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PERFORMANCE EVALUATION OF MULTIMEDIA STREAMING OVER MOBILE ADHOC NETWORKS USING MAPPING ALGORITHM

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Abstract

The main objective of this paper is to propose a novel method for enhancing the Quality of Service (QoS) of multimedia applications in wireless adhoc networks. The enhancement is achieved by implementing a cross layer mapping algorithm, between application layer and Medium Access Layer where Connectionless Light Weight Protocol (UDPLite) is used in transport layer that supports multimedia applications. The Proposed method achieves 17% improvement in reduction of delay and 5% improvement in PSNR as compared to the conventional methods under heavy traffic conditions.

Keywords

Enhanced Distributed Channel Access, MANETs, PSNR, UDPLite, Video Streaming.

1 INTRODUCTION

Recent advancements in computing techniques have become an integral part of wireless communication networks. Mobile Ad hoc networks (MANETs) have emerged amid the unprecedented growth of Internet and are increasingly attracting attention because of its ability to connect across nodes without relying on pre-existing network infrastructure. The widespread emergence of real-time voice, audio and video applications, stimulates the successful development of viable technologies to provide these multimedia applications over mobile adhoc networks. The performance of MANET is affected by various factors such as mobility of node, battery life and routing protocols, topology change etc… Hence providing Quality of service for multimedia applications in adhoc networks is difficult.

Quality of Service requirements of multimedia applications in adhoc networks have been supported by IEEE 802.11e standard. 802.11e defines four Access Categories(ACs) with different transmissions priorities. The transmission priority is the probability of successfully earning the chance to transmit when individual ACs are competing to access the wireless channel. Higher the transmission priority, better is the opportunity to transmit. But in a wireless channel the unavoidable burst loss, due to excessive delays and limited bandwidth there are challenges for good transmission over wireless network.

In this paper we argue that for 802.11e based adhoc networks, a partial checksum approach at the transport layer along with an adaptive cross layer mapping scheme between application layer and Medium Access Layer, can improve the performance of video transmission. In this paper an attempt has been made to get benefits of UdpLite along with cross layer approach. The rest of the paper is organized as follows. In this section 2 we discuss the aspects of anUdpLite and in section 3 enhanced Distributed

Channel Access(EDCA)and in section 4 adaptive cross layer mapping algorithm. In section 5 we discuss about proposed system. Section 6 establishes system simulation model and Section 7 gives results to illustrate the performance while conclusion are drawn in section 8.

2 UDPLITE

The quality of video can be increased by enabling the application layer to specify about the importance of packets and those packets can be preserved by the UDPLite protocol [1] in the transport layer along with Cross Layer Mapping approach. The notion of application-layer over transport-layer protection is not new and hence traditional real- time multimedia services have been realized as Realtime Transport Protocol (RTP) over User Datagram Protocol (UDP). User Datagram Protocol (UDP) is an unreliable protocol that is suitable for delay sensitive applications such as real-time media applications that are sensitive to network delays. UDPLite is an extension to UDP that even needs damaged data to be delivered rather than discarded by networks, so it allows partial checksums on multimedia data by enabling the applications to specify, the sensitive and insensitive parts of the multimedia stream on a per-packet basis. Errors in the sensitive part cause a packet to be discarded whereas an error in the insensitive part allows it to be delivered. The check sum is carried out on the sensitive part of the packet. UDP has a strict checksum where corrupted packets will be discarded if they contain any transmission errors. The UDPLite protocol allows the application to receive the corrupted packets instead of dropping them altogether. This is achieved by a partial checksum which only covers a fixed amount of sensitive data. Integrating UDPLite into existing UDP framework is simple. The length field in the UDP header is replaced by the coverage field, which signifies the number of bytes of the packet that are to be checksummed. With a checksum coverage value replacing the packet length, UDPLite packets are treated as classic UDP packets with the checksum enabled. To address security concerns and handle the multiplexing of other transport level flows, the packet header should always be checksummed. If corruption occurs in the Sensitive region or in the header, the packet is dropped at the receiver otherwise the packet is passed up to the application through the interface.

Source Address			IP Pseudo-header
Destination Address			
Zero	Proto	UDP Length	
Source Port		Destination Port	UDP lite header
CheckSum Coverage		Checksum	

Figure 1: The UDPLite Header

The UDPLite protocol headers are shown in Figure 1. Shaded fields are the fields of the pseudo header provided by the IP layer, and white fields belong to the UDPLite header. The UDPLite checksum covers the conceptual IP pseudo-header in order to protect against misrouted packets.

The Checksum Coverage field in the UDP Lite header denotes the number of octets (counting from the first octet of the header) that are covered by the checksum. The value of Checksum Coverage is zero indicating that the entire UDP-Lite packet is covered by the checksum. This explains that the value of the Checksum Coverage field MUST be either 0 or at least 8. The UDP Lite header and the IP pseudo-header are always verified by the checksum, which means that the least acceptable value of the coverage field is eight (the number of bytes in the UDP Lite header). A UDP-Lite packet with a Checksum Coverage value of 1 to 7 MUST be discarded by the receiver.

3. EDCA

IEEE 802.11e supports quality of service by introducing priority mechanism. All types of data traffic are not treated equally as it is done in the original standard, instead, 802.11e supports service differentiation by assigning data traffic with different priorities based on their QoS requirements. Furthermore, four different Access Categories (ACs) have been defined each for data traffic of a different priority. Access to the medium is then granted based on the priorities of data traffic, such that each frame with a particular priority is mapped to an Access Category, and service differentiation is realized by using a different set of contention parameters to contend for the medium, for each AC.

In IEEE 802.11e, the AP and STA that provides QoS services are referred to as QAP (QoS Access Point) and QSTA2 (QoS Station) respectively, and the BSS they are operating in is called QBSS (QoS Basic Service Set). IEEE 802.11e introduces a new coordination function, called Hybrid Coordination Function (HCF), to provide QoS support. Subsequent sections describe HCF together with the detailed description of its service differentiation mechanism.

IEEE 802.11e defines a new coordination function called Hybrid Coordination Function (HCF). HCF is a centralized coordination function that combines the aspects of DCF and PCF with enhanced QoS mechanisms to provide service differentiation. HCF provides both distributed and centrally controlled channel access mechanisms similar to DCF and PCF in the original standard. The distributed, contention-based channel access mechanism of HCF is called Enhanced Distributed Channel Access (EDCA), and the centrally controlled, contention-free channel access mechanism is called HCF Controlled Channel Access (HCCA).

IEEE 802.11e introduces Transmission Opportunity (TXOP), defined as the time period during which a QSTA has the right to transmit. In other words, in 802.11e when a station gets access to the medium, it is said to be granted the TXOP. TXOP is characterized by a starting time and a maximum duration, called TXOP Limit. As a QSTA gets the TXOP, it can then start transmitting frames such that the transmission duration does not exceed the TXOP limit. TXOP Limit is specified by the QAP.

The EDCA provides differentiated, distributed access to the medium using different priorities for different types of data traffic. The detailed description of the components and operation of EDCA is presented next.

3.1 Access Categories (ACs)

EDCA defines four Access Categories (ACs) for different types of data traffic, and service differentiation is introduced such that for each AC, a different set of parameters is used to contend for the medium. These parameters are referred to as EDCA parameters and are described in the next subsection.

Frames from different types of data traffic are mapped into different ACs depending on the QoS requirements of the traffic/application the frames belong to. The four Access Categories are named AC_BK, AC_BE, AC_VI and AC_VO, for Background, Best Effort, Video and Voice data traffic, respectively. To simplify the notations, we assign AC_VO as AC3, AC_VI as AC2, AC_BE as AC1, and AC_BK as AC0. Each AC has its own buffered queue and behaves as an independent backoff entity, i.e., an AC, where each queue has its own AIFS and maintains its own Backoff Counter (BC). When there is more than one AC finishing the backoff at the same time, the collision is handled in a virtual manner. That is, the highest priority frame among the colliding frames is chosen and transmitted, and the others perform a backoff with increased CW values. The priority among ACs is then determined by AC-specific parameters, called the EDCA parameter set.

Each frame from the higher layer arrives at the MAC layer along with a priority value. This priority value is referred to as User Priority (UP) and assigned according to the type of application/traffic the frame belongs to. There are four different priority values ranging from 0 to 3.

Priority	**User Priority (UP)**	**Access Category (AC)**	**Designation**
Lowest	0	AC_BK	Background
-	1	AC_BE	Best Effort
-	2	AC_VI	Video
Highest	3	AC_VO	Voice

Table 1: User Priority (UP) to Access Category (AC) Mapping

At the MAC layer, a frame with a particular UP is further mapped to an AC. AC are derived from the UPs as illustrated in Table 1.

3.2 EDCAF (Enhanced Distributed Channel Access Function)

Every station maintains four transmit queues one for each AC, and four independent EDCAFs (Enhanced Distributed Channel Access Function), one for each queue, EDCAF is an enhanced version of DCF, and contends for the medium on the same principles of CSMA/CA and back off, but based on the parameters specific to the AC it is contending for. Next section discusses these parameters, referred to as EDCA parameters.

EDCA Parameters

An EDCAF contends for medium based on the following parameters associated to an AC. The set of EDCA parameters are:

- **AIFS -** The time period the medium is sensed idle before the transmission or backoff is started.
- **CWmin, CWmax -** Size of Contention Window used for backoff.
- **TXOP Limit -** The maximum duration of the transmission after the medium is acquired.

3.2.1 AIFS (Arbitration Inter-Frame Space)

The minimum time period for which the medium must be sensed idle before an EDCAF/station may start transmission or backoff process. It is not the fixed value DIFS, as it is in DCF, but is a variable value, AIFS, that depends on the AC for which the EDCAF is contending for. AIFS is derived from the following equation:

AIFS = AIFSN x aSlotTime + aSIFSTime,

where aSlotTime is the slot time, aSIFSTime is the SIFS time period and AIFSN (Arbitration Inter-Frame Space Number) is used to determine the length of the AIFS. AIFSN specifies the number of time slots in addition to the SIFS time period the AIFS consists of. Different AIFSN values are used for different ACs such that the high priority ACs use smaller values compared to the low priority ACs. The minimum possible value of AIFSN is 2.

The default AIFSN values for all four ACs can be seen in Table 1. Figure 2 further explains how priority is given to different ACs based on the AIFS time periods.

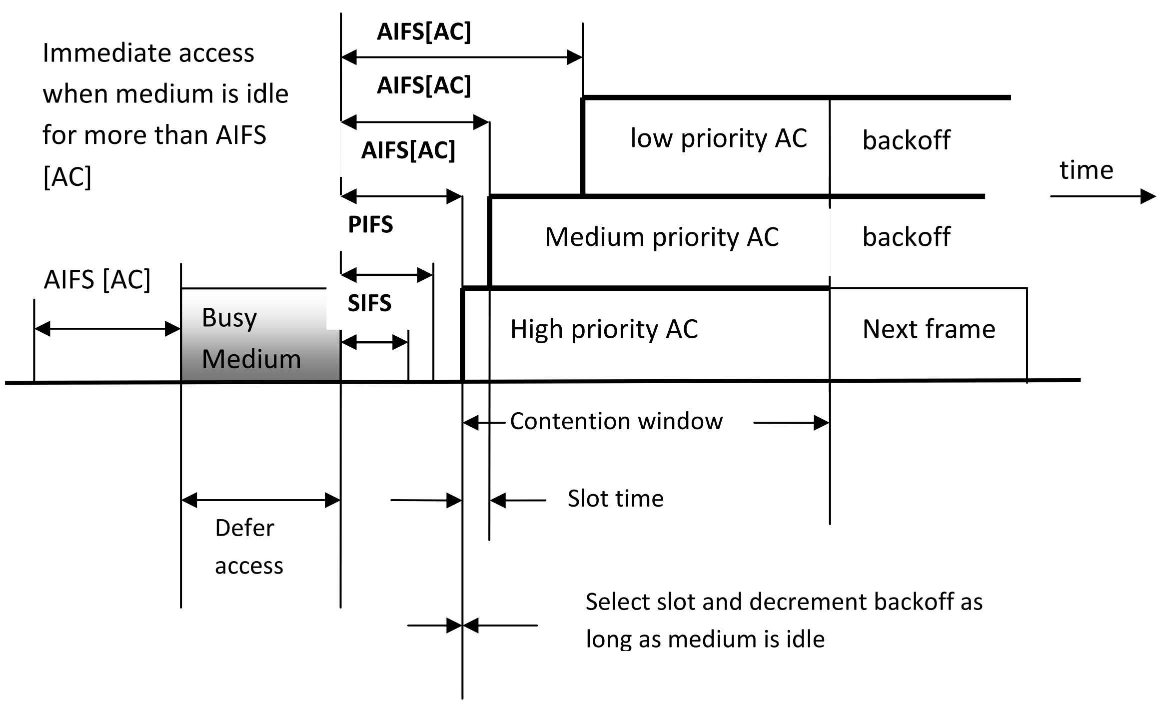

Figure 2: Prioritization based on AIFS

The smaller AIFSN value for a higher priority AC explains that the corresponding EDCAF has to wait shorter time period before it can start transmission or counting down its backoff timer compared to the EDCAF for a low priority AC. In this way, the higher priority ACs are guaranteed greater share of the bandwidth. Moreover, smaller AIFS lengths ensure that the higher priority ACs will not suffer from long delays, which are very critical for the delay-sensitive applications/traffics. The lower priority ACs may suffer from longer delays because of the larger AIFS durations they have to wait, but since these ACs are designed for delay-tolerant applications/traffics, certain amount of delays do not degrade their performance beyond an acceptable limit.

3.2.2. CWmin and CWmax

The minimum and maximum Contention Window size limits are not fixed as it is in DCF, but are variable depending on the AC. The higher priority ACs has smaller CWmin and CWmax values compared to lower priority ACs A smaller Contention Window for an AC will cause the corresponding EDCAF to choose smaller random backoff values, and thereby waiting shorter time period in addition to AIFS as the medium becomes idle. It gives such an AC priority over the AC with a larger Contention Window, which results in larger backoff values and thereby longer delays.

As seen in Table 1, for the commonly used Physical layer DSSS, the CWmin values for lower priority ACs, AC_BE and AC_BK, are same as it is for the legacy 802.11 DCF, but these values for higher priority ACs, AC_VO and AC_VI, are as small as one half or quarter of those of the lower priority ACs. This results in smaller backoff values for the high priority ACs and thereby shorter medium access delays. The negative aspect of small Contention Window sizes for higher priority ACs is that they suffer from higher number of collisions. The reason is, that the probability of choosing the same backoff values or counting the backoff timers to zero at the same time increases with the decreasing size of Contention Windows. CWmax values for high priority ACs are also set such that they are equal or less than the CWmin values for the lower priority AC This shows that after doubling the Contention Window size in case of an unsuccessful transmission, i.e., collision, its size still remains smaller than the CWmin size of lower priority ACs. Furthermore, it also indicates that while a low priority AC has to double its CW size after each unsuccessful transmission, until it reaches the CWmax, and with higher probability, has to choose a bigger backoff value for each retransmission, the Contention Window size of a high priority AC becomes constant after fewer retransmissions, allowing it to consistently choose smaller backoff values and thereby winning access to the medium. In this way, high priority AC is given consistent and greater share of the bandwidth in the situations when the network has become congested. On the other hand, this may severely degrade the performance of the low priority ACs since they might not be able to decrement their backoff timers because of the smaller post backoff durations of the higher priority ACs.

3.2.3 TXOP (Transmission Opportunity)

TXOP is the time duration an EDCAF may transmit after winning access to the medium. TXOP is characterized by a maximum duration, called TXOP Limit. As an EDCAF gets the TXOP, it can then start transmitting frames such that the transmission duration does not exceed the TXOP Limit. The transmission duration covers the whole frame exchange sequence, including the intermediate SIFS periods and ACKs, and the RTS and CTS frames if RTS/CTS mechanism is used.

Table shows the default TXOP limits for different ACs. A non-zero value of TXOP Limit indicates that the EDCAF may transmit multiple frames in a TXOP, provided that the transmission duration does not exceed the TXOP Limit and the frames belong to the same AC. This is then referred to as Contention Free Bursting (CFB).

The consecutive frame transmissions in a TXOP are then separated by SIFS time periods instead of AIFS plus the post backoff periods, as illustrated in Figure 3. It is important to note that the multiple frame transmission is granted to EDCAF (or AC) and not to the station, i.e., it is only allowed for the transmission of frames of the same AC as of the frame for which the TXOP was obtained.

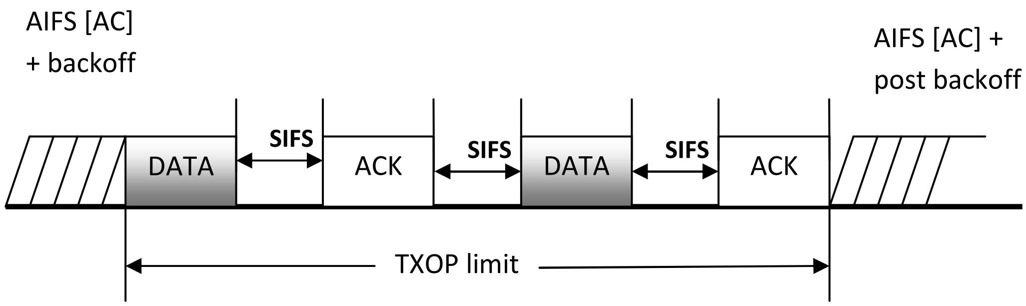

Figure 3: Contention Free Bursting

3.2.4 ARCHITECTURE OF IEEE 802.11e

Together with HCF and its two access mechanisms EDCA and HCCA, IEEE 802.11e also includes the two coordination functions from the original 802.11, DCF and PCF, in order to provide backward compatibility. Figure 4 illustrates the architecture of 802.11e MAC.

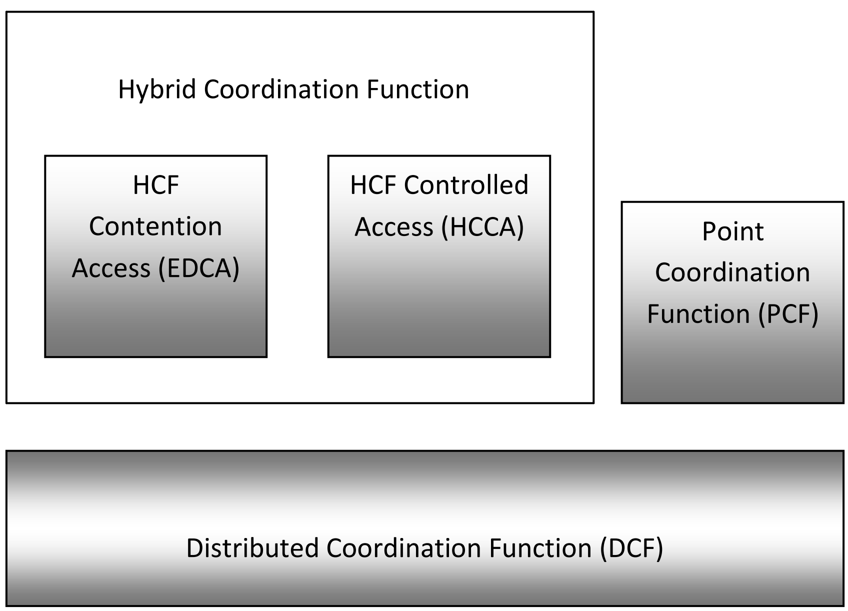

Figure 4: IEEE 802.1e MAC Architecture.

The centrally controlled, contention-free channel access mechanism of HCF, i.e., HCCA, uses a centralized coordinator called HC (Hybrid Controller), which is collocated in QAP. HC operates concurrently with the EDCA just like in the legacy 802.11, i.e., a Contention Free Period (CFP) is followed by a Contention Period (CP), such that the EDCA operates in CP while HC operates both in CP and CFP. This is in contrast with legacy 802.11 where PC can only operate in CFP. It indicates that HC is capable of polling QSTAs both in CP and CFP, and explains why it is referred to as Hybrid Controller.

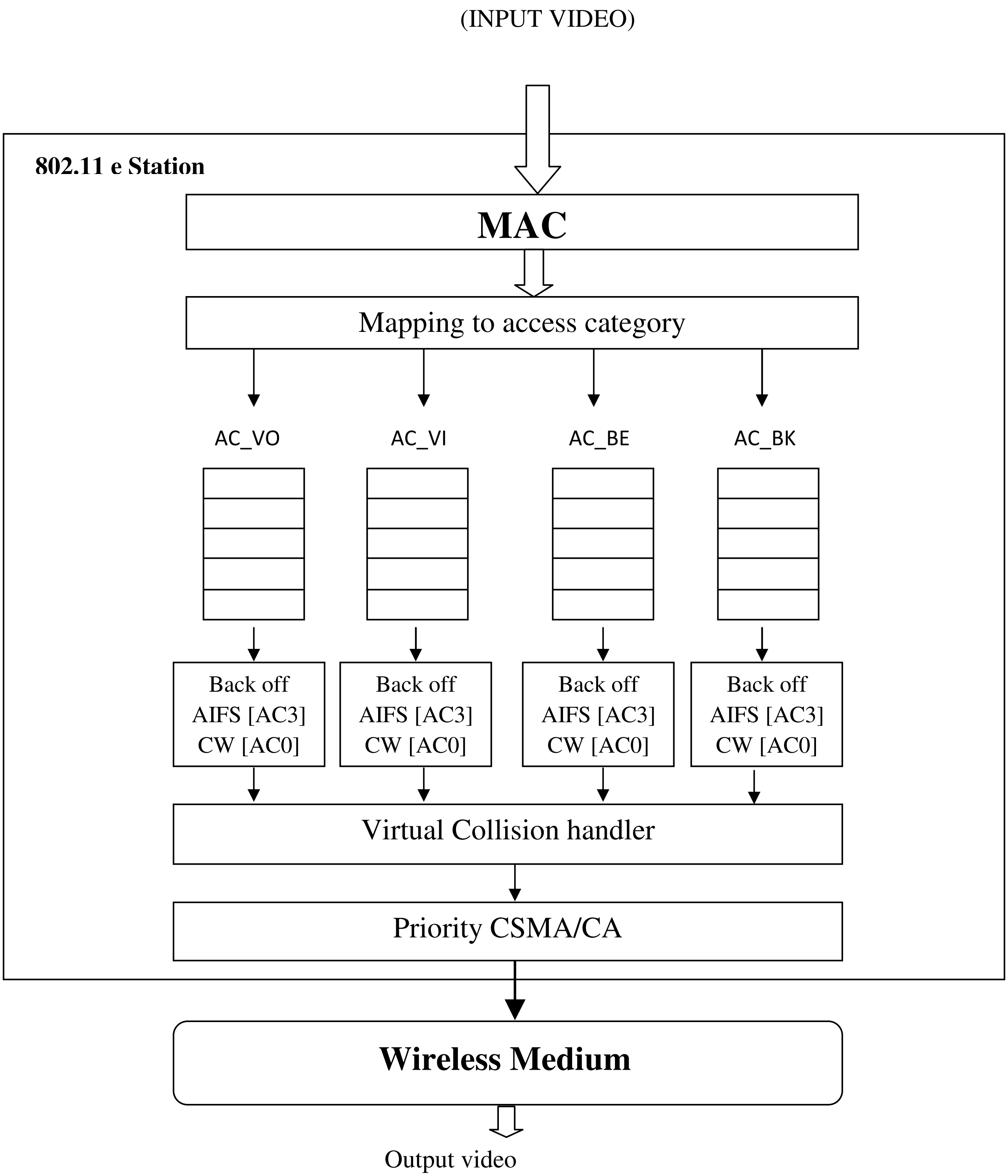

Figure 5: Four Access Categories in 802.11e

4 CROSS LAYER APPROACH

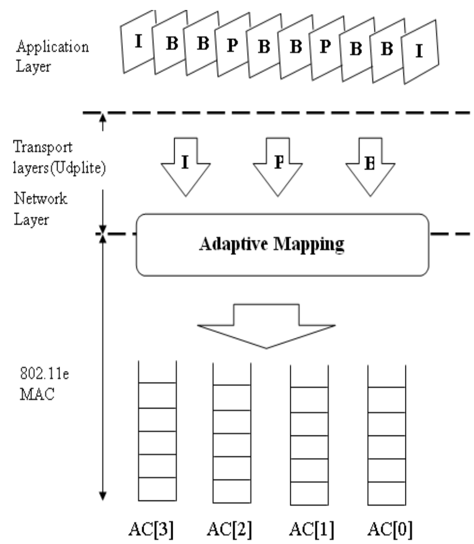

Figure 6: Architecture of adaptive mapping

From the Application Layer the video significance information is generated and transmitted to Enhanced Distributed channel access used in Medium Access Layer. In EDCA, packets arriving from Application layer and Transport layer(UDPLite layer) packets are tagged with four different user priorities and each priority is mapped to one of four Access Categories (ACs). The four different Access Categories (ACs) are Voice traffic, Video traffic, Best Effort traffic and Back Ground traffic that are represented as AC0, AC1, AC2 and AC3 respectively. Each AC maintains a local queue and an independent back off instance with a specific set of contention parameters. All ACs contend independently for access to the channel and internal collisions may occur, but are solved by allowing the AC with the highest priority to gain access to the channel.

To achieve differentiation between Access Categories(ACs), Contention Window(CW) parameters such as CW_{min} and CW_{max} and Arbitrary Interframe Space(AIFS) are used. to Instead of waiting for the normal Differentiated Inter Frame Space(DIFS) time, each AC waits a specific AIFS time. Higher priorities have lower values of the Contention Window(CW) parameters and AIFS. This leads to a higher fraction of the capacity and lower delays since the channel access frequency is increased. An additional parameter is the Transmission Opportunity (TXOP) that specifies the length of time the channel is occupied by a station. Depending on this limit, one or several packets may be transmitted when an AC has acquired the channel. Priority differentiation used by EDCA ensures better service to high priority class while offering a minimum service for low priority classes.

Priority	Access Category	Designation	AIFSN	CWmin	CWmax	TXOPlimit
3	AC_VO	Voice	2	(CWmin+1)/4-1	(CWmin+1)/2-1	0.003008
2	AC_VI	Video	2	(CWmin+1)/2-1	CWmin	0.006016
1	AC_BE	Best Effort	3	CWmin	CWmax	0
0	AC_BK	Background	7	CWmin	CWmax	0

Table 2: Default EDCA parameter set values

The AC with the smallest AIFS has the highest priority, and a station needs to defer for its corresponding AIFS interval. From Table 2 it is inferred that the smaller the parameter values (such as AIFS, CWmin and CWmax) the greater the probability to access the medium. Each Access Category within a station behaves like an individual virtual station, it contends for access to the medium and independently starts its backoff procedure after detecting the channel being idle for at least an AIFS period. When a collision occurs among different access categories within the same station, the higher priority access category is granted the opportunity to transmit, while the lower priority access category suffers from a virtual collision, similar to a real collision outside the station. This concept suits well when the traffic is very high, but the delay is still more and there occurs losses in packets though the traffic is very less since we have employed static mapping algorithm. If we employ dynamic mapping algorithm there will be unnecessary delays and high packet loss

In the cross-layer approach, the frames of the MPEG-4 video packets are dynamically mapped to the appropriate Access Category based on both the significance of the video frame and the network traffic load. Based on this Access Category the MPEG4 video packets are mapped dynamically to the appropriate Access Category(AC). Typically a MPEG video contains B-Frames ,I-Frames and P-Frames. Loss of Important frames in MPEG4 video stream would degrade the delivered video quality, whereas loss of B-Frame doesn't affect all the frames of Group of Pictures (GOP) but itself. Loss of I-Frame would cause all frames in Group of Pictures (GOP) to be undecodable. Based on the significance of the video frame, the channel access priorities are used to prioritize the transmission opportunity at the MAC layer are set with the I frame as the highest; the P frame below I but above B's priority, and the B frame set at the lowest priority. Mapping probability defined as Prob_TYPE is assigned according to its coding significance of video data. This way the important video data is alloted to high priority AC queue in 802.11e MAC layer which has been discussed in [2]. If allocating a frame into a lower priority queue is inevitable, the transmission allocating probability of lower significant frames is higher than that of important video frames. When larger Prob_TYPE is assigned to less important video frames the MPEG4 downward mapping probability relationship of the video frame types become Prob_B> Prob_ and Prob_I, values lying between 0 and 1. Moreover, to support dynamic adaptation to changes in network traffic loads, MAC queue length has been used as an indication of the current network traffic load. According to the IEEE 802.11e specification, when transmitted over an IEEE 802.11e wireless network, MPEG-4 video packets are placed in AC2 category which has better opportunity to access the channel than lower priority Access Categories (ACs). The tradeoff is, when the video stream increases, this queue rapidly jams and drops occur. So an adaptive, cross-layer mapping algorithm approach has been implemented and it is already discussed in [3]. This mapping algorithm re-arranges most recently received video packets into other available lower priority queues, while the AC2 queue is getting filled. Two parameters, ***threshold_low*** and ***threshold_high*** which was denoted in ***[2]*** has been used predicatively to avoid the upcoming congestion by performing queue management in advance. The integrated function to introduce these two parameters in the cross layer mapping approach is in the following expression:

$$\text{Prob\_New}=\text{Prob\_Type} * \frac{\text{qlen (AC[2])} - \text{threshold\_low}}{\text{threshold\_high} - \text{threshold\_low}}$$

In this function, the original predefined downward mapping probability of each type of video frame, ***Prob_TYPE***, will be adjusted according to the current queue length and threshold values, and about the result is a new downward mapping probability, ***Prob_New***. The higher ***Prob_New***, the greater the opportunity for the packet to be mapped into a lower priority queue.

In the cross layer mapping approach when a video packet arrives, first the queue length of AC2 is checked and it is compared with values of threshold_high and threshold_low. If queue length is lower than the threshold_low (light load), the video data is mapped to AC[2] irrespective of the type of video data being transferred. But if the queue length is greater than the threshold_high (heavy video traffic load) the video data is directly mapped to lower priority queues, AC[1] or AC[0]. However if queue length of AC[2] is between threshold_high and threshold_low, the mapping decision considers both the mapping probability (Prob_TYPE) and the current buffering size condition of the queue. Hence, the video data packet will be mapped to different AC's according to the calculated downward mapping probability. With such a priority scheme in mac layer along with UDPLite in Transport layer the transmissions are prioritized and the drop rate of video is minimized.

5 RELATED WORK

EDCA has been improved by adjusting the parameter adaptively to channel state or congestion level in [4]. adaptive EDCA had been implemented, where the access point adopted the contention window based on the network congestions was discussed in [5]. a two level protection had been applied for voice and video traffic by distributed admission control. The Budget calculation had been done in EDCA to protect existing video streams and the issue of bandwidth allocation for video streams had been investigated in [6]. The cross layer architecture was used which is based on the data partitioning and they have been associated to each partition within the access layer categories of EDCA was discussed in [7]. The macro and micro rate control schemes had been used at the application layer and network layer which uses bandwidth estimation and adaptive mapping of packets using video classifications was discussed in [8]. A wireless video system had been built using the error resilient low bit rate video coder by implementing UDPlite and PPP lite in transport and link layer protocols for cellular video was discussed in [9]. The H.264 had been transmitted for video over an adhoc scenario using Udplite which has reduced retransmission using unequal error protection was discussed in [10]. A multimedia network asic design had been implemented which includes the characteristics of H.264 with Udplite to reduce packet loss was discussed in [11]. A distributed algorithm had been implemented for channel time allocation among multiple video streams, and they had investigated several heuristic packet pruning schemes for rate adaption of high definition video streams were discussed in [12]. A distributed rate allocation scheme had been implemented with a goal of minimizing the total video distortion of all peers without excessive network utilization was discussed in [13]. This scheme relied on cross-layer information exchange between MAC and application layers. A work had been done on 802.11e, where the parameters of MAC layer had been adjusted to provide very good quality by adding UDPlite in transport layer was discussed in [14].

6 PROPOSED SYSTEM

The figure 7 depicts the main components of the system architecture for wireless media streaming. The media source which can be a real time encoder or pre compressed media file, generated media packets that are initially sent to the application layer buffer. Subsequently the UDPLite header is added by the transport layer, IP header is added by the network layer. The IP packet is sent to the 802.11e MAC layer and creates an MAC protocol data unit for wireless transmission. All packets are stored in link layer buffer. Similarly the above process is been reversed at the receiver end.

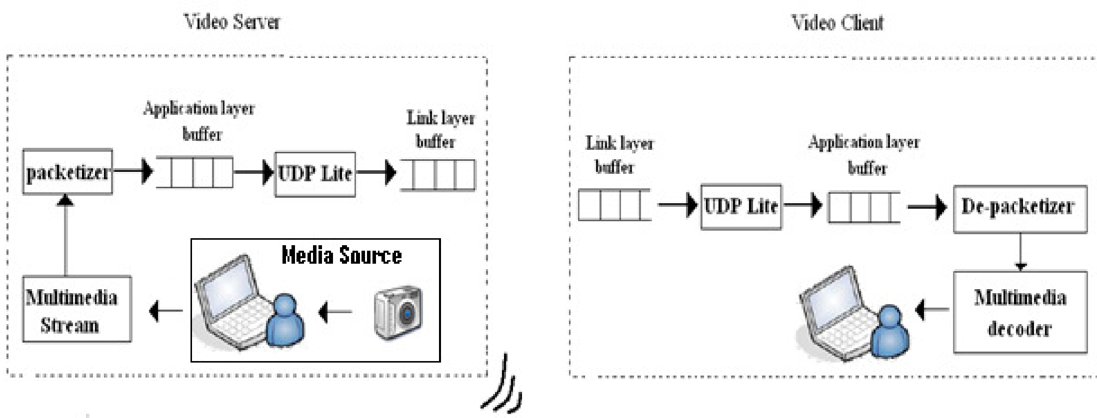

Figure 7: Multimedia Streaming System

7 SIMULATION RESULTS AND DISCUSSIONS

A.Implementation and simulation setup of Cross Layer Mapping along with UDPLite .

The simulation of Cross Layer Mapping approach along with UDPLite using NS2 simulator has been done , and the performance of the received video at the receiver has been examined. We simulate a Cross Layer Mapping approach along with UDPLite over a ad-hoc network using the NS2 simulator. By exploiting the cross layer mapping approach, prioritization is done in the transmission of essential video data that improves the queue space utilization and also support dynamic adaptation for changes in network traffic loads. MAC queue is used as the indication of the current network traffic load. When video is transmitted over an IEEE 802.11e WLAN, MPEG4 video packets are placed in AC2 which increase efficiency of accessing channels when the video stream increases in the AC2 queue rapidly filled and problems such as jams and drops occur. Inorder to overcome these problems,the proposed approach arranges most recently received video packets into other available lower priority queues. since UDPlite is used in the transport layer the packets, which are prone to errors due to radio channel variations are also delivered to the receiver by the UDPlite protocol in the transport layer,if the error has occurred in the payload. the corrupted packet will be discarded ,if the error has occurred in the header.

7.1 Simulation Topology

The simulation is performed with 3 types of video sources like YUV QCIF (176 x 144) Foreman, Claire, Akiyo. Each video frame was fragmented into packets before transmission and the maximum packet size over the simulator network is 1000 bytes. Figure 8 presents the simulation topology in the experiment. There are eight ad hoc wireless nodes where one is video server and another is video receiver. The data rate of wireless link is 1Mbps.

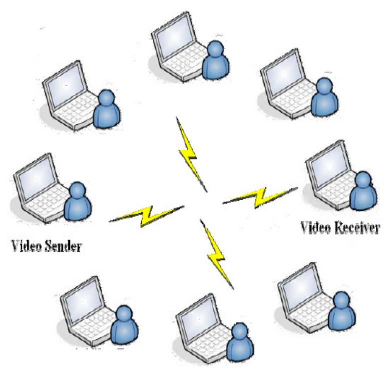

Figure 8: Network topology used in simulation.

7.2 Experiments and Results

In all simulation experiments the performance of 4 cases -MPEG4 UDP EDCA , MPEG4 UDPLite EDCA , cross layer mapping with UDP and cross layer mapping with UDPLite has been compared ,for the video sequences.

7.2.1 Delay

Figure 9 represents the delay produced by MPEG4 UDP EDCA and MPEG4 EDCA CLWP while the video source transmitted is Foreman, Figure 10 represents the Delay produced by cross layer mapping with UDP and cross layer mapping with UDPLite while transmitting Foreman of 400 frames as video source. The delay produced by cross layer mapping with UDP is 0.72 sec because no priority is given for the video packet whereas the delay is 0.6 when adaptive mapping is employed .

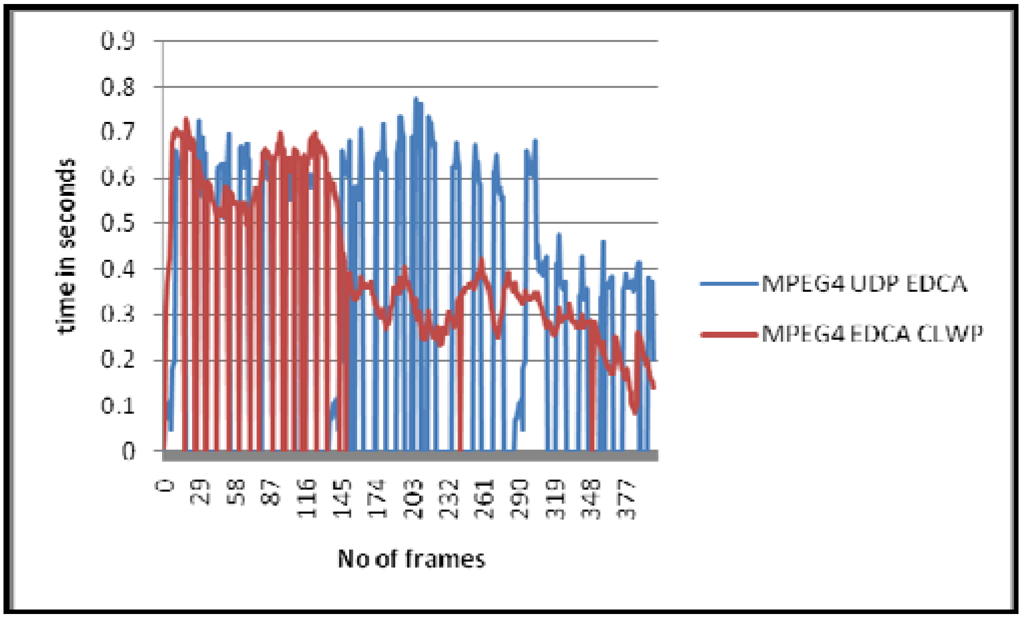

Figure 9: Delay produced by MPEG4 UDP EDCA and MPEG4 EDCA CLWP (Foreman).

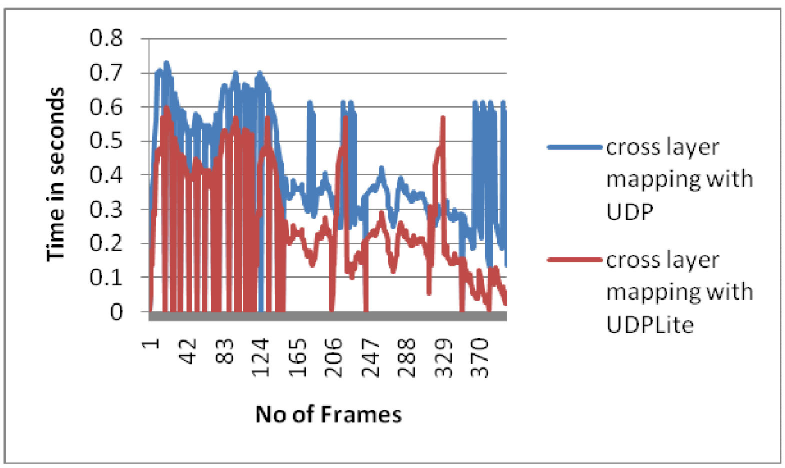

Figure 10: Delay produced by cross layer mapping with UDP and cross layer mapping with UDPLite (Foreman).

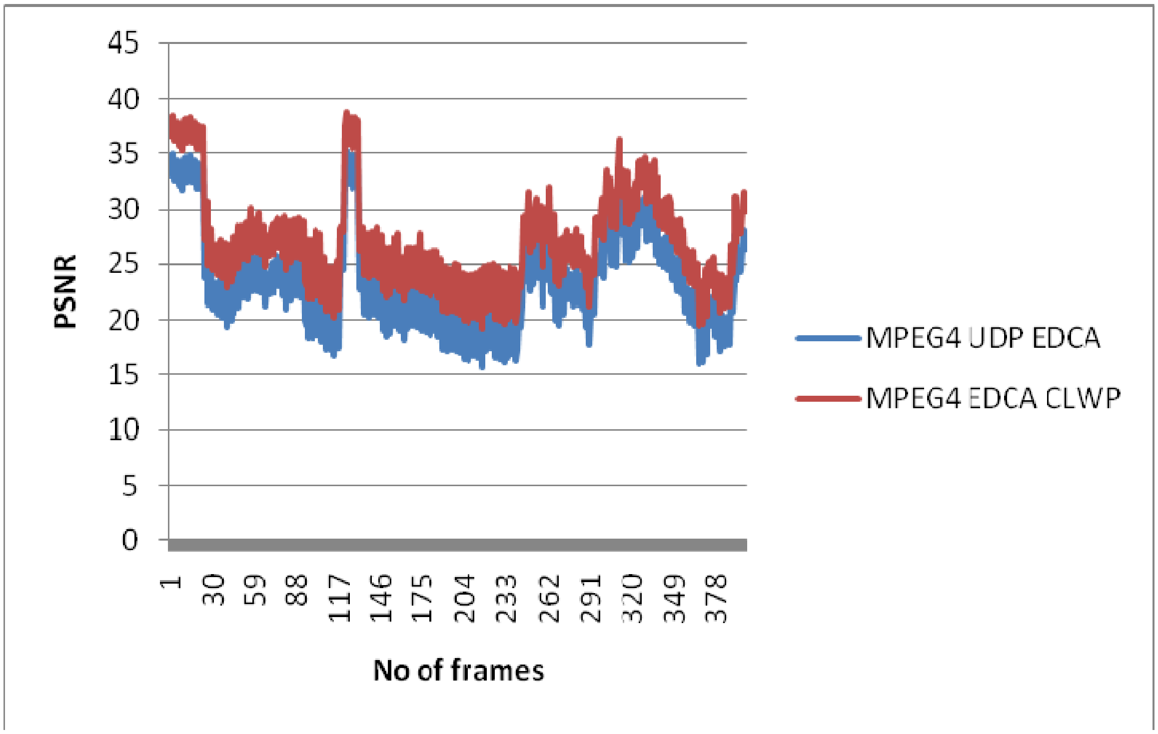

Figure 11: PSNR produced by MPEG4 UDP EDCA and MPEG4 EDCA CLWP (Foreman)

7.2.2 Peak Signal Noise Ratio

Figure 11 represents the PSNR produced by MPEG4 UDP EDCA and MPEG4 EDCA CLWP while the video source transmitted is Foreman, Figure 12 represents the PSNR produced by cross layer mapping with UDP and cross layer mapping with UDPLite while the video source transmitted is Foreman. Without mapping the PSNR is 38 and with mapping the PSNR is 40.

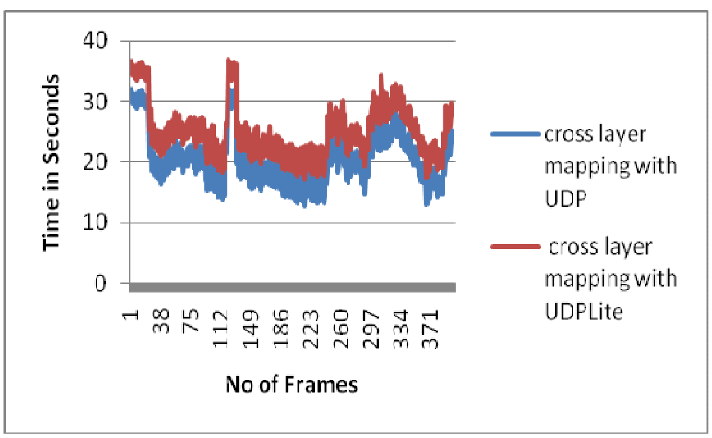

Figure 12: PSNR produced by cross layer mapping with UDP and cross layer mapping with UDPLite (Foreman).

8 CONCLUSION

Although, IEEE 802.11e EDCA has some features for QoS support it is not effective in providing priority to real time traffic such as delay sensitive video. By using dynamic mapping technique, video packets are mapped to the appropriate access category based on the significance of video data and network traffic load. Our proposed approach combines the benefits of UDPLite along with cross layer mapping thereby increasing the PSNR and decreasing the delay to a great extent.

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