wireless communication zone. MANET has different application in different fields, due to its distributed, adaptive and self-formation capabilities. Providing quality of service communication is one of the important considerable issue in MANET. One of the major fact to achieve the QoS communication is efficient MAC protocol. This paper defines an adaptive – QoS MAC protocol (AQMP) for IEEE 802.11 based MANET. AQMP protocol improve the QoS based on majorly four considerations I). Prioritize the nodes based on their network load, II). Assignment of nodes for medium access, III). Prioritize the traffic based on their sensitivity, and IV). Assignment of MAC settings to prioritized traffic. Performance results indicates that proposed MAC protocol out perform in comparison with existing adaptive MAC protocols.

Index Terms: MANET, MAC, QoS, Priority, Voice.

I. INTRODUCTION

Mobile ad-hoc network, contains wireless mobile heterogeneous devices circulated in wireless communication zone to provide communication in critical sensitive applications without pre-defined infrastructure or central base station [1]. Devices communicate each other directly if both are exist in a radio communication medium of one another, else communication is possible with the help of intermediate nodes. This type of communication is called multi hop communication, and which leads to degradation of quality of communication. More over peer to peer network causes the implementation QoS in communication to more challenging attribute [2]. Particularly providing QoS in communication for real time traffic is more challenging in MANET. Quality of service is the strength to give the dissimilar priorities to different applications, data and end users. QoS goal is to achieve particular level of efficient performance in communication.

QoS achievement is parameters are bandwidth, delay and packet loss. Based on application and type of communication QoS parameters can vary. Real time traffic such as voice and multimedia are very sensitive to delay, so QoS parameters for voice and multimedia communication is delay. Achieving QoS communication is challenging in MANET is more challenging in comparison with wired or wireless infrastructure based network due to its characteristics such as mobility, heterogeneity in mobile devices and constrained resources. Moreover shared medium in MANET is also limited. One can achieve the QoS communication in any network through efficient Medium access control protocol and it is come under the data link layer. The primary functionality of MAC protocol in MANET is to access the channel for communication of packets between two nodes. MAC protocol is responsible to manage and coordinate multiple radio communication mediums. This protocol responsible to solve the issues between different channel accesses [3-4].

Thus MAC protocol is responsible for medium access there by reliable and efficient data transmission between communicating entities. So efficient MAC protocol is a key to achieve QoS communication in MANET.

In this paper we achieve the QoS communication in MANET by the following contributions.

- I). Prioritize the nodes based on their network load
- II). Assignment of nodes for medium access
- III). Prioritize the traffic based on their sensitivity, and
- IV). Assignment of MAC settings to prioritized traffic

II. BACKGROUND

The IEEE 802.11 specifies the two methods for channel access. First method is distributed coordination function (DCF), which is contention based approach for channel access and second method is point coordination function (PCF), which is contention free approach for channel access [5-6]. Distributed coordination function work based on the principle of Carrier Sense multiple Access/Collision Avoidance (CSMA/CA), which is default approach for accessing channel in wireless networks. It is provide equal opportunity or priority to all the nodes to access the medium. For instance, any node want to participate in communication, then it check the medium for its status whether it is engage in other communication or it is free. If it finds the medium as free, then node needs to wait for some predefined interval of time i.e., DIFS (distributed coordination function inter frame space). Again it check whether the medium is free or busy, if it is free then it start the communication [7-8]. If it finds that medium is not free then node wait for the wait for random interval of time and sense the medium again. The another mechanism of IEEE 802.11 to access the channel is Point coordination function [9,] which access the medium through centralized method by controlling and coordinating the channel access mechanism with the help of polls other nodes and permit them to access the

contention free channel. Both, point coordination function (PCF) and Distributed coordination function (DCF) used to access the contention free channel and are used for inter frame spacing coordination and controlling.

Later IEEE 802.11 e is developed to provide QoS communication by developing EDCA [10] mechanism, which is the enhancement of existing 802.11 distributed coordination function (DCF) by enabling the distributed channel access shown as Figure 1. It is the method of enabling the service differentiation to the various traffic patterns. IEEE 802.11 e EDCA is shown in figure 1. Enhanced DCF categorized the incoming traffic towards the node into four sub categories i.e., access sub category for best effort, voice, video and background traffics. EDCA defines different channel access settings to different sub categories incoming traffics, so that every incoming traffic has its own priority to access the channel. Parameters setting of EDCA are $CW_{max}()$, $CW_{min}()$, $AIFS()$, and $retry-limit$. Every sub access subcategory has buffer space, queue and behave as a self-sufficient back-off entity.

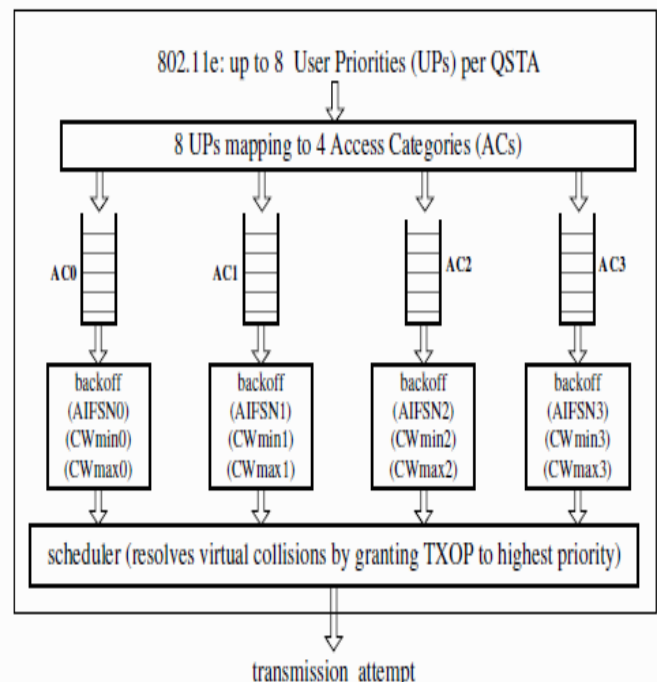


Figure.1 IEEE 802.11e EDCA

III. PROPOSED WORK

Prioritize the nodes based on their network load and Assignment of nodes for medium access

This work prioritize the nodes based on their network load. This prioritize of nodes is done because of providing the authority to nodes to participate in medium access contention. Moreover here we do not have any frame format or structure. Only those nodes can participate in channel contention, which has the sufficient network load to handle the current communication scenario. When the number of nodes has the enough load to participate the channel contention, which interns improve the channel utilization. In contrast, when the number of nodes have the not enough load to participate the channel contention, which interns reduces the channel utilization. Network load of nod is commuted as follows. Network load status is computed by queuing theory by computing the packet delay inside the node queue and buffer space occupied the packets . Without the loss of generality, we calculated the number of packets is arrived towards the queue (A) in the particular interval of time It is calculated by exponential weighted moving average method [11] in Eqn (1) as

$$A = \alpha A_c + (1 - \alpha) A_p \quad (1)$$

Where, (A_p) is an average number of packets arrived to queue during preceding time interval, and (A_c) is an average number of packet arrived to queue during present time interval. α and β are known as weighted constants, and values of α and β are in the range of 0 to 1. Initially we assume that the no packets are arrived and considered it as zero event.

In the same manner, we calculated the average packets departed from the queue of node in particular time interval as Eqn (2) follows.

$$T = \alpha D_c + \beta \quad (2)$$

Where, (D_p) is an average number of packets departed from queue during preceding time interval, and (D_c) is an average number of packet departed from queue during present time interval. α and β are known as weighted constants, and values of α and β are in the range of 0 to 1. Initially we assume that the no packets are departed and considered it as zero event.

- If packet arrival rate is more than departure rate towards the node buffer queue. Then queue size will get increases and it does not have chance to participate in channel contention
- If packet arrival rate is less than departure rate towards the node buffer queue. Then queue size will get increases here also, but node has chance to participate in channel contention.
- However this condition is satisfied only to the infinite size queue size buffers, but MANET has limited queue size buffers. In this conditions, node can participate in channel contention effectively only when node queue size does not reach the certain threshold level and packet waiting time inside the queue should also less than certain time limit.
- In this paper we are only considering the packet waiting time inside the node queue. If it is low enough to certain threshold value then it indirectly reduces the queue size at node buffer. The threshold value is computed based on the transmission delay form source to destination and number of hops needed to reach the destination.
- Thus, we calculated the packet waiting time inside the node queue by following Eqn (3) as, packet waiting time inside the node queue

$$(W_t) = (1 / (D - A)) * \text{Size of packet} \dots(3)$$

- If $(W_q) \geq \text{Threshold}$, then the node will not enter into the channel contention. We assign this node as the low priority nodes.
- Only high priority nodes are participated in channel contention mechanism.

Initially every node assign with high priority to act in channel contention mode. So every node has ability to act in a channel contention mode. During communication, each node computes the

packet waiting time inside the node queue (W_q) in that time period and if it finds that the node packet waiting time inside the node queue (W_q) value is greater than the threshold value then the node loss its priority to act in channel contention, it means that node has high load. However, after some fixed amount of time node again compute the

packet waiting time inside the node queue (W_q), if it is less than the threshold value then node get the higher priority and act in channel contention mode.

Prioritize the traffic based on their sensitivity

According to calculated *Rating factor* values and ITU-T recommendation [12], voice sessions are categorized

1. Voice session value above 80 is higher priority
2. Voice session value between 70 to 80 is medium priority
3. Voice session value below 70 is low priority

According to calculated *Rating factor* values and ITU-T recommendation, voice sessions are categorized

We assume that at application layer video data is encoded with H.264/AVC [13] and it provides substantial importance in in compression efficiency .H.264 supports temporal scalability and data partitioning with different priorities, and known as partitions . H.264's VCL layer that divides the original streams through DP (data partitioning). Whenever DP is aided, alongside the

slices of IDR (Instantaneous Decoder Refresh) pictures, additional three other types of slices are generated i.e., partition A , Partition B and Partition C as shown in Table 1. All these slices are directly passed through NAL in order with hearer indicting with slice type.

1. First slice transmitted is PSC.
2. Next slice transmitted as IDR
3. Next slices transmitted as partition A, partition B, and partition C.

Priority of the slices

1. First slice transmitted is PSC. Higher priority
2. Next slice transmitted as IDR. Medium priority
3. Next slices transmitted as partition A (Medium priority)
4. Partition B, and partition C (Low priority).

Table 1. Priority of the slices

Slice type	NRI Value	Priority
Parameter set information	11	Higher priority
IDR picture slice	10	Medium Priority
partition A	10	Medium Priority
partition B	01	Low Priority
partition C	01	Low Priority

Assignment of MAC settings to prioritized traffic

Assignment of MAC setting to voice data is explained in details in our previous paper, reader can refer the paper [2]. We plausibly divided video access category into three parts, such as AC_video_1, AC_video_2 and AC_video_3. Where, AC_video_1 has highest chance to contact the medium and AC_video_2 get the chance to access the medium only when AC_video_1 contains no packets. AC_video_3 has lowest priority to access the medium.

Our proposed system animatedly assign the higher priority video slice to AC_video_1 so that the

performance of video can be enhanced and it grasps the user agreeable level. Projected system is exposed underneath

Algorithm 1:- dynamic video slice mapping mechanism

AC_Voice capture the medium

1. Video slice [i] subject to AC_voice_3
2. If computational time == 1
3. For
4. video slice [i]
5. Check the NRI value
6. End for
7. While Tuning time == 1
8. If NRI value == 11
9. Video slice [i] subject to AC_voice_1
10. Else NRI value == 10
11. Video slice [i] subject to AC_voice_2
12. Else Video slice [i] subject to AC_voice_3
13. End if
14. End if
15. Wait for tuning time == 1
16. End if

IV. PERFORMANCE ANALYSIS

We calculate the proposed work performance with NS-2 [14] simulator by necessary extension in predefined MAC libraries of simulator. Results are shown in figure 2 and 3. Performance calculation metrics are Throughput and packet delivery ratio. Performance results are compared with existing QoS MAC protocol . Simulation parameters are shown in Table 2.

Table 2: Simulation Parameters

Network Parameters	Values
Number of Nodes	50 to 100
Link layer type	Logical Link(LL)
MAC Type	802.11e, DSMA
Routing	AODV

Traffic	CBR, FTP
Network Area	1500mx1500m
Transmission Range	250m

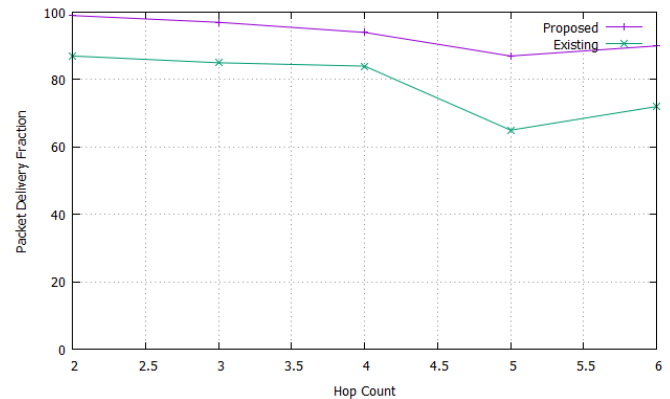


Figure 2:- packet delivery fraction comparison

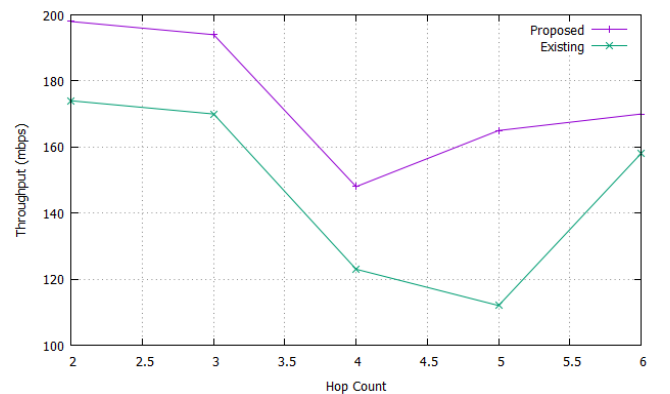


Figure 3:- Throughput (mbps) comparison

Performance results shows that the proposed work extend the quality of voice and video data dynamically mapping the packets to suitable MAC access categories and thereby enhancing the throughput and packet delivery

V. CONCLUSION

This paper discussed the adaptive MAC protocol to enhance the real time traffic performance in MANNET such as video data and voice data. Proposed work dynamically maps the video and voice data to suitable MAC access categories to enhance the performance. Quality of voice data is computed by rating factor and quality of video quality is determined by compression slice type by

H.264/AVC. Simulation results indicating that proposed algorithm improve the quality of video and voice over 802.11 based MANET.

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