# Cross Layer Optimization in Multimedia Sensor Network

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# Cross Layer Optimization in Multimedia Sensor Network

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By

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# Certificate

This is to certify that the Major Project(Part-II) entitled "Cross Layer Optimization in Multimedia Sensor Network" submitted by Sameer Kapadia (11MCEC24), towards the partial fulfillment of the requirements for the degree of Master of Technology in Computer Science and Engineering of Nirma University, Ahmedabad is the record of work carried out by him under my supervision and guidance. In my opinion, the submitted work has reached a level required for being accepted for examination. The results embodied in this major project part-II, to the best of my knowledge, haven't been submitted to any other university or institution for award of any degree or diploma.

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# Abstract

Multimedia applications over wireless sensor networks are emerging rapidly. There is an increasing interest in the research community to design and develop critical services which require video monitoring or emergency voice calls over WSNs. On the other hand, storage, processing and bandwidth limitations of sensor nodes make multimedia data transmission a challenging issue for WSNs. In such context, multimedia coding and cross-layer optimization may enhance the expected efficiency of WMSN applications by considering the multimedia traffic characteristics and dynamic tuning of the MAC parameters.

IEEE 802.11e is the indisputable standard for supporting multimedia traffic in modern Wireless Local Area Networks. However, it has been proven incapable of handling efficiently multimedia flows in congested networks. The main reason for this suboptimal behavior roots from the static nature of resource allocation specified in IEEE 802.11e. These thesis surveys what is multimedia sensor network, what are the WMSN restrictions, Functions of different layers in WMSN and cross layer design techniques to improve the throughput and performance of the network. In this thesis, a cross-layer mechanism is introduced, under the modified EDCA. The mechanism is designed to cope with high load situations in IEEE 802.11e wireless infrastructure networks by selectively prioritizing and protecting sensitive multimedia frames.

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# Chapter 1

# Introduction

## 1.1 General

A Wireless multimedia sensor network (WMSN) is composed of tiny wireless sensing devices that allow fetching video, audio and still images and reporting to a sink node. In traditional multimedia systems, the streaming server is capable of implementing complex encoding algorithms while the receiving clients just use a simple algorithm to decode the receiving multimedia data. In contrast, the multimedia sensors in WMSNs have severe resource constraints including bandwidth, energy, storage, computation, etc. Therefore, the traditionally complex encoding techniques are not applicable to WMSNs. Multimedia support in wireless networks has been widely explored. These include broadband wireless networks, satellite networks, mobile ad hoc networks. However, there are several main peculiarities that make multimedia content delivery in WMSNs challenging, which is largely unexplored. Most of these challenges are unique in WMSNs due to the limited resources imposed on sensor nodes, and transporting video with guaranteed QoS in WMSNs is of prime importance due to higher rate requirements on limited and variable capacity channels. By taking into account the resources in WMSNs, joint optimization of networking layers, i.e., cross-layer design, stands as the most promising alternative to inefficient traditional layered protocol architectures.

## 1.2 Motivation

The purpose of this project is to address the need of optimization in WMSN using cross layer technique.

## 1.3 Scope of Project

The main motive of the project is to improve the performance, packet delivery ratio and assure the quality of multimedia content in wireless sensor network using cross layer communication.

## 1.4 Thesis Organization

The Thesis on "Cross Layer Optimization in Multimedia Sensor Network" has been divided in chapters as follows:

- Chapter 2, Literature Survey, presents basic concept of WMSN, it's limitation, Problems with WMSN, existing algorithms to overcome that problems.
- Chapter 3, Approach and Method, includes the proposed approach in which a cross layer technique is used by combining the application and MAC features.
- Chapter 4, Implementation and results of proposed approach.
- Chapter 5, Conclusion and future work.

# Chapter 2

# Literature Survey and Important Observations

## 2.1 What is WMSN?

Wireless Multimedia sensor network is a network of wireless interconnected devices that consist of CMOS cameras and microphones that retrieve multimedia content such as audio, video streams, still images scalar sensor data from the environment. Multimedia sensor networks enhance our understanding of the physical world, improving applications like surveillance, disaster monitoring, wildlife observation, automated assistance for elderly and disabled people, traffic avoidance, industrial process control and localization services. In sensor networks comprised of visual sensors, video and/or images are collected from the environment in a different way from traditional WSNs. Most cameras installed at source nodes are not omnidirectional, resulting in a directional sensing capability usually referred as the cameras Field of View (FoV). In fact, the FoV is a sector-like visible region emanating from the camera, defining a direction of viewing (the cameras pose). Viewing angle, lens quality and zoom capabilities, as well as the type of the camera used in source nodes are major factors that influence the resulting FoV and the way visual data are retrieved from the environment.[1]

## 2.2 WMSN Restrictions

The visual data sensed by source nodes have to be digitalized and transmitted to the sink over the sensor network. The energy and processing constraints of the sensor nodes, as well as the nature of the wireless links that interconnect them, restrict the attainable bandwidth of the communication path(s) and impose a considerable packet loss rate. Among the adopted solutions, multimedia coding techniques are used to compress the original data, reducing the required transmission rate and potentially saving energy of the source node and over the entire path(s) toward the sink. Additionally, some multimedia coding provides error resilience, which may sustain the minimum acceptable end-to-end quality of the application, even when some packets are lost while transmitted over error-prone wireless links.

## 2.3 Functions of Layers[2]

### 2.3.1 Application

The functionalities handled at the application layer of a WMSN are characterized by high heterogeneity, and encompass traditional communication problems as well as more general system challenges. The services offered by the application layer include: (i) providing traffic management and admission control functionalities, i.e., prevent applications from establishing data flows when the network resources needed are not available; (ii) performing source coding according to application requirements and hardware constraints, by leveraging advanced multimedia encoding techniques; (iii) providing flexible and efficient system software, i.e., operating systems and middleware, to export services for higher-layer applications to build upon; (iv) providing primitives for applications to leverage collaborative, advanced in-network multimedia processing techniques.

### 2.3.2 Transport

In applications involving high-rate data, the transport layer assumes special importance by providing end-to-end reliability and congestion control mechanisms. Particularly, in WMSNs, the following additional considerations are in order to accommodate both the unique characteristics of the WSN paradigm and multimedia transport requirements.

- Effects of congestion: In WMSNs, the effect of congestion may be even more pronounced as compared to traditional networks. When a bottleneck sensor is swamped with packets coming from several high-rate multimedia streams, apart from temporary disruption of the application, it may cause rapid depletion of the nodes energy. While applications running on traditional wireless networks may only experience performance degradation, the energy loss (due to collisions and retransmissions) can result in network partition. Thus, congestion control algorithms may need to be tuned for immediate response and yet avoid oscillations of data rate along the affected path.

- Packet re-ordering due to multi-path: Multiple paths may exist between a given source-sink pair, and the order of packet delivery is strongly influenced by the characteristics of the route chosen. As an additional challenge, in real-time video/ audio feeds or streaming media, information that cannot be used in the proper sequence becomes redundant, thus stressing on the need for transport layer packet reordering.

- Reliability: Multimedia streams may consist of images, video and audio data, each of which merits a different metric for reliability. When an image or video is sent with differentially coded packets, the arrival of the packets with the ROI field or the I-frame respectively should be guaranteed. The application can, however, withstand moderate loss for the other packets containing differential information. Thus, we believe that reliability needs to be enforced on a per-packet basis to best utilize the existing networking resources. If a prior recorded video is being sent to the sink, all the I-frames could be separated and the transport protocol should ensure that each of these reach the sink.

### 2.3.3 Network

The network layer addresses the challenging task of providing variable QoS guarantees depending on whether the stream carries time-independent data like configuration or initialization parameters, time-critical low rate data like presence or absence of the sensed phenomenon, high bandwidth video/ audio data, etc.

- Addressing and localization: In the case of large WMSNs it is required that the individual nodes be monitored via the Internet. Such an integration between a randomly deployed sensor network and the established wired network becomes a difficult research challenge. The key problem of global addressing could be solved by the use of IPv6 in which the sensor can concatenate its cluster ID with its own MAC address to create the full IPv6 address. However, the 16-byte address field of IPv6 introduces excessive overhead in each sensor data packet. Location information is a key characteristic of any sensor network system. The ability to associate localization information to the raw data sampled from the environment increases the capability of the system and the meaningfulness of the information extracted.

- Routing: Data collected by the sensor nodes needs to be sent to the sink, where useful information can be extracted from it. The concerns of routing in general differ significantly from the specialized service requirements of multimedia streaming applications. As an example, multiple routes may be necessary to satisfy the desired data rate at the destination node. Also, different paths exhibiting varying channel conditions may be preferred depending on the type of traffic and its resilience to packet loss.

### 2.3.4 MAC Layer

Owing to the energy constraints of the small, battery- powered sensor nodes, it is desirable that the medium access control (MAC) protocol enable reliable, error-free data transfer with minimum retransmissions while supporting application-specific QoS requirements. Multimedia traffic, namely audio, video, and still images can be classified as separate service classes and subjected to different policies of buffering, scheduling and transmission.

- Channel access policies: The main causes of energy loss in sensor networks are attributed to packet collisions and subsequent retransmissions, overhearing packets destined for other nodes, and idle listening, a state in which the transceiver circuits remain active even in the absence of data transfer. Thus, regulating access to the channel assumes primary importance.

- Scheduling: MAC layer scheduling in the context of WMSNs differs from the traditional networking model in the sense that apart from choosing the queueing discipline that accounts for latency bounds, rate/power control and consideration of high channel error conditions needs to be incorporated.

## 2.4 Cross Layer Design

The stringent requirements of visual sensor networks have led to the design of cross-layer architectures. In order to attain high efficiency, reducing energy consumption and achieving lower communication delay, the protocols and algorithms of the MAC, network, transport and applications layers can operate in a cooperative way that disrupts the conventional data flow and the understanding of protocol layers. In such context, visual codecs play a crucial role prioritizing packets, splitting the source media in multiple streams and generating redundant information that are exploited by protocols and algorithms of different conceptual layers.

An interesting example of multimedia-based cross-layer design is just congestion mitigation by transport layer. If the classical layer organization is respected, a typical transport protocol will reduce the current transmission rate to resolve congestion issues, potentially impacting the quality of the received media by the sink (loss of video frames, higher delay) However, a cross-layer design may benefit from a multimedia coding technique that prioritizes the transmitted packets. In such case, the transport protocol would only reduce the transmission rate of the less-relevant packets, satisfactorily addressing congestion and potentially resulting in better end-to-end quality of the application.

## 2.5 Multimedia Coding Fundamentals

The algorithm for multimedia coding is referred as codecs (enCOder/DECoder). These algorithms aim to reduce the required transmission bandwidth of digitalized multimedia data and to allow some kind of recovery of information lost due to packet dropping. The maximum quality of the received multimedia data is obtained when the sensed data is transmitted with lossless data compression techniques and with enough redundancy to compensate packet loss. However, the resulting transmitted data will require high bandwidth, what could be prohibitive for many links. In order to achieve lower transmission rates, codecs reduce the total amount of the original data using some compression technique, demanding computational time and processing resources, but efficient data compression usually inflict some information loss. In general, information loss grows with increasing compression rate. The multimedia coding techniques in WMSN applications should combine high compression efficiency, low complexity and error resilience, saving energy in source node and through the entire path(s) toward the sink.

## 2.6 MPEG Compression[3]

The I-frames are intra coded, i.e. they can be reconstructed without any reference to other frames. The P-frames are forward predicted from the last I-frame or P-frame, i.e. it is impossible to reconstruct them without the data of another frame (I or P). The B-frames are both, forward predicted and backward predicted from the last/next I-frame or P-frame, i.e. there are two other frames necessary to reconstruct them. P-frames and B-frames are referred to as inter coded frames.

### 2.6.1 The discrete cosine transform(DCT)

$F(u,v) = C_u/2C_v/2 \sum_{y=0}^{7} \sum_{x=0}^{7} f(x,y)cos[(2x+1)u\pi/16])cos[(2y+1)v\pi/16])$

with:

$C_u = \begin{Bmatrix} 1/\sqrt{2} & if u = 0 \\ 1 & u > 0 \end{Bmatrix}; C_v = \begin{Bmatrix} 1/\sqrt{2} & if v = 0 \\ 1 & v > 0 \end{Bmatrix}$

Where f(x,y) is the brightness of the pixel at position [x,y].

### 2.6.2 Inverse Discrete cosine transform(IDCT)

$f(x,y) = \sum_{u=0}^{7} \sum_{v=0}^{7} F(u,v)C_u/2C_v/2cos[(2x+1)u\pi/16])cos[(2y+1)v\pi/16])$

Where F(u,v) is the transform matrix value at position [u,v].

The transformed values are quantized. That means they are (integer) divided by a certain value greater or equal 8 because the DCT supplies values up to 2047. To reduce them under the byte length at least the quantization value 8 is applied. The decoder multiplies the result by the same value.

## 2.7 802.11e

As a response to the market trend, IEEE has released a series of amendments improving the functionality of the initial WLAN standard. Most of these amendments have introduced new physical layers (PHYs) that incorporate modulation techniques in order to increase available data rates (IEEE 802.11a/b/g/n). However, simply augmenting data transmission rates has been proven an insufficient approach for multimedia applications support due to the increased Medium Access Control (MAC) overhead imposed by the basic channel access mechanism (Distributed Coordination Function DCF) along with the lack of traffic prioritization. To this direction, IEEE introduced a new QoS-aware

MAC layer protocol capable of service differentiation, accompanied by a series of MAC layer enhancements, under the name IEEE 802.11e.

As demonstrated by several research studies, EDCA still lacks adequate multimedia support in high load conditions in wireless infrastructure networks. There have been more than a few research proposals to mitigate this problem which focus mainly on dynamically assigning network resources to multimedia flows. Even though, altering the static network resource allocation proposed by IEEE 802.11e has been proven beneficial in terms of QoS provisioning, most of the existing mechanisms follow a layered approach without explicitly considering the specific characteristics of multimedia applications. The outcome is a simple independent implementation which leads in suboptimal multimedia performance. Thus, it is marginally more beneficial to approach the multimedia QoS provisioning problem by following a cross-layer design that combines the multimedia traffic characteristics along with lower layer strategies at the OSI stack.

## 2.8 802.11e EDCA

By default, the IEEE 802.11 legacy MAC is incapable of providing service differentiation due to the fact that its basic access method provides equal access probabilities to all stations (STAs) and the existence of a single MAC queue within each node. The EDCA access mechanism defined by IEEE 802.11e is a modified DCF scheme designed to provide differentiated and distributed channel access.

The service differentiation is realized by introducing four User Priorities (UPs), from 0 to 3, with 3 having the highest priority. Each frame from the higher layer arrives at MAC layer with a specific UP which is marked, afterwards, to its MAC header. An 802.11e STA (called QSTA), shall implement four Access Categories (ACs), from 0 to 3, with 3 having the highest priority. The four access categories are: AC(3) for voice traffic, AC(2)for video traffic, AC(1) for best effort traffic and AC(0) for background traffic. Each access category acts as an independent back-off entity, and the priority among access categories is given by access category specific parameters. The differentiation between these access categories is achieved by sending different contention window minimum value, contention maximum value and Arbitration Interframe Space (AIFS). Hence a QSTA has four MAC queues, where each queue corresponds to an AC. Each AC is an enhanced variant of DCF (called EDCAF) and each frame is mapped to an AC according to its UP value as shown in Table 2.1.

| **Priority** | **Access Category (AC)** | **Designation (Informative)** |
|---|---|---|
| 0 | 0 | Background traffic |
| 1 | 1 | Best effort |
| 2 | 2 | Video |
| 3 | 3 | Voice |

Table 2.1: Priority to access category mappings

MSDUs (MAC service data unit) are now delivered through multiple backoff instances within one station, each backoff instance parameterized with TC-specific parameters. In the CP, each TC within the stations contends for a TXOP and independently starts a backoff after detecting the channel being idle for an Arbitration Interframe Space (AIFS); the AIFS is at least DIFS, and can be enlarged individually for each TC. After waiting for AIFS, each backoff sets a counter to a random number drawn from the interval [1,CW+1]. The minimum size (CWmin[TC]) of the CW is another parameter dependent on the TC. Priority over legacy stations is provided by setting CWmin[TC]¡15 (in case of 802.11a PHY) and AIFS=DIFS. See Fig. 2.1 for illustration of the EDCF parameters.

The key feature of EDCA is that for each AC a different set of MAC parameters are assigned in order to achieve prioritized medium access.

Another feature introduced by IEEE 802.11e is the concept of Transmission Opportunity (TXOP). This is defined as the interval of time in which a QSTA, after winning contention, has the right to initiate multiple frame transmissions separated by a SIFS (Short Inter-Frame Space) idle period, as long as the total transmission time does not exceed a limit called TXOP limit. This procedure is called Contention Free Burst (CFB) and is optional for a QSTA to utilize it. There is a set of default values specified by the IEEE 802.11e standard for the TXOP limit under the EDCA access mechanism. These values depend on the AC type and on the underlying physical layer. Multimedia ACs

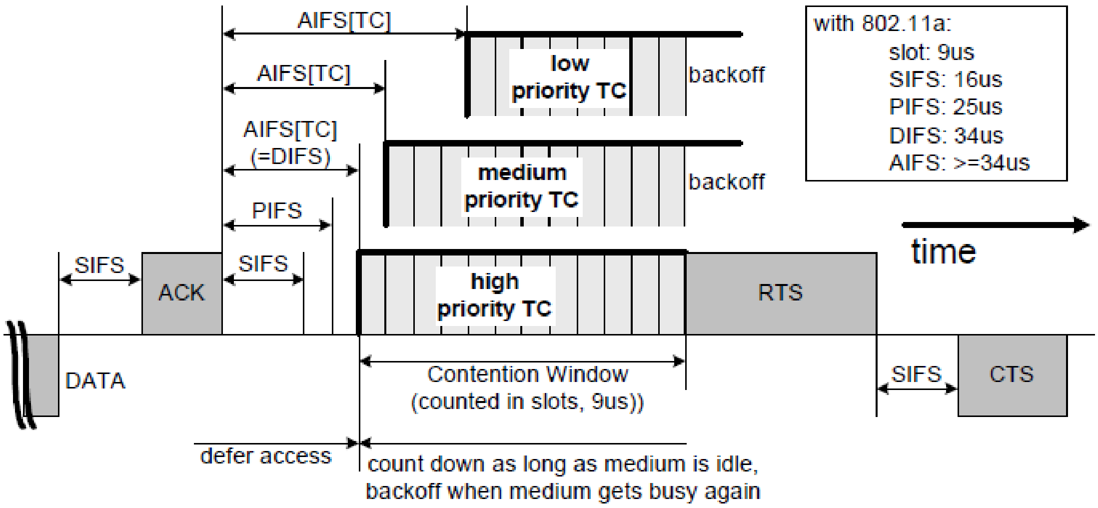


Figure 2.1: Multiple backoff of MSDU streams with different priorities.

(AC(3) and AC(2) are enabled with the CFB feature while background and best effort traffic (AC(1) and AC(0) are allowed to transmit single data frames before they re-enter the contention phase.

By allowing multiple frame transmissions after winning a contention, CFB reduces the number of backoff and idle periods as well as the number of RTS/CTS frames exchanged, thus resulting in lower overhead. A CFB transmission chronicle is depicted in Fig. 2.2. It is a straightforward conclusion that assigning large values to TXOP limit will allow higher throughput and lower delays to a specific AC. The default TXOP limit values proposed by IEEE 802.11e are, however, statically assigned. Nonetheless, the standard permits dynamic allocation of TXOP limit values. Hence, careful dynamic allocation of transmission resources enables multimedia performance improvement in terms of throughput and delay.

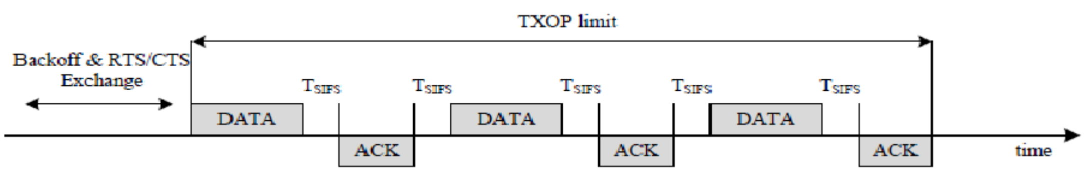


Figure 2.2: CFB timing structure

The following Table 2.2 shows the EDCA-parameters selected for the three priorities, summarizing the EDCA parameters we mainly use.

| Parameters | AC(0) | AC(1) | AC(2) | AC(3) |
|---|---|---|---|---|
| AIFS | 7 | 3 | 2 | 2 |
| CW_MIN | 31 | 31 | 15 | 7 |
| CW_MAX | 1023 | 1023 | 31 | 15 |
| TXOPLIMIT | 0 | 0 | 0.006016 | 0.003264 |

Table 2.2: Parameters of EDCA for different access categories

$$\textbf{AIFS} = \textbf{SIFS} + \textbf{AIFS[AC]} * \textbf{Slot time}$$

Towards reducing the MAC layer overhead even further, IEEE 802.11e introduced two new acknowledgment policies besides the default DATA-ACK handshake. As depicted in Fig.2.2 each frame is required to be individually acknowledged before the QSTA may proceed with the next frame in its sequence. Hence, the transmission of the subsequent data frame commences not before the passing of 2 * TSIFS + TACK period of time. Accumulating all these waiting periods, the final amount of time dedicated to the exchange of control frames (ACK) may occupy a significant portion of the available TXOP limit assigned to the QSTA.

In order for a QSTA to fully utilize the available TXOP limit, the Block Acknowledgment (BlockACK) and the No Acknowledgment (NoACK) policies are defined under the IEEE 802.11e standard. In short, the BlockACK policy accumulates acknowledgment indicators into a single frame at the end of the frame burst, while the NoACK scheme suppresses acknowledgments completely, imitating a UDPlike behavior. These acknowledgment policies combined with the CFB feature may drastically improve channel utilization and MAC efficiency.

## 2.9 Multimedia Characteristics And Evaluation

### 2.9.1 Streaming Video

Digital video consists of frames (images) that are displayed at a prescribed frame rate. For transport over networks, video is typically encoded to reduce bandwidth requirements. We focus on the encoding characteristics of MPEG since most commercial products are derived from this standard.

One of the main principles of MPEG is inter-frame coding. This method introduced three frame types: intracoded (I frame), inter-coded (P frame) and bidirectional coded (B frame). These different frame types are organized into so-called Group of Pictures (GoP) and the pattern of I, P and B frames that make up a GoP is called GoP pattern. The sequence of frames from a given I-frame up to and including the frame preceding the next I-frame forms one GoP. A GoP pattern is determined by the total number of frames, N, comprising it and the number of B-frames, M, enclosed by successive P-frames. Thus the notation GNBM is used to symbolize the GoP pattern of a video sequence. Typical GoP patterns include: G6B2, G9B2, G12B2 and G15B2, depending on the required video quality.

I-frames contain large portions of image information and exhibit the lowest compression ratios. They are selfdecodable, meaning that they can be decoded without the need for any other frames. P-frames are inter-coded with reference to the preceding I or P-frame using motion estimation and compensation. By exploiting temporal redundancies, P-frames achieve higher compression ratios than I-frames. B-frames are also inter-coded but need either I or P preceding frame and succeeding I or P-frame as a reference. Since they are predicted from both previous and following frames they hold the highest compression ratios. Having the highest frame sizes, I-frames are considered as the largest delay consuming factor in a video sequence transmission. At the same time they are identified as the most significant frame type, since their absence will render a GoP completely undecodable.

The importance of P-frames, roots from their relative position in the GoP. For demonstration purposes, let us consider two cases of P-frame loss at the same GoP with GoP pattern G9B2, as shown in Fig. 2.3

Absence of a P-frame in the GoP will have a chain effect on all frame references rooting from that particular frame. The case of a loss occurrence on the first P-frame will make 88% of the GoP undecodable at the receiver Fig.2.3(a). Similarly, the case pre-

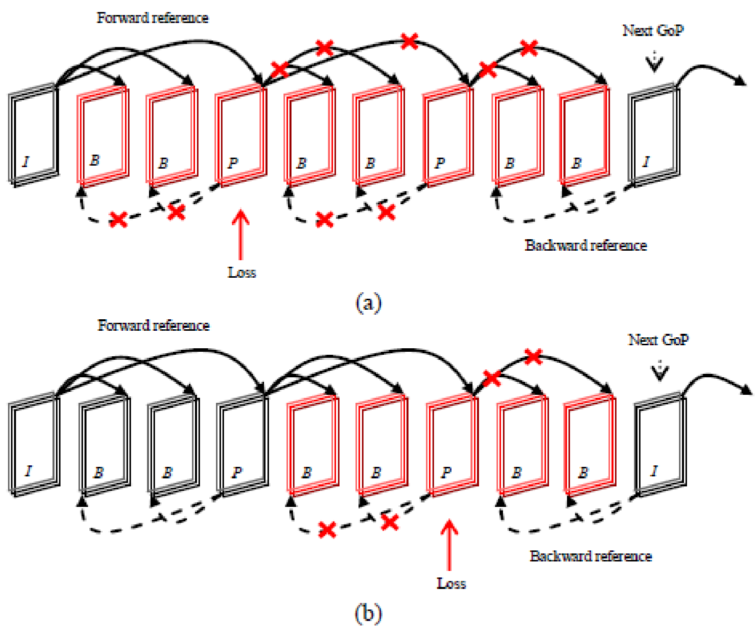


Figure 2.3: Cases of P-frame loss: (a) first P-frame and (b) second P-frame.

sented in Fig. 2.3(b) will produce a total loss of 55% of the GoPs length. Furthermore, in the case of scalable video encoding, where both I and P-frames are needed to provide a basic quality video, the importance of Pframes is strengthened even further. B-frames that are not reference frames may be considered as the lowest significance frames. This particularity of video streaming does not apply to VoIP applications due to the lack of the frame reference scheme in voice packets.

The standard method for assessing the perceived video quality is to calculate the Peak Signal to Noise Ratio (PSNR) between the original (transmitted) and the received (possibly distorted) image. It is a differential metric which is determined image-wise and yields a quality indicator for each received image of the video sequence. Typical values of PSNR for video compression lies between 30 to 50 dB with higher values preferred over lower ones.

## 2.10 Related Work

The related work is divided into two categories: MAClayer mechanisms and cross-layer mechanisms. In the first category an analysis of existing TXOP limit adaptation algorithms is provided while the second is focused on crosslayer mechanisms presented in the literature the past few years.

### 2.10.1 MAC-layer mechanisms

In [6], Majkowski and Palacio developed a dynamically TXOP limit adaptation scheme based on the average number of packets allocated in QAP queues in an attempt to alleviate the downlink/uplink problem. The new TXOP limit value of each AC is computed at the beginning of each beacon interval (i.e., 100 ms).

Similarly to [6], Andreadis and Zambon in [4] propose a dynamic TXOP limit allocation mechanism at the QAP which achieves fairness on the channel allocation times between the QAP and the other stations. Regarding VoIP streams (which are intrinsically symmetric), this scheme assigns half of channel resources at the QAPs AC(3) to balance the transmission of VoIP frames between upstream and downstream directions. Regarding video flows, the TXOP limit values are computed by accounting the number of lost packets in the uplink and downlink direction during an observing interval (set equal to the beacon interval).

In [11], the authors provide a distributed enhanced TXOP (ETXOP) limit adjustment mechanism that computes new values each time a multimedia AC's AC(3) and AC(2) wins the contention. This mechanism is based on ACs priority and ACs flow data rate. As in [6], the values for TXOP limit are computed based on the number of packets present in the ACs MAC queue.

All previously presented mechanisms do not take into account the time varying characteristics of certain multimedia traffic, such as video streaming, which exhibits high variance in frame sizes. It is worth noticing that in the simulation experiments of all the proposed algorithms the traffic considered for video streaming applications had constant bit rate characteristics.

### 2.10.2 Cross-layer mechanisms

In [12], Ksentini et al propose a QoS cross-layer architecture based on both application and MAC layer features for improving H.264 video transmission over IEEE 802.11e networks. The mechanism relies on a data partitioning technique at the application layer and an appropriate QoS mapping at the IEEE 802.11e MAC layer. The application layer video generated slices are mapped to appropriate ACs at the MAC layer according to their significance. AC(3), AC(2) and AC(1) are used for this purpose while AC(0) is left for serving all other traffic. Furthermore, the retry count parameter at the MAC layer is exploited to unequally protect the high priority information against lower significance frames.

A similar mechanism is presented in [13] for MPEG-4 video transmission in IEEE 802.11e networks. This scheme introduced a single-video multilevel queue by assigning Iframes to AC(3), P-frames to AC(2), B-frames to AC(1) and non-video frames to AC(0).

In [14], Goel and Sarkar propose a mechanism that resides in the interface between LLC and MAC layers to provide QoS for streaming video traffic. The essence of this scheme is to mark I-frames of a video sequence as the Most Valuable Video Packet (MVVP) and en-queue these frames to a higher priority queue called Video Friendly Queue (VFQ) in the interface between LLC and MAC layers. Other frames are en-queued in the so-called Interface Queue (IFQ) and receive FIFO treatment. Whenever frames need to be send to the MAC layer the VFQ receives priority against IFQ.

In[4], the MPEG-4 video packets are dynamically maps to 802.11e appropriate access categories. In 802.11e mesh networks, packets are differentiated and higher priorities are given to forward packets. When queue length of AC(2) fills up, forward packets are remapped to lower access category AC(1).

The above mentioned mechanisms assign them equal priority with non-multimedia traffic. Furthermore, none of these schemes make use of existing MAC layer strategies (such as acknowledgment policies or TXOP limit adaptation techniques) that may be extremely beneficial on multimedia performance.

# Chapter 3

# Proposed Approach

## 3.1 Proposed Scheme

In this work, a cross-layer architecture is provided, that positively affects multimedia traffic performance in infrastructure IEEE 802.11e networks under heavy load conditions by considering application level information and combining them with MAC level strategies. The proposed mechanism is distributed, and its effectiveness is proven by means of simulation.

Following the work proposed in [3], and considering the congestion conditions that are frequently present in infrastructure wireless networks, I have propose a cross layer mechanism, under the name Enhanced-EDCA, that positively affects the QoS characteristics of video traffic. Enhanced-EDCA is an integrated QoS provisioning mechanism that combines application and MAC layer features to achieve its purpose.

Enhanced-EDCA is composed by two modules, namely mapping module and second strategy selection module. Fig. 3.1 shows the overall Enhanced-EDCA architecture.

### 3.1.1 Mapping Algorithm

The mapping algorithm maps I and P frames to AC(3) as I and P video frames are considered as the most important video frames, they must be transmitted with the highest possible priority. B frames are mapped to AC(2) as they have less significance in a video sequence. CBR traffic is mapped to AC(1). The flow diagram of the mapping module is depicted in Fig. 3.2.

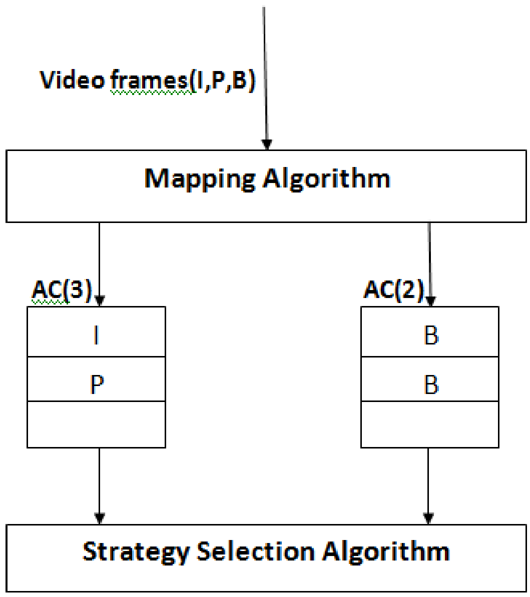


Figure 3.1: Enhanced-EDCA architecture

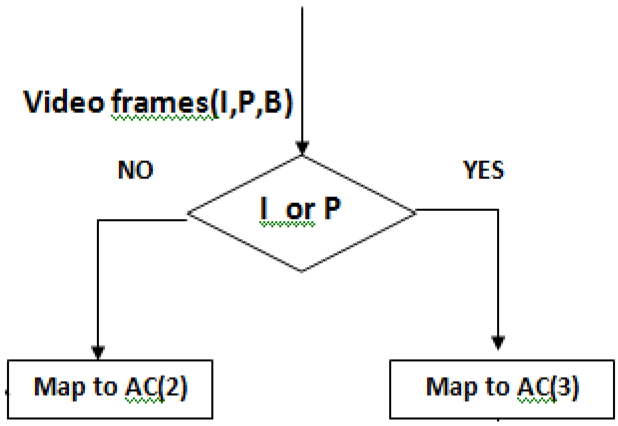


Figure 3.2: Mapping Algorithm flow Diagram

### 3.1.2 Strategy Selection Algorithm

The MAC-layer Strategy Selection Algorithm selects the appropriate strategy for each of the multimedia AC (AC(3) and AC(2)) after contention is won by that AC. since the AC(3) queue contains the most significant frames with the largest sizes so it will suffer from possible buffer overflows in congestion periods, in this case SSA use the TXOP(Transmission Opportunity) limit adaptation algorithm for that AC. Furthermore, the standard acknowledgment policy is selected for all the frames in this AC in order to protect each and every frame from possible losses.

Now to over come from congestion queue length is checked against one parameter i.e Thresholvalue of queue and it is taken as 60% of the initial queue length. When queue length of AC(3) is above thresholdvalue that means network traffic is high and in this case two mechanisms will be applied, first the incoming P frames are directly mapped to AC(2) and second the traffic is delayed by 0.2 unit of time. However when queue length reaches to it's stable condition i.e less than or equal to Thresholdvalue, P frames are again remapped to AC(3) and traffic will come to it's previous condition. Similarly when queue length of AC(2) is above thresholdvalue, three mechanisms will be applied, first the incoming B frames are mapped to AC(1), second TXOP LIMIT will be increased by 1 for AC(2) and it will de-queue fast because AC(2) will get full in short time as P frames are more than I frames, and third the traffic is delayed by 0.2 unit of time. However when queue length reaches to it's stable condition i.e less than or equal to Thresholdvalue, B frames are again remapped to AC(2), TXOP LMIT will be same as before congestion and traffic will come to it's previous condition.

The new TXOP limit value for AC's are calculated every Service Interval (SI) which is defined as the time between the start of two subsequent TXOPs. At the beginning of the SI the actual queue length is calculated and the frame size of all packets contained is determined. Then the TXOP limit is computed as the time needed to successfully transmit the en-queued frames:

In existing 802.11e EDCA the TXOP limit parameter is static and different for all AC's but in Enhanced-EDCA, I have tuned the TXOP limit parameter according to the queue length of AC's.

for ( Queuelength[AC] )
TXOPLIMIT[AC]= TXOPLIMIT[AC] $+ T_{SIFS} \quad + T_{DATA} \quad + T_{SIFS} \quad + T_{ACK}$

The terms $T_{DATA}$ and $T_{ACK}$ are the transmission times of the data and acknowledgment frames respectively for a specific PHY data rate and including PHY and MAC overhead. Fig. 3.3 depicts the flow diagram of the strategy selection module and Fig. 3.4 depicts the proposed approach model.

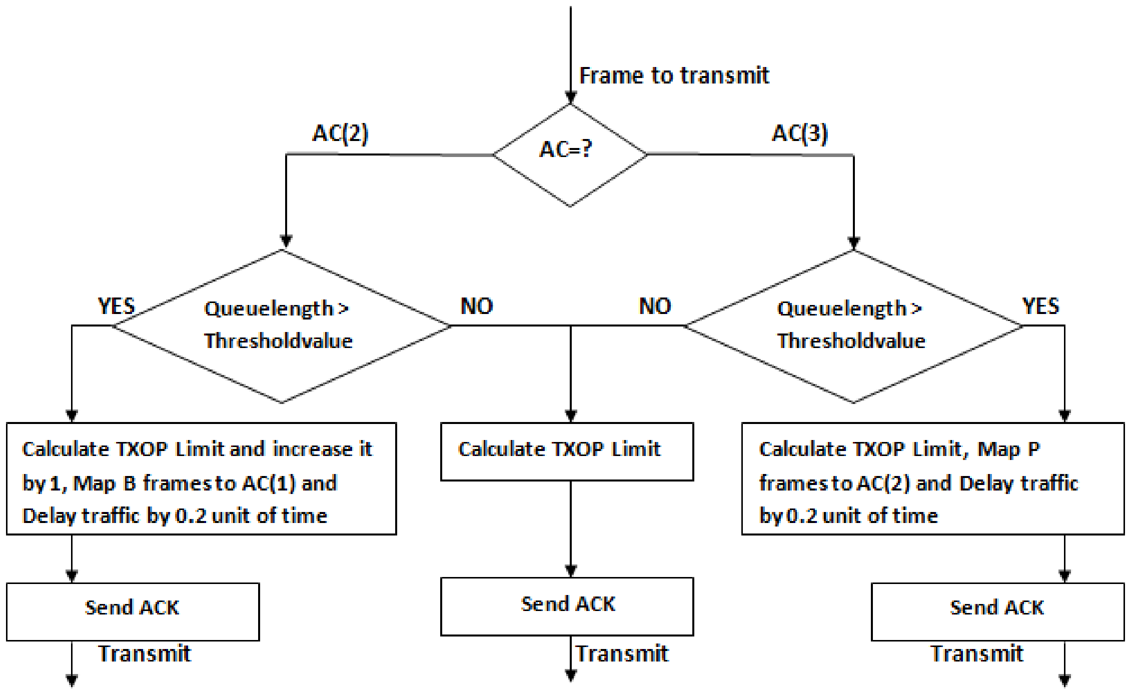


Figure 3.3: Strategy Selection flow Diagram

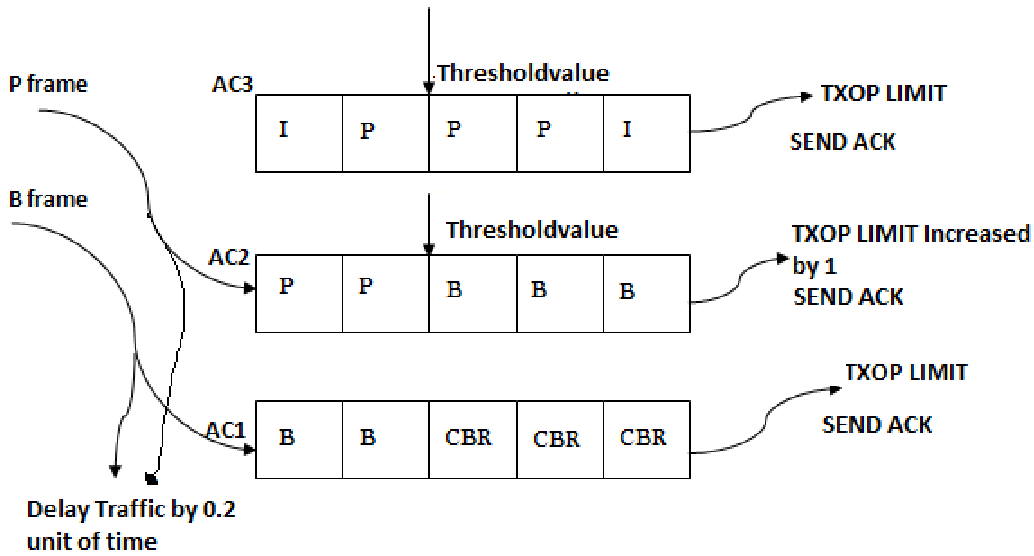


Figure 3.4: Proposed Approach model

# Chapter 4

# Simulations And Results

## 4.1 Implementation

- Fedora 8 Linux environment is used which supports network simulator NS-2.
- Network simulator NS-2, version ns-2.26 is used to simulate the proposed approach.
- Nrlsensor patch is used to make support for sensor nodes in ns-2.
- ffmpeg module is used to convert yuv to mk4 format.
- MP4BOX module is used to convert mk4 to mp4 format.
- mp4trace module is used to convert mp4 file to a text file i.e vedio traces consist of I, P, B frames.
- Evalvid framework is used to support multimedia transmission in NS-2.It is a multimedia traffic generator and responsible for the transmission of vedio traces over sensor network in ns-2.
- 802.11e module is used for the simulation of proposed approach and cross layer mechanism.
- AWK files are created to measure the performance of multimedia traffic over sensor network.

## 4.2 Simulation Setup

- Number of nodes is not static, it varies by user command line argument.

- Number of traffic sources is not static, it varies by user command line argument.
- Type of traffic - video + cbr
- Initial energy of all nodes is 20 joules.
- There is only one sink node and it can't be traffic source.
- Queue length for all nodes is taken as 50 packets per node.
- Finally, the wireless channel was assumed to be error-free, hence no packets were lost due to fading effects. In the wireless network all nodes implemented the IEEE 802.11e Enhanced- EDCA and transmitted frames at a PHY rate of 11 Mbps. Fig. 4.1 depicts the simulation scenario.

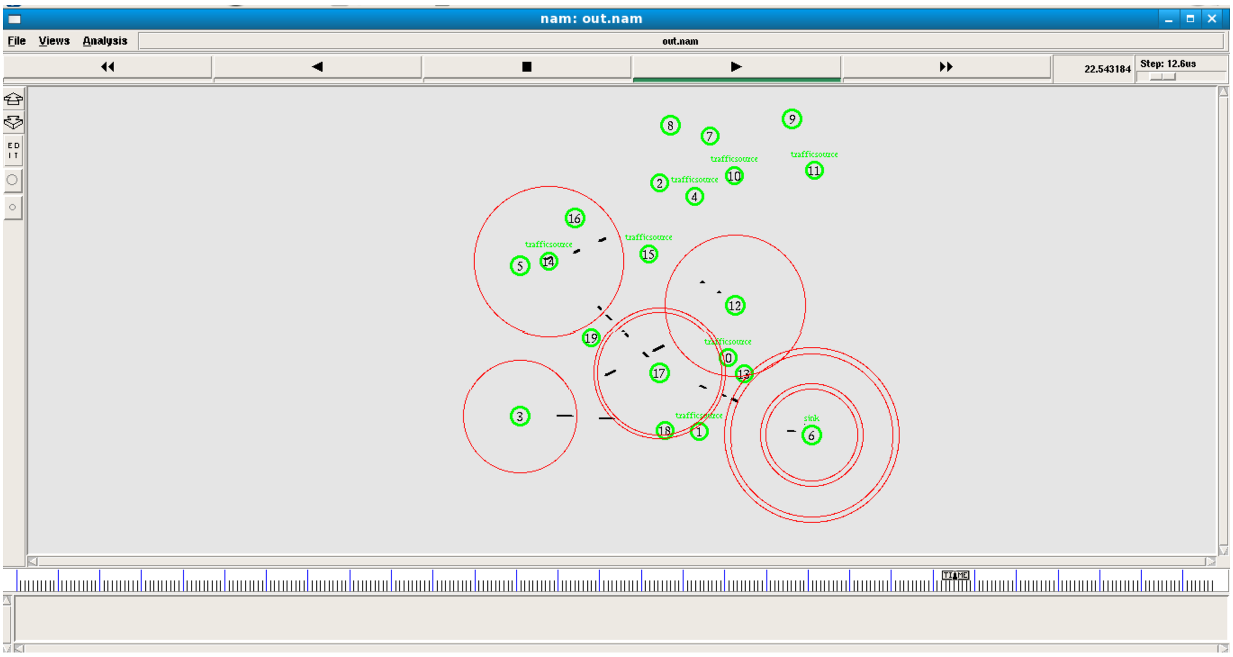


Figure 4.1: Simulation scenario

## 4.3 Result Analysis

Two simulations scenarios were considered: the first scenario applied standard EDCA functionality, the second one implemented the modified version of EDCA according to the proposed Enhanced-EDCA mechanism. The metrics used for evaluation and comparison purposes, were the PDR(Packet delivery ratio), End-to-end delay, Standard deviation and PSNR(Peak signal to noise ratio) quality metrics for video streaming applications and CBR traffic, respectively. Fig. 4.2 to Fig. 4.7 depicts the results obtained from simulations for the average PDR(video), Reconstructed video, average PDR(video+CBR), average PSNR, average End-to-end delay and standard deviation.

### PDR Results(Video)

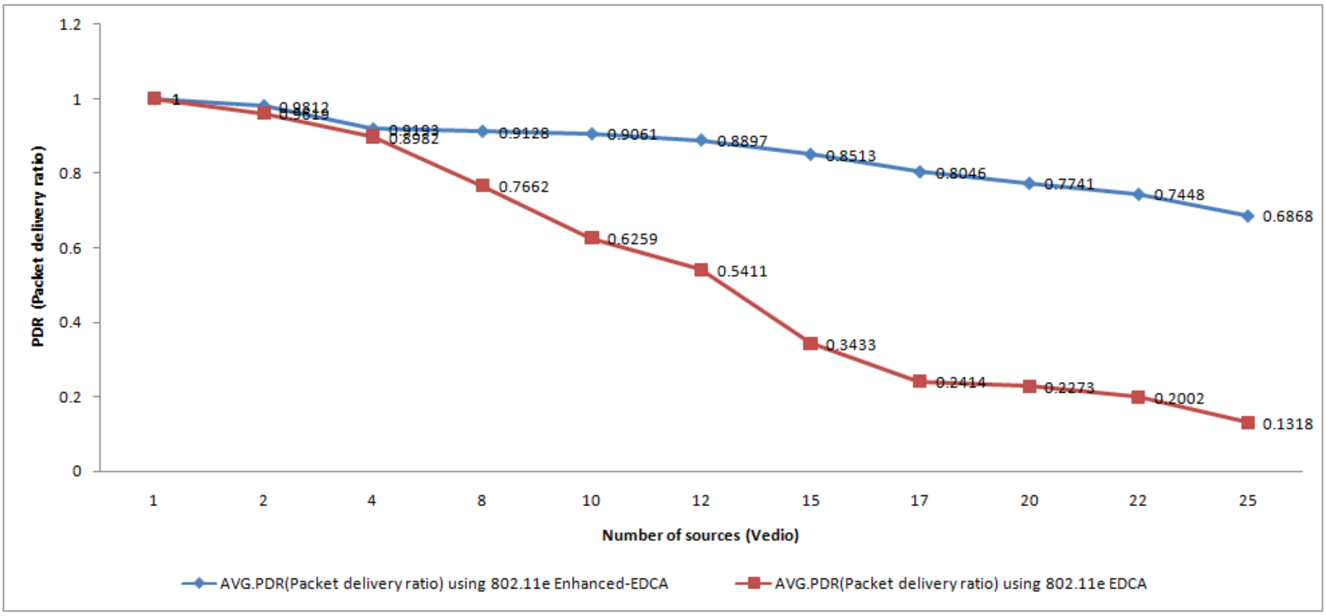


Figure 4.2: AVG PDR(Packet delivery ratio) with 51 nodes

Figure 4.3: Reconstructed Video

**PDR Results(Video+CBR)**

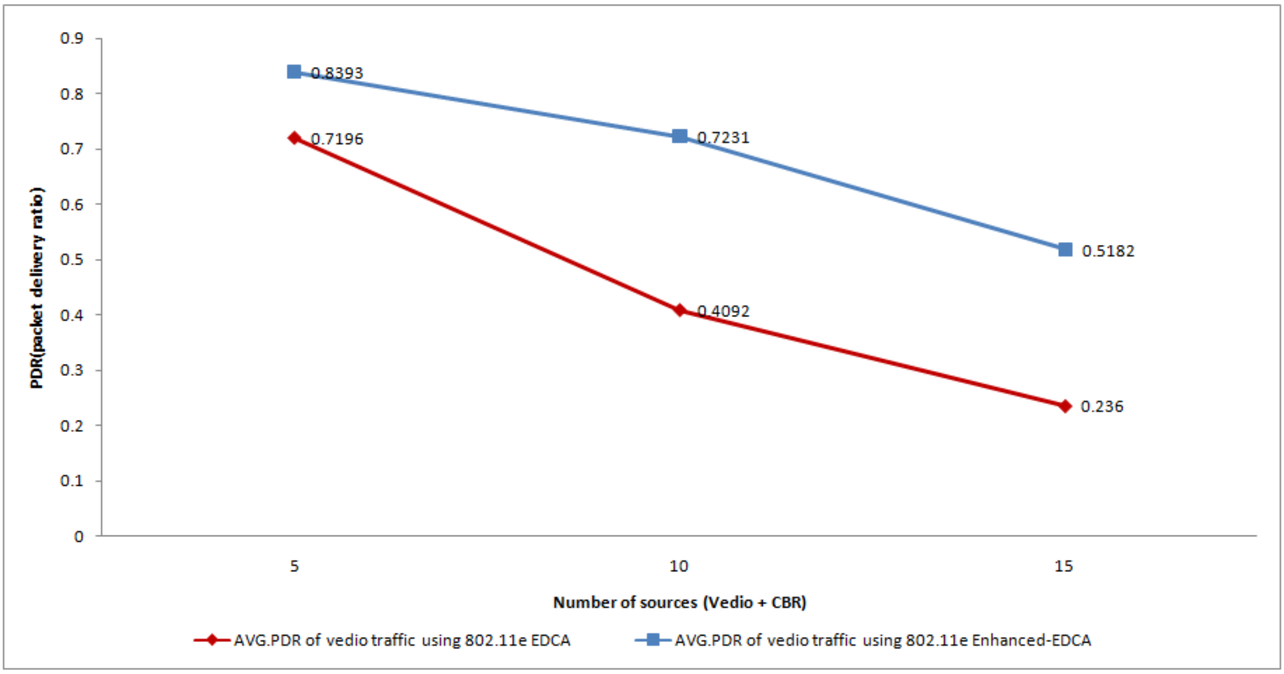


Figure 4.4: AVG PDR(Packet delivery ratio) with 51 nodes

**PSNR Results**

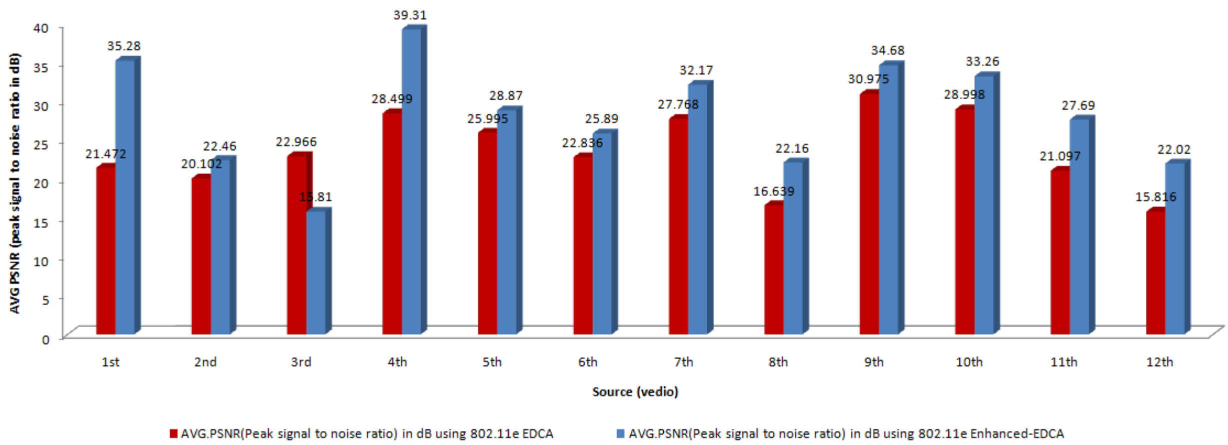


Figure 4.5: AVG PSNR(Peak signal to noise ratio) with 51 nodes

The video streaming application enjoys a large improvement in performance as is shown in Fig. 4.5. The vast majority of PSNR values are well above 25 dB, while exising EDCA exhibit significant performance degradation due to congestion at sensor nodes. Values of PSNR above 30 dB generally indicate a good video quality, while values below 25 dB are interpreted as poor playout experience. Obviously, Enhanced-EDCA outperforms the standard EDCA access method, mainly due to its selective prioritization of significant video frames and dynamic mapping of least significant frames to appropriate AC's during congestion.

### End-to-end delay Results

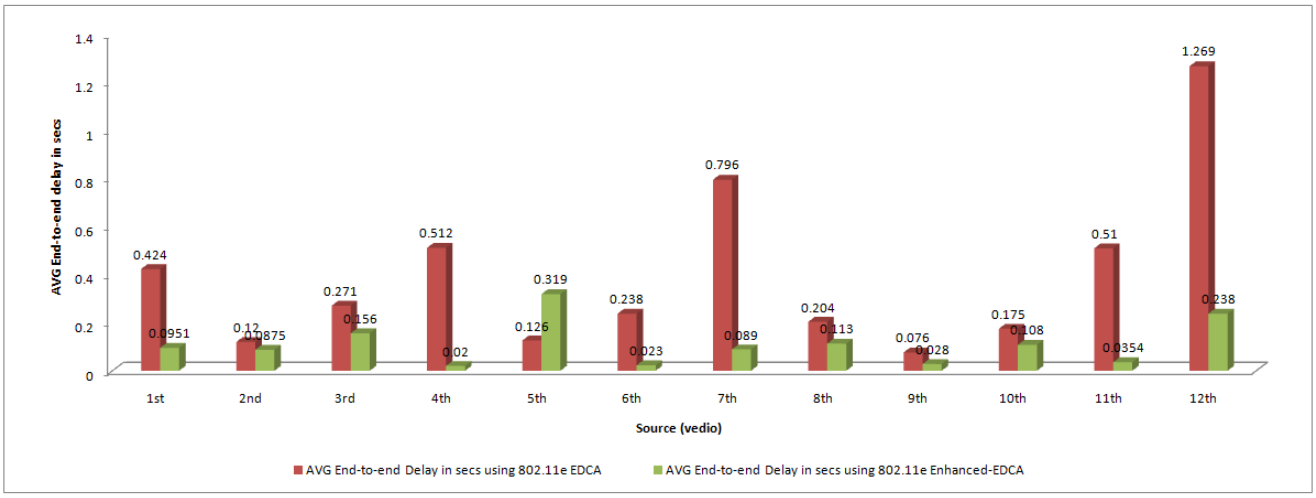


Figure 4.6: AVG.End-to-end delay with 51 nodes

### Standard Deviation Results

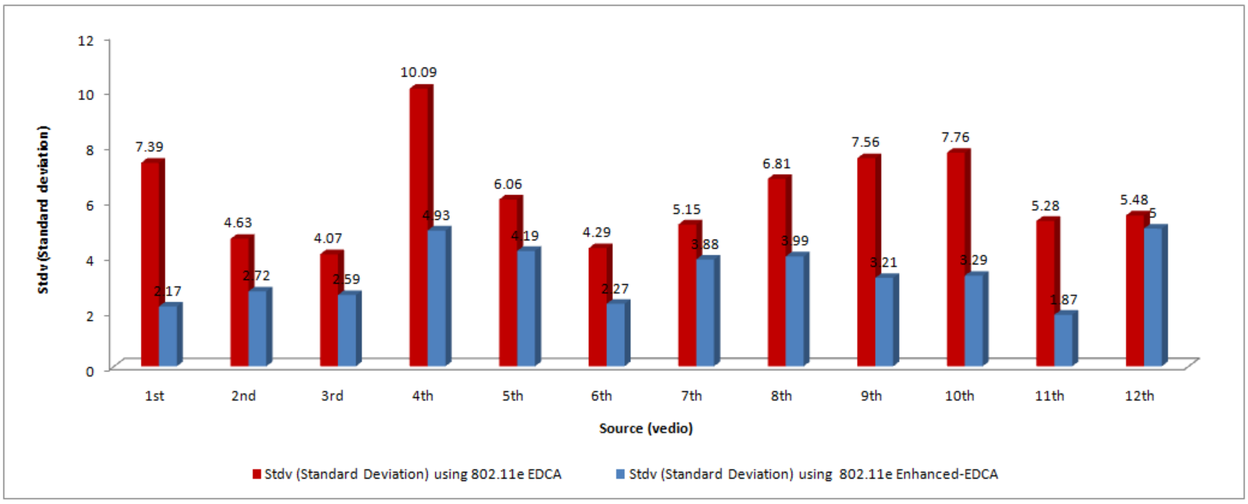


Figure 4.7: Standard Deviation with 51 nodes

# Chapter 5

# Conclusion and Future Work

In this work a cross-layer mechanism is presented, under the name Enhanced-EDCA. The mechanism is positioned to all the nodes in a wireless infrastructure network. Enhanced-EDCA exploits the multimedia packet semantics to selectively prioritize and protect important multimedia frames as well as dynamically assign network resources and exploits the inner characteristics of multimedia coding techniques together with the joint design of network protocols to achieve higher efficiency in visual sensor networks.Furthermore, none of the existing schemes make use of MAC layer strategies (such as acknowledgment policies or TXOP limit adaptation techniques), dynamically tuning the TXOP limit parameter and mapping of frames to appropriate AC's according to the queuelength, improves the performance of multimedia traffic over sensor network. By combining application level information and MAC layer strategies, Enhanced-EDCA has proven to be extremely beneficial in terms of QoS evaluation metrics.

In these approach acknowledgment policies of 802.11e EDCA is not efficiently used. One can use no ack policy of 802.11e EDCA for B frames and cbr packets and blocked ack policy for I and P frames so that number of transmissions can be reduce and as a result it will reduce the average energy consumption of all nodes.

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