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CROSS LAYER FRAMEWORK FOR EFFICIENT MPEG-4 VIDEO STREAMING OVER IEEE 802.11e EDCA IN MANET

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CROSS LAYER FRAMEWORK FOR EFFICIENT MPEG-4 VIDEO STREAMING OVER IEEE 802.11e EDCA IN MANET

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A thesis submitted in partial fulfillment of the requirements for the degree of Master of Science in Information and Communication Engineering

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Recommendation

The undersigned certify that they have read and recommended to the Department of Electronics and Computer Engineering for acceptance, a thesis entitled **"CROSS LAYER FRAMEWORK FOR EFFICIENT MPEG-4 VIDEO STREAMING OVER IEEE 802.11e EDCA IN MANET"**, submitted by **Ujjwal Pokharel** in partial fulfillment of the requirement for the award of the degree of "**Master of Science in Information and Communication Engineering**".

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This thesis entitled **"CROSS LAYER FRAMEWORK FOR EFFICIENT MPEG-4 VIDEO STREAMING OVER IEEE 802.11e EDCA IN MANET"** submitted by **Mr. Ujjwal Pokharel** for the partial fulfillment of the requirements for the award of degree of **"Master of Science in Information and Communication Engineering"** has been accepted as a bonafide record of work carried out by him under our department.

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ABSTRACT

Cross layer design framework is a new model in network architecture design that takes into account the dependencies and interactions among layers and supports optimization across layer boundaries. The new idea behind this research is a cross layer framework between Application and MAC & MAC and Network layer. It is carried out for efficient MPEG-4 video transmission over 802.11e Enhanced Distributed Channel Access. The joint optimization between Application and MAC is proposed to provide adaptive mapping of video frames based on its priority information and network traffic load. The MAC and Network layer is jointly optimized to design congestion adaptive routing protocol that will allow video streaming in a multi hop mobile ad hoc networks. The proposed algorithm will be evaluated with the help of performance parameters such as Peak Signal to Noise ratio, Average End to End Delay, Throughput and Packet Delivery Ratio.

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LIST OF ABBREVIATIONS

AC	Access Category
AIFS	Arbitration Inter Frame Space
AODV	Ad-hoc on Demand Distance Vector
AQM	Active Queue Management
AVC	Advanced Video Coding
DCF	Distributed Coordination Function
DCT	Discrete Cosine Transform
DIFS	Distributed Coordination Function Inter Frame Space
DP	Data Partitioning
DSR	Dynamic Source Routing
EDCA	Enhanced Distributed Channel Access
ERED	Enhanced Random Early Detection
EWMA	Exponentially Weighted Moving Average
GOP	Group of Picture
HCCA	HCF Controlled Channel Access
HCF	Hybrid Coordination Framework
HDTV	High Definition Television
ICMP	Internet Control Messaging Protocol
IEEE	Institute of Electrical and Electronic Engineers
IPTV	Internet Protocol Television
ISO	International Standard Organization
ITU	International Telecommunication Union
JPEG	Joint Photographic Expert Group
MAC	Medium Access Control
MPEG	Motion Picture Expert Group
NAL	Network Abstraction Layer
PSNR	Peak Signal to Noise Ratio
QoS	Quality of Service
QLR	QoS Library Redirection

RED	Random Early Detection
RREQ	Route Request
RREP	Route Reply
VCL	Video Coding Layer
WCI	Wireless Class Information
WEH	Wireless Extension Header
WLAN	Wireless Local Area Network
WRED	Weighted Random Early Detection

LIST OF SYMBOLS

α	Lag Parameter when $q < min_{th}$
β	Lag Parameter when $q > max_{th}$
CW_{max}	Maximum Contention Window Size
CW_{min}	Minimum Contention Window Size
hop_count	Hop Count
max_{th}	Maximum Threshold
max_p	Large Presupported Threshold of P_b
min_{th}	Minimum Threshold
n	Constant Proportional to Average Hop Count
P_b	Dropping Probability
Prob_New	New Dropping Probability
Prob_X	Dropping Probability of Video Frames.
P_B	Dropping Probability of B Frame
P_I	Dropping Probability of I Frame
P_P	Dropping Probability of P Frame
P_{LT}	Packet Loss
q	Current Queue Size
q(n)	Instantaneous Queue Size
$Q_{avg}(n)$	Average Queue at time instant n
$Q_{avg}(n-1)$	Average Queue at time instant n-1
TXOPlimit	Transmission Opportunity Limit
W_q	Weighted Average

CHAPTER 1
INTRODUCTION

1.1 Background

Over the past decades, wireless network access and video streaming services have become more popular than ever [21]. The use of wireless networks has spread further than simple data transfer to delay sensitive and loss tolerant multimedia applications. The applications such as video streaming and telephony are becoming an important part of the network user experience and expectation [16]. Raising the quality of the streaming service in wireless networks has been the focus of intensive research over the years

Mobile multimedia communication is especially challenging due to time varying transmission characteristics of the wireless channel and the dynamic Quality of Service (QoS) requirements of the application. Setting the control modes and tuning the parameter of the protocols at design time and for the worst case scenario lead to poor performance and inefficient utilization of resources. Instead, a network observing a behavior of the application and the physical channel dynamically adapting to the changes is able to maintain optimal allocation of resources. This requires timely exchange of parameter across layers and periodic reconfiguration of modes and parameters of the protocol layers during network operation [38].

The unique characteristics of wireless and ad-hoc networks call for new design paradigms that move beyond conventional layering [31]. Cross layer design framework is a new model in network architecture design that takes into accounts the dependencies and interactions among layers and supports optimization across layer boundaries [38]. This new approach allows sharing knowledge between different layers to obtain the highest possible adaptivity and can provide increased network efficiency and better QoS support [18].

In this research, the main objective of cross layer design framework is to share the knowledge between Application and Medium Access Control (MAC) & MAC and Network Layer to provide efficient MPEG-4 video transmission over IEEE 802.11e Enhanced Distributed Channel Access (EDCA) networks. To provide better video quality at the receiver, adaptive queue management in MAC and congestion adaptive routing in Network layer will be implemented with the aid of above cross layer framework. The performance parameter like Queue space utilization, Delay and Frame Jitter, Packet Loss Rate are important for evaluating the system.

Intensive research has been carried out in recent years in the domain of cross layer design with the aim of developing efficient cross layer framework for video transmission over wireless networks [28].

1.2 Statement of Problems

Various researches have been done in the domain of cross layer framework to support efficient MPEG-4 video streaming over IEEE 802.11e networks. Many challenging problems are still present in this domain. Among them, the problems that are focused in this research are enlisted below.

- Cross layer with static queue management is used to map MPEG-4 video frames (I/P/B) into Access Categories (ACs) of IEEE 802.11e MAC. No adaptive queue management approach is imposed. This non adaptive queue management approach results loss of important video frames which lead towards degradation of video quality at the receiver.

- The non congestion adaptive routing protocol such as Ad-hoc On demand Distance Vector(AODV) or Dynamic Source Routing (DSR) is implemented on top of MAC. No cross layer congestion adaptive routing protocol is implemented with adaptive mapping of video frames into ACs of MAC. This non congestion adaptive routing protocol results loss of important video frames for multi hop video streaming in mobile ad hoc networks.

1.3 Objectives

The main goal of this research is to address the problems listed above and to design efficient cross layer framework for MPEG-4 video streaming over IEEE 802.11e EDCA networks. It will help to increase the perceived video quality at the receiver. The major objectives that will be fulfilled are enlisted below.

- To propose and evaluate cross layer adaptive queue management technique which involves the mapping of MPEG-4 video frames (I/P/B) to appropriate access categories of IEEE 802.11e according to the priority information of video frames and network traffic load of ACs using Enhanced Random Early Detection Mechanism.

- To propose and evaluate efficient multi hop video streaming in mobile ad hoc networks with the help of cross layer congestion adaptive routing.

1.4 Scope and Limitations

The scope and limitations of this research are discussed below.

- Only unicast mode of video streaming will be implemented, and sender node does not retransmit to recover the loss of packets.

- The concept of multipath routing will not be implemented in this research. Only unipath routing will be considered.

1.5 Report Structure

This report is organized in three chapters including the following chapters.

- **Chapter 1 "Introduction"** explains the background of the research, statement of problems, objectives and scope and limitations.

- **Chapter 2 "Literature Review"** describes the various concepts of Cross Layer frameworks used in video transmission along with problems and solutions achieved so far.

- **Chapter 3 "Design and Methodology"** explains the framework and algorithm of the proposed system.

- **Chapter 4 "Result and Discussion"** explains about the analysis of simulation results.

- **Chapter 5 "Conclusion and Recommendation"** provides brief conclusions of this thesis along with the future enhancements.

CHAPTER 2
LITERATURE REVIEW

2.1 Video Streaming in Wireless Networks

Video streaming over wireless networks is compelling for many applications, and an increasing number of systems are being deployed. Video streaming of news and entertainment clips to mobile phones is now widely available. The application of video streaming range from real time audiovisual communication to search and rescue operation in wireless ad-hoc networks which can save lives of people. An ad-hoc wireless network is a collection of wireless nodes that self organize into a network without the help of an existing infrastructure. Some or possibly all of these nodes are mobile [31]. The figure of video streaming over an ad hoc wireless network is given below.

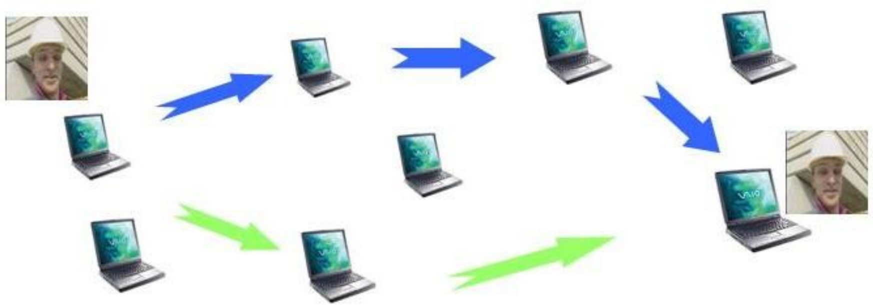

Figure 2.1: Video streaming over an ad hoc wireless network [31]

The video streaming requires a steady flow of information and delivery of packets by a deadline; wireless radio networks have difficulties to provide such a service reliably. The problem is challenging due to contention from other net-work nodes, as well as irregular interference from external radio sources such as microwave ovens or cordless phones. For mobile nodes, multi-path fading and shadowing might further increase the variability in link capacities and transmission error rate.

For such systems to deliver the best end-to-end performance, video coding, reliable transport and wireless resource allocation must be considered jointly, thus moving from the traditional layered system architecture to a cross-layer design [31].

2.2 Mobile Ad hoc Networks

2.2.1 Overview of MANET

Mobile Ad hoc Network (MANET) is self manageable and quickly deployable network. It consists of numerous mobile nodes scattered in a defined area. The communication of mobile nodes are done through the wireless link. It is performed in single hop or in multi hop fashion. It is somewhat synonymous to mobile, multi hop and wireless networking MANET is infrastructureless. Due to this feature, it enables ad hoc network to be installed at any time and location. The nodes of MANET can move freely. The sender node provides data packets to the destination node via intermediate nodes. These intermediate node acts as a router to send data packets to appropriate destination. There is a high probability of change in intermediate nodes due to the mobility. This enforces the mobile nodes to leave or join the network at any instant [39].

2.2.2 Applications of MANET

MANET is playing a key role in every facet of human civilization. It not only provides ubiquitous services but also help to save life of mankind. The evolution of mobile multimedia computing and services provide lots of opportunities in MANET. The modern information and communication technologies aid for the development of MANETs.

The application of MANET ranges from simple data acquisition to emergency rescue operation. MANETs are generally installed in an area where communication infrastructure is not available. The communication via MANET in rural area will enable people to raise their socio economic standards. The various other applications of MANETs are enlisted below [40].

- Ubiquitous multimedia services
- Virtual class rooms
- Security-sensitive computing environments.
- Conferences and workshops
- Context aware systems
- Location based systems
- Military communication in rescue and war
- Emergency communication in disaster areas [39]

2.2 Video Compression

2.2.1 Requirement of Video Compression in Video Streaming

Uncompressed video generally exceeds a network's bandwidth capacity, does not display properly, and requires too much disk space for storage purpose. Consequently, it is not practical to transmit video sequences without using compression.

Video compression technique reduces the high bit rate and large file sizes associated with digital video and allow the efficient transmission and storage of video data. In essence, these compression technologies reduce the quantity of data used to represent video content, making video files smaller with little perceptible loss in quantity. Compressed files are easier to transmit over a network and easier to store.

Most compression techniques register the differences within a frame or between frames in order to reduce the quantity of data used to represent video content. For differences within a single frame, compression techniques take advantage of the fact that the human eye is unable to distinguish small differences in color. These areas are averaged out without perceptible change to the viewer. For difference between frames, only the changes from one frame to the next are encoded. By ignoring redundant pixels, only the changed portion of a video sequence is compressed, thereby reducing overall file size. The above justification proves that video compression plays a vital role in video streaming [33].

2.2.2 Video Codec

A video codec is a software module that enables video compression or decompression of digital video. The word codec is a combination of any of the following: Compressor Decompressor, Coder Decoder, or Compression or Decompression. Codec encode a stream of signal for transmission, storage, or encryption and decode it for viewing or editing.

Most professional MPEG-based video codec use motion compensated differential coding known as P and B frames to improve compression. These codec generate key frames also known as Intra or I-frames which are set at user defined intervals. Motion compensated differential coding compares two compressed images to be transmitted over a network and uses the first compressed image as a reference frame (I-frame), sending only the parts of the following images (B and P frames) that differ from the reference image. [33]

2.2.3 Video Codec Standards

Video codec standards are developed and maintained by two international organizations.

- The International Telecommunication Union (ITU) publishes standards for videophone and videoconferencing applications, including H.264.

- The International Standards Organization (ISO) publishes computing standards through the Joint Photographic Experts Group (JPEG) and Moving Pictures Expert Group (MPEG).

The various Video Codec standards are discussed below.

MPEG-2

MPEG-2 was approved as a standard in 1994 and was designed for high frame and bit rates. MPEG-2 extends the earlier MPEG-1 compression standard to produce high quality video at the expense of a lower compression ratio and at a higher bit-rate. [33]

MPEG-2 is a standard currently in 9 parts. The first three parts of MPEG-2 have reached International Standard status, other parts are at different levels of completion. One has been withdrawn (ISO,2000).

MPEG – 4

MPEG-4 is an ISO/IEC standard developed by Moving Picture Expert Group (MPEG). It is the result of another international effort involving hundreds of researchers and engineers from all over the world. MPEG-4, with formal as its ISO/IEC designation "ISO/IEC 14496", was finalized in October 1998 and became an International Standard in the first month of 1999. MPEG-4 builds on the proven success of three fields. They are digital television, interactive graphics applications and interactive multimedia. It provides the standardized technological elements enabling the integration of the production, distribution and content access paradigms of the three fields. (ITU, 2000)

H.264 (MPEG-4 AVC)

H.264/MPEG-4 AVC is the latest video coding standard of ITU-T Video Coding Experts Groups (VCEG) and the ISO/IEC Moving Picture Expert Group (MPEG).

H.264/MPEG-4 AVC has recently become the most widely accepted video coding standard since the deployment of MPEG-2 at the dawn of digital television. It covers all common video applications ranging from mobile services and video conferencing to Internet Protocol Television (IPTV), High Definition Television (HDTV), and High Definition video storage [13].

2.3 Hierarchical Video Coding

With the increase of video and multimedia services within digital broadband communication systems, compatibility among different video standards has become a major issue in recent years. In particular the questions arise as how to use the available bandwidth as efficient as possible, how to achieve access to all services using only a single display device, and how to avoid intermediate format conversations between standards as much as possible. The hierarchical or compatible encoding concept has been quite successful recently because it leads to elegant video compression system providing for the above mentioned requirements.

The basic idea of hierarchical encoding is first to decompose a video signal into a low pass version and several bands containing high-frequency detail information. Next, the frequency bands are encoded independently using data compression algorithms that are highly adapted to statistical properties of individual bands. The decomposition must be done in both spatial and temporal direction to attend full compatibility among different video standards. By combining the (decoded) low resolution version with certain (decoded) high frequency bands, a signal can be recovered that has a higher spatial resolution [2].

Packets will carry data from only one layer and can be marked according to their importance for intelligibility for the end-user. The network would use this information as a measurement metric to determine the sort of packets should be dropped or delayed, and which should take higher priority. It is notable the priority bits already exist in some protocols such as the IP protocol.

Hierarchical coding will also be ideal to deal with multicasting transmission over links with different bandwidths. With hierarchical coding, low level packets can be filtered out whenever a low bandwidth link is encountered thus preserving the intelligibility of the video/audio for the sites affected by these links and still delivering a better quality to sites with higher bandwidth. [25]

2.3.1 MPEG-4 Video Structure

In order to reduce the required bandwidth, the basic coding scheme of MPEG-4 is to predict motion from frame to frame in the temporal direction, and then use Discrete Cosine Transform (DCT) to organize the redundancy in the spatial direction. The data rate of MPEG-4 is smaller than 64 Kbps, attempting to provide a compression scheme suitable for video conference. [12]

The MPEG-4 standard (ITU, 2000) defines three types of video frames for the compressed video stream, including I (Intra-coded), P (Predictive-coded) frame, and B (Bi-directionally predictive-coded) frame. The MPEG I frame is encoded independently and decoded by itself. Thus, the I frame is just a frame coded as a still image, without any relationship to any previous or successive frames. The P frame is encoded using prediction from the preceding I or P frames in the video sequence. Thus the P frame requires the information of the most recent I frame or P frame for encoding and decoding. The B frame is encoded using predictions from the preceding and succeeding I or P frames. According to the coding relation, in MPEG-4 video stream the most important video type is the I frame with the P frame being more important than B frame.

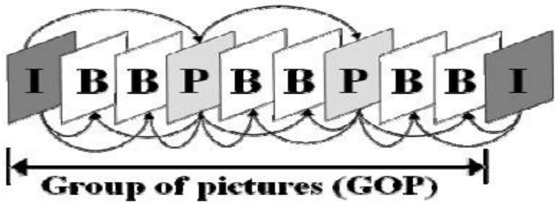

Figure 2.2: Prediction encoding of MPEG-4, GOP (N=9, M=3) [25]

The video sequence can be decomposed into smaller units termed as Group of Picture (GOP). A GOP pattern is characterized by two parameters N and M written as G (N,M). N is the I to I distance and M is the I to P distance. For example G (9,3) means that the GOP includes one I frame, two P frames, and six B frames. Similarly, the second I frame in the figure marks the beginning of the next GOP. The arrows indicate that the B frames and P frames decoded are dependent on the preceding or succeeding I or P frames. [41]

2.4 IEEE 802.11e MAC

IEEE 802.11 Task Group E currently defines enhancements to the 802.11 MAC called 802.11e. It aims to support QoS by providing differentiated classes of service in MAC layer and to enhance the ability of all physical layers so that they can deliver time-critical multimedia traffic. In IEEE 802.11e, a new MAC access method named hybrid coordination function (HCF) is introduced. The HCF consists of two parts. One is EDCA and the other is HCF controlled channel access (HCCA). The EDCA is the fundamental and mandatory mechanism of IEEE 802.11e, while HCCA is optional and requires centralized polling and scheduling algorithms to allocate the resources. [37]

3.4.1 Enhanced Distributed Channel Access (EDCA)

EDCA is designed to enhance the DCF mechanism and to provide a distributed access method that can support service differentiation among classes of traffic. It can provide up to four ACs. The four access categories include AC_VO (for voice traffic), AC_BE (for best effort traffic), AC_VI (for video traffic) and AC_BK (for background traffic). To simplify the notations, we assign AC_VO as AC[3], AC_VI as AC[2], AC_BE as AC[1], and AC_BK as AC[0]. Each AC has its own buffered queue and behaves as an independent back off entity. The priority among ACs is then determined by AC specific parameters called the EDCA parameter set. The EDCA parameter set includes minimum Contention Window Size

(**CW_{min}**), maximum Contention Window Size (**CW_{max}**), Arbitration Inter Frame Space (AIFS), and Transmission Opportunity limit (**TXOPlimi**t). [6]

To achieve differentiation, instead of using fixed DIFS, EDCA assigns higher priority ACs with smaller **CW_{min}**, **CW_{max}** and **AIFS** to influence the successful transmission probability (statistically) in favor of high-priority ACs. The ACs with smallest AIFS has the highest priority. The smallest the priority values (such as AIFS, **CW_{min}** and **CW_{max}**) the greater the probability of gaining access to the medium. The AIFS [AC] is determined by the expression given below.

AIFS [AC] = SIFS + AIFS [AC] * SlotTime

Where AIFS [AC] is an integer greater than zero. Moreover, the backoff counter is selected from [1, 1+CW [AC]] (Wen-Yen Lin, et.al, 2007). The default values of EDCA parameters are shown below in the table 2.1

Table 2.1: IEEE 802.11e EDCA Parameter Set [26]

Priority	**Access Category (AC)**	**Designation**	**AIFS**	**CW_{min}**	**CW_{max}**
3	AC_VO	Voice	2	7	15
2	AC_VI	Video	2	15	31
1	AC_BE	Best Effort	3	31	1023
0	AC_BK	Background	7	31	1023

From the above table, it is clear that priority of AC queue is higher with minimum value of parameters. The parameter value of AC that handles voice traffic is lower than other AC queues. The pictorial representation of four AC queues are shown below in figure 2.3

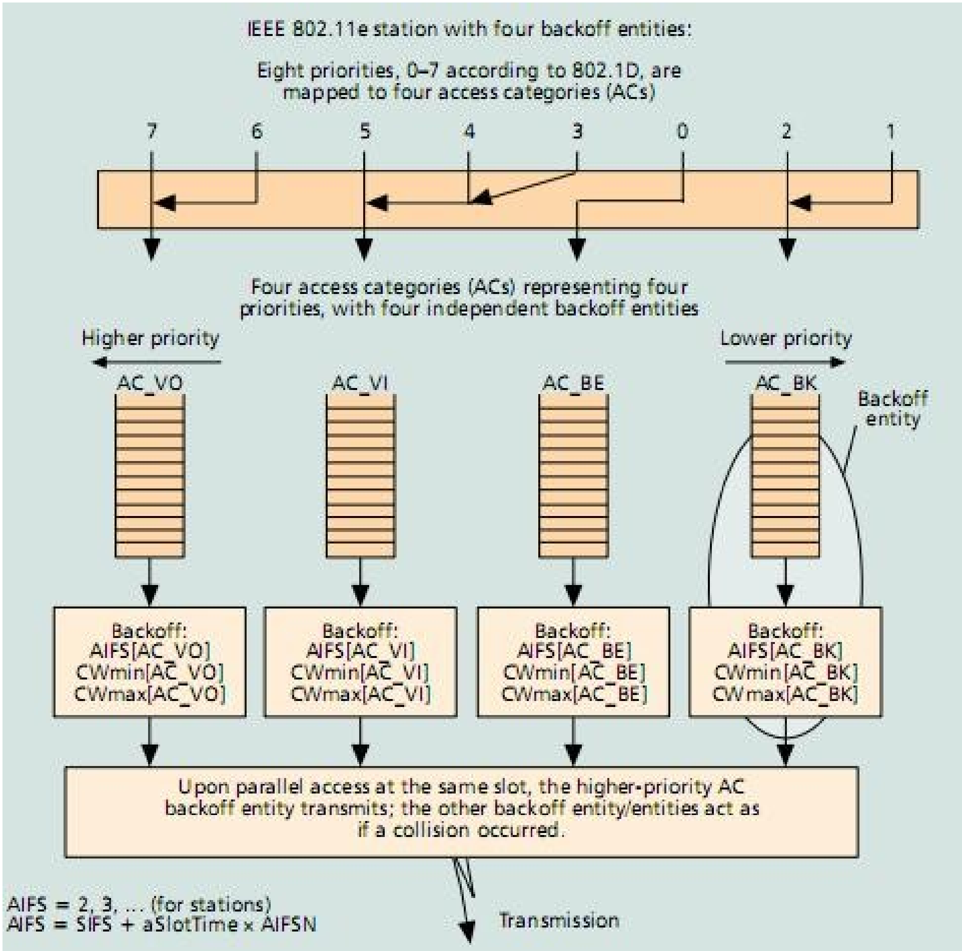

Figure 2.3: Four Access Categories in IEEE 802.11e [26].

Each AC 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 ideal for at least an AIFS period. The backoff procedure of each AC is the same as that of DCF. When a collision occurs among different ACs within the same station, the higher priority AC is granted the opportunity to transmit, while the lower priority AC suffers from a virtual collision, similar to real collision outside the station. IEEE 802.11e EDCA defines a TXOPlimit as the time interval during which a particular station can initiate transmissions. During this period, defined by a starting time and a maximum duration, stations are allowed to transmit multiple data frames from the same AC continuously within the time limit defined by TXOPlimit. The 802.11e EDCA mechanism parameters operation is shown below in the figure 2.4.

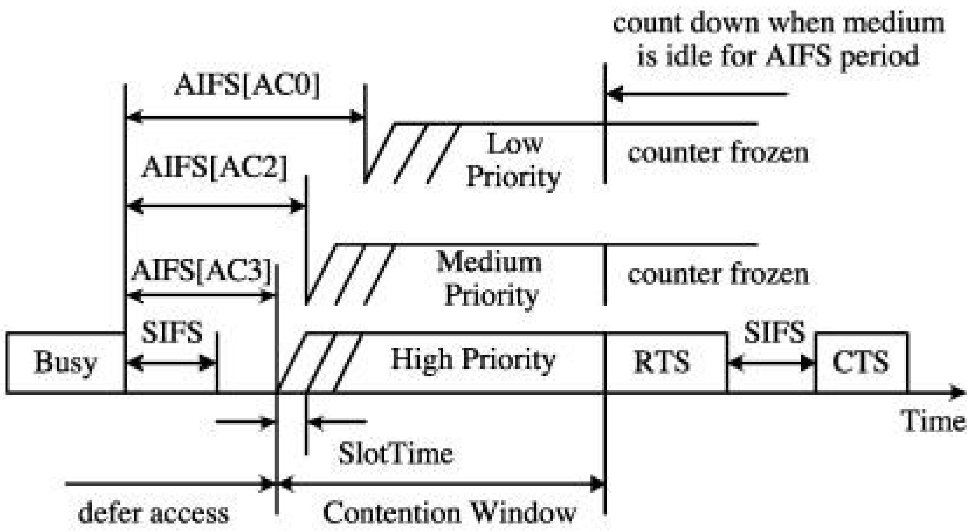

Figure 2.4: IEEE 802.11e EDCA mechanism parameters [37]

Priority differentiation used by EDCA ensures better service to high priority class while offering a minimum service for low priority traffic. Although this mechanism improves the quality of service of real-time traffic, the performance obtained is not optimal since EDCA parameters cannot be adapted according to the network conditions. [6]

2.5 Cross Layer Design

2.5.1 Cross Layer Design Principle

As wireless communication and networking fast occupy center stage in research and development activity in the area of communication networks, the suitability of one of the foundations of networking, the layered protocol architecture, is coming under close investigation from the research community. It is repeatedly argued that although layered architecture have served well for wired networks, they are not suitable for wireless networks.

To fully optimize wireless broadband networks, both the challenges from the physical medium and QoS demands from the applications have to be taken into account. Rate, power and coding at physical layer can be adapted to meet the requirements of the applications given the current channel and network conditions. Knowledge has to be shared between (all) layers to obtain the highest possible adaptivity. Hence, the whole idea behind cross layer design is to combine the resources available in the different communities, and create a network which can be highly adoptive and QoS-efficient by sharing state information between different processes or modules in the system [18]. Different approaches of cross layer architecture for video streaming are discussed below.

Top-down

The higher layer optimizes their parameters and the strategies at the next lower layer.

Bottom-up

In this architecture the lowest layer isolates the higher layers from losses and bandwidth variations.

Application centric approach

The application layer optimizes the lower-layer parameters one at a time in either bottom-up or top-down manner, based on its requirements.

MAC-centric approach

In this cross-layer technique the application layer passes its traffic information and requirements to the MAC, it then decides which application layer packets/flows should be transmitted at what QoS level.

Integrated approach

The strategies to design cross layer architecture are determined jointly by all the open system interconnection (OSI) layers. [24]

2.5.2 Cross Layer Design Architecture

The various architectures of cross layers are creation of new interfaces, merging of adjacent layers, design coupling with new interfaces and vertical calibration across layers. The diagrammatic representation of various cross layer architecture are shown below.

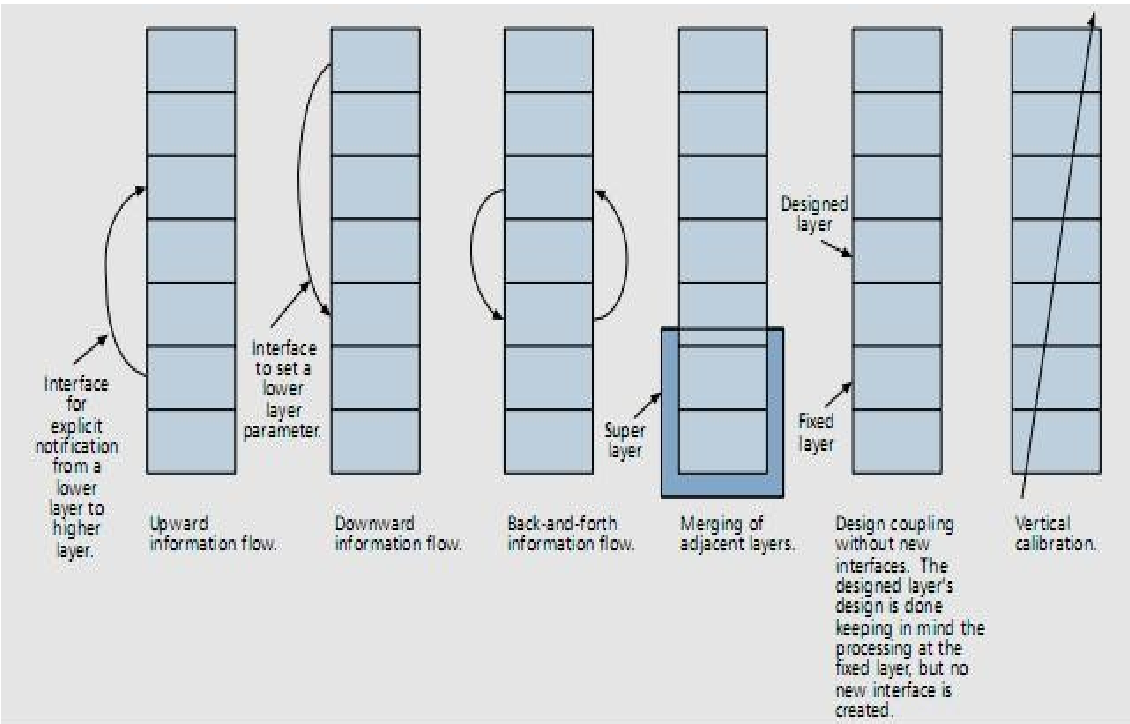

Figure 2.5: Different Cross Layer Architectures [32]

Creation of New Interfaces

The creation of new interfaces architecture is based on the design of new interfaces that are used for information sharing between the layers at runtime. This architecture is subdivided into three subcategories depending on the direction of information flow along the new interfaces. They are upward information flow, downward information flow, back and forth information flow [32].

Merging of adjacent layers

Merging of adjacent layers is the way to do cross layer design where two or more adjacent layers together are merged such that the service provided by the new super layer is the union of the services provided by the constituent layers. This does not require any new interfaces to be created in the stack. The super layer can be interfaced with the rest of the stack using the interfaces that already exists in the original architecture [32].

Design coupling

Design coupling without new interfaces involves coupling two or more layers at design time without creating any extra interfaces for information sharing at runtime. Here no new interfaces are created; the architectural cost here is that it may not be possible to replace one layer without making corresponding changes to another layer [32].

Vertical calibration

Vertical calibration across layers refers to adjusting parameters that span across layers. Basically, the performance seen at the level of the application is a function of the parameters at the layers below it. Hence, it is conceivable that joint tuning can help to achieve better performance than individual settings of parameters. It can be done in static as well as dynamic manner. In static manner, parameters are set across the layers at design time with the optimization of some metric [32].

2.5.3 Cross Layer Signaling Methods

The signaling concept is an import in any cross layer design approach. It deals the way to extract the information from other layers. Various signaling approaches are explained below.

Packet Headers

In IPV6, optional Network layer information can be encoded in additional headers. The Interlayer signaling pipe takes advantage of this new feature by storing cross-layer information in the Wireless Extension Header (WEH). This method makes use of IP data packets as in-band message carriers with no need to use a dedicated internal message protocol. However, an IP packet normally can only be processed layer by layer, and it is not easy for higher layers to access to IP-level header [29].

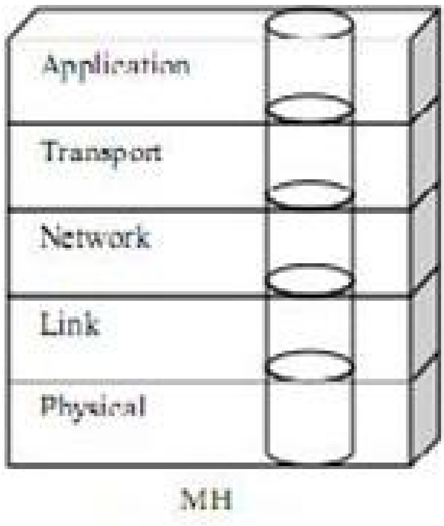

Figure 2.6: Packet Header with Signaling Pipe [29]

ICMP Messages

Internet Control Message Protocol (ICMP) is widely deployed signaling protocol in IP based networks. Compared to the above method, it is to "punch holes in the protocol stack" and propagate information across layers by using ICMP messages. In this scheme, desired information is abstracted to parameters, measured by corresponding layers wherever convenient. A new ICMP message is generated only when a parameter changes beyond the threshold. However, an ICMP message is always encapsulated in IP packet and this indicates that the message has to pass by Network Layer even if the signaling is only desired between Link Layer and Application Layer [29].

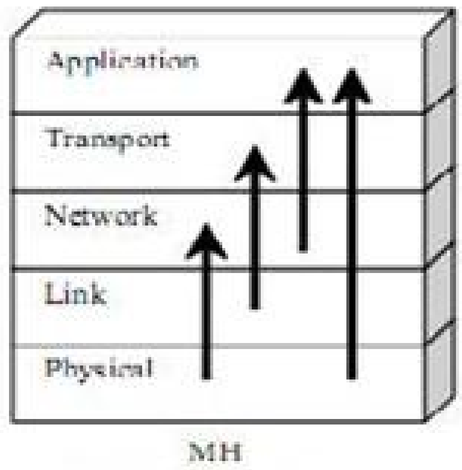

Figure 2.7: ICMP Messages with Selected Holes [29]

Network Service

In this scheme, channel and link states from physical layer and link layer are gathered, abstracted and managed by third parties, the distributed Wireless Class Information (WCI) servers. Interested applications then access to their required parameters from the lowest two layers. The figure of this method is given below. [29]

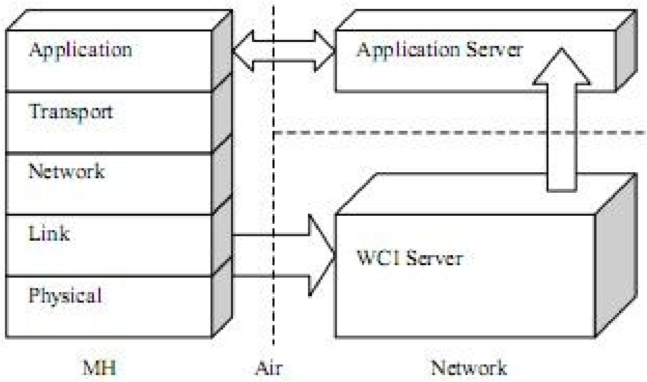

Figure 2. 8: Network Service Model [29]

Local Profiles

Local profiles are used to store periodically. Cross layer information is abstracted from each necessary layer respectively and stored in separate profiles with the host. Other interested layers can then select the profile(s) to fetch the desired information. In this method, internal profiles are maintained rather than external servers [29].

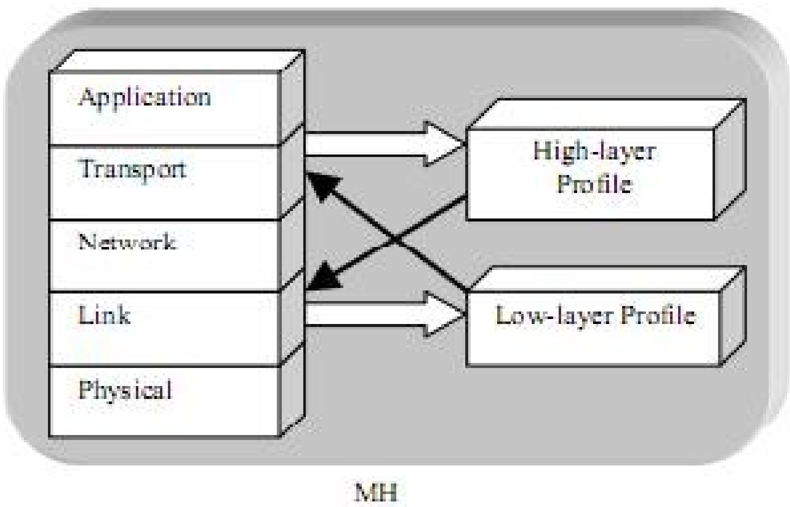

Figure 2.9: Local Profiles [29]

2.6 Active Queue Management Algorithm

Packet loss is an important mechanism to notify congestion. Active Queue Management (AQM) algorithm is the key technology of congestion control which is recommended by IETF. The various AQM algorithms are Random Early Detection (RED), Adaptive RED (ARED), Weighted RED (WRED) and Enhanced RED (ERED). [10]

2.6.1 Random Early Detection

RED algorithm measures congestion by the average queue size and drops packets randomly before the queue overflows. When a packet arrives at the queue and if the average queue length is less than minimun threshold (**min_{th}**) no drop action will be taken and the packet will be enqueued. If the average queue is greater than **min_{th}** but less than maximum threshold (**max_{th}**) an early drop test will evaluated by **P_b**. If the average queue is greater than **max_{th}**, a forced drop operation will occur. (Babek Abbasov, et.al, 2008)

The core of RED is to calculate the average queue length from the current queue length by the Exponentially Weighted Moving Average (EWMA). The average length of the queue is calculated below.

$$Q_{avg}(n) = (1-W_q).\ Q_{avg}(n-1) + W_q.q(n) \tag{2.1}$$

Here **$Q_{avg}(n)$** is the average queue size of time instant n and **$Q_{avg}(n-1)$** is the average queue size of time instant n-1. **W_q** is the weighted average and **$q(n)$** is the instantaneous queue. The formula for discard probability of RED is expressed below.

$$P_b = \begin{cases} 0 & Qavg(n) < min_{th} \\ 1 & Qavg(n) > max_{th} \\ \frac{Q_{avg}(n) - min_{th}}{max_{th} - min_{th}}\ max_p & min_{th} \le Q_{avg}(n) \le max_{th} \end{cases} \tag{2.2}$$

Here $\mathbf{max_p}$ is the largest presupposed threshold of $\mathbf{P_b}$ which steps into 1 from $\mathbf{max_p}$ while $\mathbf{Q_{avg}} > \mathbf{max_{th}}$. In practice $\mathbf{P_b}$ must be modified properly to make discard interval uniform. [10]

2.6.2 Adaptive RED

Delay being a major component of the quality of service delivered to their customers, network operators would naturally like to have a rough a priori estimate of the average delays in their congested routers. To achieve such predictable average delays with RED would require constant tuning of RED's parameters to adjust to current traffic conditions. A second related weakness of RED is that the throughput is also sensitive to the traffic load and to RED parameters. In particular, RED often does not perform well when the average queue becomes larger than maxth resulting in significantly decreased throughput and increased dropping rates. Avoiding this regime would again require constant tuning of the RED parameters.

Adaptive RED reduces both the packet loss rate and the variance in the queuing delay. Hence it appears to solve the problems of setting RED parameters which has been one of the banes of RED's existence. It is implemented in four ways.

- $\mathbf{max_p}$ is adapted not just to keep the average queue size between min_{th} and max_{th}, but to keep the average queue size within a target range half the way between minth and maxth.

- $\mathbf{max_p}$ is adapted slowly, over time scales greater than a typical round-trip time and in small steps.

- $\mathbf{max_p}$ is constrained to remain within the range [0.01,0.5]

- Instead of multiplicatively increasing and decreasing maxp, we use an Additive Increase Multiplicative Decrease (AIMD) policy.

The guideline of adapting maxp slowly and infrequently allows the dynamics of RED of adapting the packet dropping probability in response to changes in average queue size to dominate on smaller time scales. (Floyd S, et al., 2001)

2.6.3 Enhanced RED

It is very difficult to provide proper RED parameters to adapt to the dynamic changes of network in the practical application. A large discard probability must be set in order to keep the average queue length between $\mathbf{min_{th}}$ and $\mathbf{max_{th}}$. In some cases, even if the probability is close to maxth, it still cannot meet actual requirements. So the RED algorithm should be improved.

The basic idea of ERED is that the current queue length and the average queue length is combined to determine $\mathbf{Q_{avg}(n)}$. The calculation of $\mathbf{Q_{avg}(n)}$ should be adjusted separately when the current queue length $\mathbf{q < min_{th}}$, $\mathbf{min_{th} \le q \le max_{th}}$ and $\mathbf{q > max_{th}}$. It is done to improve the $\mathbf{Q_{avg}(n)}$ and enhance the adaptability of RED. The $\mathbf{Q_{avg}(n)}$ computation is given below.

$$Q_{avg}(n) = \begin{cases} \frac{1-W_q}{\alpha}.Q_{avg}(n-1) + w_q.q(n), & q < min_{th} \\ (1-w_q).Q_{avg}(n-1) + w_q.q(n), & min_{th} \le q \le max_{th} \\ (1-w_q).Q_{avg}(n-1) + \frac{w_q.q(n)}{\beta}, & q > max_{th} \end{cases} \quad (2.3)$$

$\boldsymbol{\alpha}$ and $\boldsymbol{\beta}$ are called lag parameters. After the computation of $\mathbf{Q_{avg}(n)}$ then only we can define packet dropping probability based on $\mathbf{P_b}$ which is given below.

$$P_b = \begin{cases} 0 & Qavg(n) < min_{th} \\ 1 & Qavg(n) > max_{th} \\ \frac{Q_{avg}(n)-min_{th}}{max_{th}-min_{th}}\, max_p & min_{th} \le Q_{avg}(n) \le max_{th} \end{cases} \quad (2.4)$$

The experiment shows that by introducing the lag parameters (**α** and **β**), the average queue length of ERED can better track the changes of current queue length closely in comparison with RED. It also shows that the packet loss and stability are better than RED algorithm. [10]

2.7 Routing in Ad-hoc Networks

An ad hoc routing protocol is a convention, or standard that controls how nodes decides which way to route packets between computing devices in a mobile ad hoc network. It is divided into three broad categories namely flat routing, hierarchical routing and geographic position assigned routing. Flat routing is again subdivided into two categories proactive (table-driven) and reactive (on demand) routing protocol. There are two on demand routing protocols namely Dynamic Source Routing (DSR) and Ad hoc On-Demand Distance Vector Routing (AODV) (Krishna Gorantala, 2006).

2.7.1 Dynamic Source Routing

The DSR is a simple and efficient routing protocol designed specifically for use in multi-hop wireless ad hoc networks of mobile nodes. DSR allows the network to be completely self-organizing and self-configuring, without the need for any existing network infrastructure or administration. The protocol is composed of the two mechanisms of Route Discovery and Route Maintenance, which work together to allow nodes to discover and maintain source routes to arbitrary destinations in the ad hoc network. The use of source routing allows packet routing to be trivially loop-free, avoids the need for up-to-date routing information in the intermediate nodes through which packets are forwarded, and allows nodes forwarding or overhearing packets to cache the routing information in them for their own future use. All aspects of the protocol operate entirely on-demand, allowing the routing packet overhead of DSR to scale automatically to only that needed to react to changes in the routes currently in use.

The DSR protocol allows nodes to dynamically discover a source route across multiple network hops to any destination in the ad hoc network. Each data packet sent then carries in its header the complete, ordered list of nodes through which the packet must pass, allowing packet routing to be trivially loop-free and avoiding the need for up-to-date routing information in the intermediate nodes through which the packet is forwarded (David B. Johnson, et al., 2001)

2.7.2 Ad hoc On-Demand Distance Vector Routing

The AODV algorithm enables dynamic, self-starting, multi hop routing between participating mobile nodes wishing to establish and maintain an ad hoc network. AODV allows mobile nodes to obtain routes quickly for new destinations, and does not require nodes to maintain routes to destinations that are not in active communication. AODV allows mobile nodes to respond to link breakages and changes in network topology in a timely manner. The operation of AODV is loop-free, and by avoiding the Bellman-Ford "counting to infinity" problem offers quick convergence when the ad hoc network topology changes (typically, when a node moves in the network). When links break, AODV causes the affected set of nodes to be notified so that they are able to invalidate the routes using the lost link. New distinguishing feature of AODV is its use of a destination sequence number for each route entry. The destination sequence number is created by the destination to be included along with any route information it sends to requesting nodes. Using destination sequence numbers ensures loop freedom and is simple to program. Given the choice between two routes to a destination, a requesting node is required to select the one with the greatest sequence number. The AODV messaging is represented below in the diagram. (C. Perkins, et al., 2003)

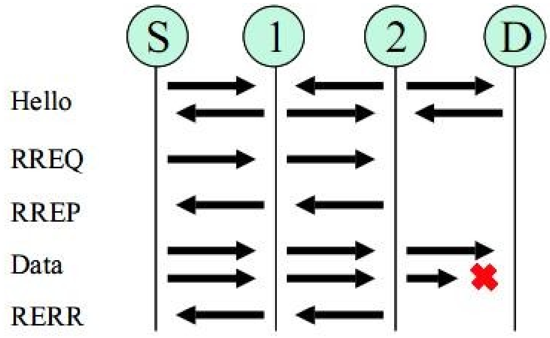

Figure 2.10: AODV Protocol Messaging [4]

2.8 Related Works

Many works have been done in the domain of cross layer design for efficinet video streaming over IEEE 802.11e EDCA networks. The various researches that have been done in past are disussed below.

[24] proposed a cross layer architecture for an improvement of H.264 video transmission over IEEE 802.11e. This proposed cross layer architecture relies on a data partitioning (DP) technique at the application layer and an appropriate QoS mapping at the 802.11e based MAC layer. Through employing DP, the H.264 encoder partitions the compressed data in separate units of different importance called partitions. Based on QoS requirement of those different partitions, a marking algorithm at the MAC layer that associates each partition with an AC is provided by 802.11e EDCA. The proposed QoS mapping algorithm at MAC layer is static and video streaming is also limited to single hop.

[21] proposed a hybrid design framework for video streaming in IEEE 802.11e wireless network. In this research, a hybrid design framework was proposed. It consists of MAC centric cross layer architecture to allow MAC layer to retrieve video streaming packet information, a hybrid retransmission deadline and retry limit to save unnecessary packet waiting time, and a single video multi-level queue to prioritize I/P/B slice. The mapping of video frames into different ACs is static. The approach of video streaming is also limited to only single hop.

[5] proposed a cross layer architecture for improving scalable video transmission over IEEE 802.11e networks. In this research, a cross layer architecture is enabled between Network Abstraction Layer (NAL) and IEEE 802.11e MAC layer. It provides differentiated services according to the importance of scalable video packets. The Scalable Video Coding (SVC) Video Coding Layer (VCL) transforms video frames into base layer and enhance layer. The six kinds of slices (I, P, B1, B2, B3 and PR) are obtained. Each slice is sent to NAL and the priority of slice is defined according to its importance. Mapping algorithm is used in MAC layer based on the importance of video slices. The mapping of all video slices are done in a static manner and video transmission is also limited to single hop.

(Robin, et al., 2008) proposed a cross layer design for H.264 video stream over wireless local area networks. It is based on top-down cross layer approach and it completely takes support of the characteristics of both application and MAC layers. This proposed scheme successfully associates the characteristics of priority access control to improve the video quality. The basic concept of this algorithm is that firstly, the video information is sent from the application layer to network layer. The network layer in turns communicates the priority information to the MAC layer. The MAC layer makes use of this proposed algorithm to map the video packets to the appropriate access categories based on the significance of the video data. The RED approach is used as a queue management mechanism for preventing the packet drop. This approach of video frames mapping is only based on simple RED. It considers only AC[2] queue. It has not incorporated multiple traffic flows and the video streaming is also limited to single hop.

(C.-H.Lin, et al., 2009) proposed an adaptive cross layer mapping algorithm for MPEG-4 video transmission over IEEE 802.11e WLAN. This adaptive cross layer approach is used to map video frames(I/P/B) into Access categories of MAC based on its priority information. The proposed algorithm uses RED technique to map video frames only to AC[2] but not to other Access Categories. This approach does not provide the support of multi hop video streaming. It is only limited to single hop.

[27] proposed congestion avoidance routing protocol for QoS aware Mobile Adhoc Networks. This work was based on cross layer between MAC and Network layer. In this research, Type of Service Aware (TSA) routing protocol was proposed. It was an enhancement of AODV routing protocol. TSA avoids congestion by distributing the load over a potentially greater area and therefore e improving the spatial reuse. It uses the load of ACs to determine the level of congestion. The adaptive mapping of packets into ACs is not carried out in this research.

In summary, it was found that many researches have done in the domain of cross layer design for efficient video streaming. Various cross layer frameworks between Application and MAC have been proposed. But still, the approaches of mapping video frames (I/P/B) into Access Categories of MAC are static. In an advancement, simple RED based approach has been proposed but it cannot map video frames in an adaptive manner based on priority of video information and network traffic load. These cross layer frameworks are also limited to single hop streaming. Congestion avoidance routing protocols have been proposed in previous researches. But, it cannot incorporate the adaptive mapping of video frames into AC queues of MAC.

This research is done in order to address the problems of past research works. Our approach of cross layer design is different than others because it includes the cross layer framework between Application and MAC & MAC and Network layers. Cross layer is created between Application and MAC layer to dynamically map video frames (I/P/B) into Access Categories of IEEE 802.11e MAC. The cross layer between MAC and Network layer is created to design congestion adaptive routing in order to provide video streaming in multi hop ad hoc networks.

CHAPTER 3
DESIGN AND METHODOLOGY

3.1 Background

This chapter deals with design and methodology of a cross layer framework for efficient MPEG-4 video streaming over IEEE 802.11e EDCA. The discussion about cross layer model, algorithm development and simulation model is done.

3.2 Cross Layer Model

The cross layer model of this research involves knowledge sharing between Application and MAC & MAC and Network layer. The generic cross layer model is represented in figure 3.1.

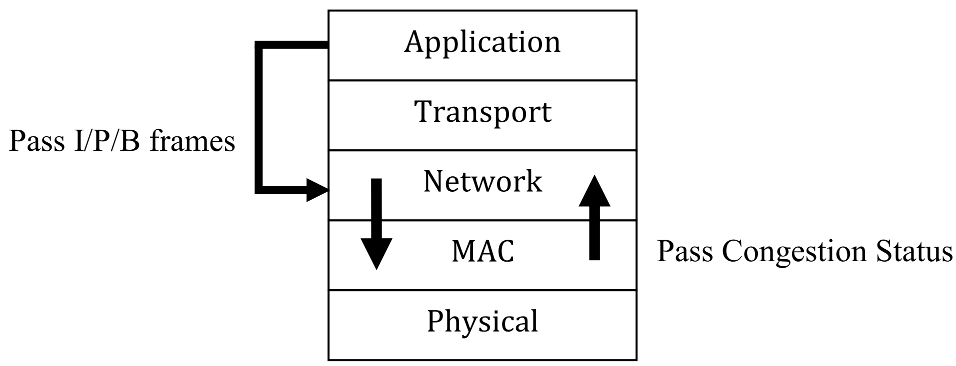

Figure 3.1: Generic Cross Layer Model

From the above figure, it is clear that application layer passes its video frames (I/P/B) information to MAC layer through the network layer. QoS Library Redirection (QLR) (Ce-Kuen Shieh, et al., 2001) method will be used in an application layer to pass video frames information to MAC layer. This method passes video frame information into Differentiated Service Code Point (DSCP) field of network layer. The network layer in turn will pass the video frame information to EDCA based MAC layer. The main advantage of QLR is that applications can transparently pre-mark their video frames without any modification of source codes. Based on the frame received, MAC layer will perform adaptive queue management technique to allocate video frames (I/P/B) into different Access category queues.

In order to make congestion adaptive routing, MAC layer passes its congestion status information to the network layer. The status of congestion is determined based on the traffic of Access category queues. Based on the congestion information received, Network layer will make its routing decision.

3.3 Algorithms

In this research, two algorithms are proposed for efficient MPEG-4 video streaming over IEEE 802.11e MAC. They are Adaptive mapping algorithm in MAC layer and Congestion adaptive routing algorithm in Network layer.

3.3.1 Adaptive Mapping Algorithm

The idea of adaptive mapping algorithm is to map MPEG-4 video frames (I/P/B) to different Access categories of MAC layer based on their significance information. The framework of adaptive mapping is represented in figure 3.2.

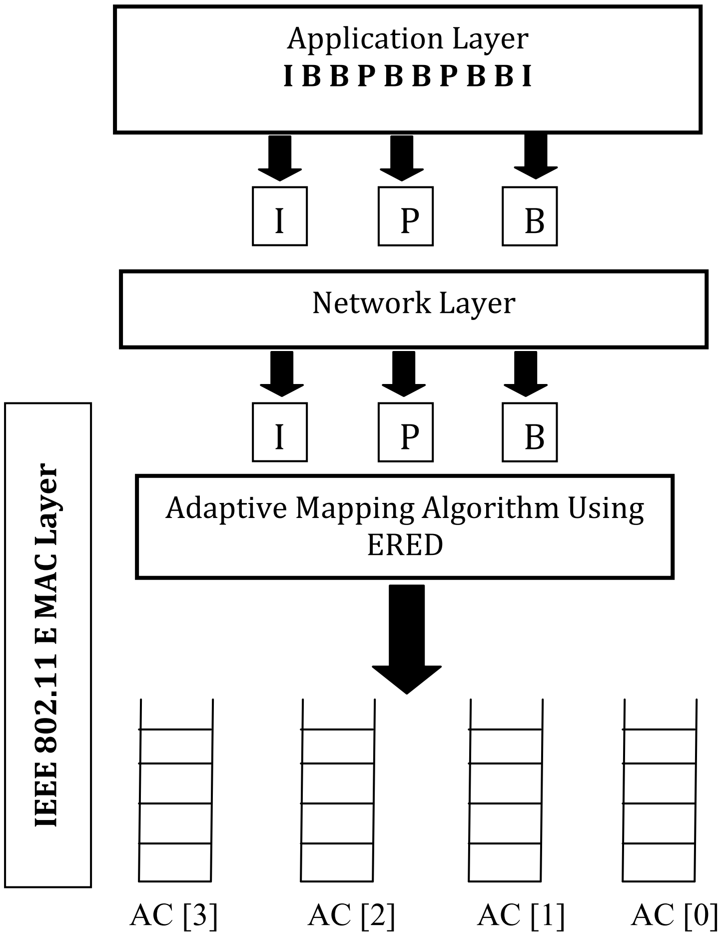

Figure 3.2: Cross Layer framework of Adaptive Mapping Algorithm using ERED

This framework is designed in order to improve the quality of delivered video. The proposed algorithm dynamically allocates the video to the most appropriate AC at the MAC layer based on significance of video type and network traffic load. To allocate important video frame into higher AC queue, we use different mapping probabilities defined as **Prob_X** to different video frames. The dropping probability of B frame is higher than that of P and I frame i.e. $\mathbf{P_B > P_P > P_I}$.

The AC [2] queue is dedicated for video traffic. Other queues AC [1] and AC [0] have least transmitting priority than AC [2]. The queue AC [2] ,AC [1] and AC[0] is monitored with ERED algorithm in order to predict upcoming congestion and dropping of packets. The two parameters of queue threshold namely $\mathbf{min_{th}}$ and

max_{th} are used to monitor the queues to avoid dropping of video frames. The parameters **min_{th}** and **max_{th}** helps to determine the average queue (**Q_{avg}**) length and packet dropping probability (**P_b**).

The approach of ERED [10] is to use current queue length (**q**) to evaluate average queue length **$Q_{avg}(n)$** based on different conditions. The lag parameters (**α** and **β)** are the used to make average queue length more adaptive. **q(n)** is the instantaneous queue and **W_q** is the weighted average defined for the queue. Then only dropping probability (**P_b**) is calculated.

$$Q_{avg}(n) = \begin{cases} \frac{1-W_q}{\alpha}.Q_{avg}(n-1) + w_q.q(n), & q < min_{th} \\ (1-w_q).Q_{avg}(n-1) + w_q.q(n), & min_{th} \le q \le max_{th} \\ (1-w_q).Q_{avg}(n-1) + \frac{w_q.q(n)}{\beta}, & q > max_{th} \end{cases} \quad (3.1)$$

$$P_b = \begin{cases} 0 & Qavg(n) < min_{th} \\ 1 & Qavg(n) > max_{th} \\ \frac{Q_{avg}(n)-min_{th}}{max_{th}-min_{th}}\, max_p & min_{th} \le Q_{avg}(n) \le max_{th} \end{cases} \quad (3.2)$$

Based on these two mathematical models, we can evaluate the mapping probability of a video frame as **Prob_New** based on dropping probability of video frames **Prob_X.**

$$\text{Prob\_New} = \text{Prob\_X} * \frac{Q_{avg}(n)-min_{th}}{max_{th}-min_{th}} \quad (3.3)$$

The flowchart of average queue length calculation and adaptive mapping of video frames into a queue is given below in Figure 3.4 and Figure 3.5.

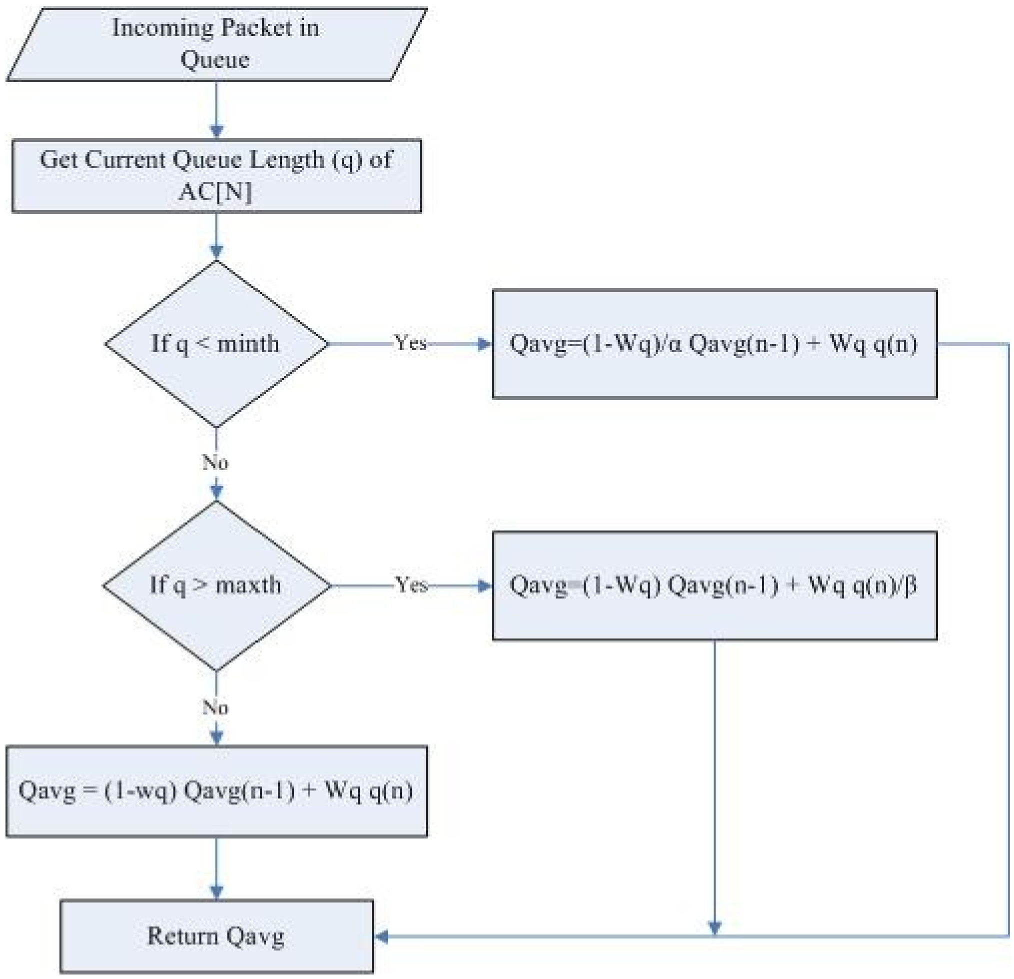

Figure 3.3: Calculation of Average Queue Length (Qavg)

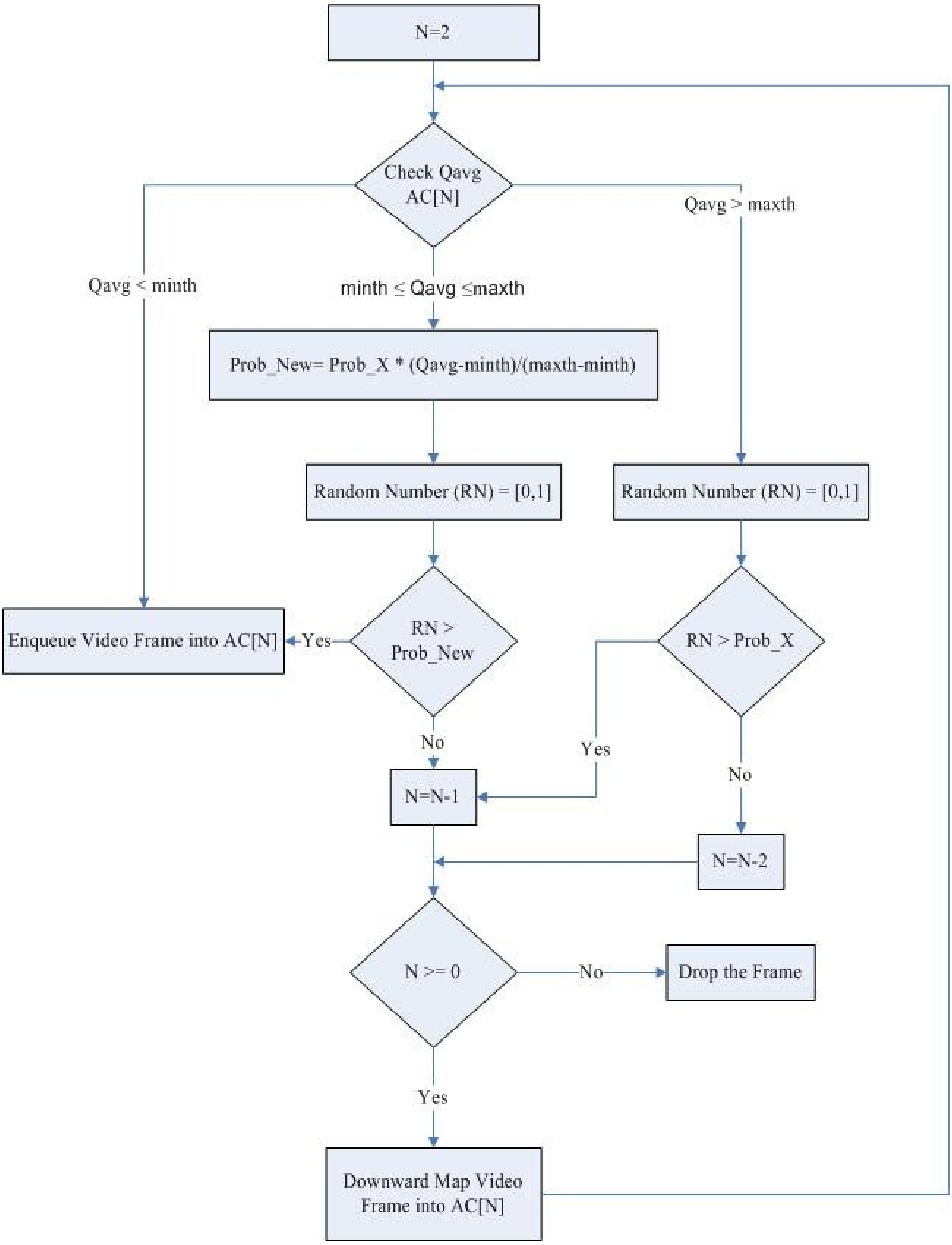

Figure 3.4: Adaptive Mapping of Video Frames into Queue

3.3.2 Congestion Adaptive Routing Algorithm

The prime idea behind this algorithm is to use MAC layer information (congestion level) to design adaptive routing protocol in network layer. Enhancement to AODV routing protocol will be done that will help video streaming to be implemented in a multi hop scenario. The different steps carried out in this algorithm are discussed below.

- Node Categorization
- Route Discovery
- Route Maintenance

Node Categorization

Node categorization is the main idea which is imposed in of our design. It is performed in MAC layer. It is a weight given to every intermediate node based on its congestion level. The 802.11e MAC is provided with four AC queues. We defined congestion level based on the priority of an AC queues [27].

Access Category	**Traffic Type**	**Congestion Level (weight)**
AC[3]	Voice	3
AC[2]	Video Frames	2
AC[1]	Best Effort	1
AC[0]	Background	0

Table 3.1: Congestion Level Based on Access Category [27]

From the above table we are going to show that if the queue of AC [3] is above the threshold its congestion level is high based on its priority. The node categorization algorithm is developed as follows.

If (AC [3] > maxthreshold) then weight = 3

Else if (AC [2] > maxthreshold) then weight = 2

Else if (AC [1] > maxthreshold then weight = 1

Else weight = 0

Using this classification, a node with no traffic or with delay insensitive traffic is considered more flexible to receive more traffic than a node with delay sensitive traffic, which is likely to be busy for a prolonged time.

Route Discovery

The route discovery of this algorithm is based on RREQ and RREP broadcasts. The source node seeking a route to a destination broadcasts a RREQ to neighbor nodes. When intermediate node receives RREQ, it then obtains the congestion level (weight) of a node based on node categorization algorithm from MAC layer. After determining the weight of each intermediate node, RREP message is forwarded to the source.

Apart from hop count information in RREP, we propose to pass the congestion level (weight) information to the source node. The resulting hop count is represented by the expression given below.

hop_count = actual_hopecount + total_weight * n

total_weight = $\sum_{i=1}^{n} weight_i$

Where **total_weight** is the total congestion level of all intermediate nodes and **n** is kept as a constant proportional to the average hop count of the network. On receiving RREP from multiple directions, the route with the smallest **hop_count** is chosen.

Route Maintenance

The congestion level of every intermediate node is updated every time when there is change in traffic type and, the congestion level is periodically propagated to neighbors in the HELLO messages. The MAC layer will be responsible to update the network layer whenever there is a change in traffic and generates an interrupt. This interrupt is directly provided to network layer and updated weight is passed to source node in order to update the **hop_count**. Based on the updated **hop_count** new route is selected.

3.4 Simulation Network Model

The simulation network model is designed to implement our cross layer framework for efficient MPEG-4 video streaming over IEEE 802.11e EDCA networks. This model will be used to observe the performance of proposed cross layer framework. The simulation will be carried out using NS2 integrated with EvalVid [22]. The simulation network model of this research consists of simulation topology, simulation parameters and simulation process.

3.4.1 Simulation Topology

In our simulation, we use network topology from [7]. The various parameters used in this topology are tabulated below.

Parameter	**Value**
Simulation Environment	NS2
Area Size	500 x 500 m^2
Node Placement	Random
Number of Mobile Nodes	20
No of Source Destination Pairs	10
Mobility Model	Random Waypoint Model
Pause Time	1 second
Speed of Mobile Node	(0,1, 2, 3, 8, 10) m/s
No of Scenarios	10
Link Bandwidth	2 Mbps
Routing Protocol	AODV
Dropping Probability of I frame (P_I)	0
Dropping Probability of P frame (P_P)	0.6
Dropping Probability of B frame (P_B)	0.9
Maximum Queue Length	50 Packets
Maximum Queue Threshold (max_{th})	40 Packets
Minimum Queue Threshold (min_{th})	10 Packets

Weighted Average (W_q)	0.1
Video Source	Foreman YUV QCIF (176x 144) (foreman_qcif.yuv)
No of Video Frames	1200 Frames
No of Video Packets	1977 Packets
Group of Picture (GOP)	9
Frame Sequence in GOP	I B B P B B P B B
Simulation Time	200 Seconds
Access Category for Voice (AC_VO)	AC[3]
Access Category for Video (AC_VI)	AC[2]
Access Category for Best Effort (AC_BE)	AC[1]
Access Category for Background (AC_BK)	AC[0]

Table 3.2 : Simulation Parameters for Verification

The parameter values regarding dropping probability of video frames (I/P/B), maximum and minimum queue threshold, maximum queue length, video source, video frames, simulation time, communication type, selection of video server & client and input traffic in AC queues are considered from [41]. The lag parameters **α**, **β** and the value of weighted average ($\mathbf{W_q}$) are taken from [10].

Simulation Scenario

The simulation scenarios are created in order to evaluate our system in different loading conditions of Access Categories of IEEE 802.11e EDCA. We will consider three traffic flows. The three traffic flows along with its descriptions are tabulated below.

Table 3.3: Simulation Scenarios

Traffic Flows	Description	Remarks
1	Video Flow	Video Flow = Foreman YUV QCIF (176 x 144) Packet Size = 1024 bytes (no stuffing bit used) Data Rate = 1200 and 2400 kbps
2	Voice Flow	Voice Flow = CBR Packet size = 512 bytes Data Rate = 64 kbps
3	Best Effort Flow	Best Effort Flow = CBR Packet Size = 256 bytes Data Rate = 128 kbps

The combination of these traffic flows will help us to study the impact of different flows in video streaming. The scenarios from the above traffic flows will be video only flows, (voice + video) flows and (voice + video + best effort) flows. For (voice + video) flows, both the traffics are passed simultaneously where voice flow is allocated in its own access category. In the case of (voice + video + best effort) flows, all three traffics are passed simultaneously where voice and best effort flows are allocated in its own access category.

3.4.2 Simulation Process

The following list shows the Generic Script Structure that will be used for simulation setup in NS2.

- Accept Input parameters from command line.
- Create Topology
- Define wireless mobile node
- Define node movement model
- Create Routing agents
- Create Protocol Agents

- Schedule Events
- Post processing procedure
- Start the simulation

3.5 Performance Parameters

The performance evaluation of this research will be done with EvalVid. It is a complete framework and a toolkit for a unified assessment of the quality of video transmission. To show cross layer framework for efficient MPEG-4 video streaming, the main performance parameters will Packet Drop Rate, and queue space utilization. These parameters will help to compare our research with existing work that was done before.

Packet Delivery Ratio (PDR)

The ratio of number of video packets received by the destination to the number of video packets provided by the source is known as Packet Delivery Ratio. Mathematically it is represented as

$$\text{Packet Delivery Ratio (\%)} = \frac{\text{No of Packets Received}}{\text{No of Packets Sent}} \text{ x } 100$$

Throughput

The measure of total number of bits that can be transmitted per unit time is known as throughput. It is generally expressed as number of bits per second and is measured in kbps or mbps. Mathematically it is represented as

$$\text{Throughput (bps)} = \frac{\text{No of Received Packets x 8}}{\text{Total Time Taken}}$$

Average End to End Delay

Average of the total time to deliver packet from source to destination is known as Average End to End Delay.

End to End Delay (ms) = Packet Receive Time – Packet Send Time

$$\text{Average End to End Delay} = \frac{\sum(\text{Packet receive Time} - \text{Packet Sent Time})}{\text{Total Number of Packet Received}}$$

CHAPTER 4

RESULTS AND DISCUSSIONS

4.1 Overview

The detail analysis and evaluation of proposed cross layer framework is done in this chapter. The cross layer between application and MAC is verified with the three mapping techniques. The three mapping techniques include EDCA mapping, static mapping and our proposed dynamic mapping. Three mapping approaches are tabulated below.

Table 4.1: Simulation Cases

Case	Approach	Remarks
1.	EDCA (MacKenzie et al., 2009)	All Video Frames in AC[2]
2.	Static (Soni et al, 2008)	I = AC[2], P = AC[1],B = AC[0]
3.	Proposed Approach (Dynamic)	I,P and B Frames are allocated from AC[2] to AC[0] based on its priority and traffic load in AC queues.

The cross layer framework between MAC and network is verified in order to enhance the performance of video streaming. This framework deals with the creation of MAC aware network layer in order to regulate data rate based on the congestion information acquired from MAC. This MAC aware network is also verified in the above mentioned mapping techniques. It helps to study the role of MAC aware network layer in mapping MPEG-4 video frames. The performance parameters used for the evaluation are PDR, average end to end delay, throughput.

4.2 Analysis of Result

The analysis of results are done in three different aspects. They are effect of node mobility ,effect of traffic load, and effect of MAC aware network layer. The effect of node mobility is analyzed by varying the speed of mobile node from 0 m/s to 10 m/s. The effect of traffic load is studied by varying different traffic. The impact of other traffics such as voice and best effort is studied by allowing simultaneous voice and best effort data along with video traffic. The effect of MAC aware network layer is also verified with video data rate of 1200 kbps. The setting of video data rate is entirely based on congestion information acquired from MAC. The detail discussion of each aspect is done below.

4.2.1 Effect of Node Mobility

The effect of node mobility is studied by varying velocity of node from 0 m/s to 10 m/s (0,1,2,3,8 and 10) m/s. In this case, the video data rate is fixed at 1200 kbps per total connection. In order to study this node mobility effect the video only traffic is considered. The detailed description along with the graphs are shown below for all the traffic scenarios with the video data rate at 1200 and 2400kbps.

a) Data Rate= 1200 kbps Per Total Connection (video Only)

In this case only video traffic is considered with a data rate as 1200 kbps per total connection.

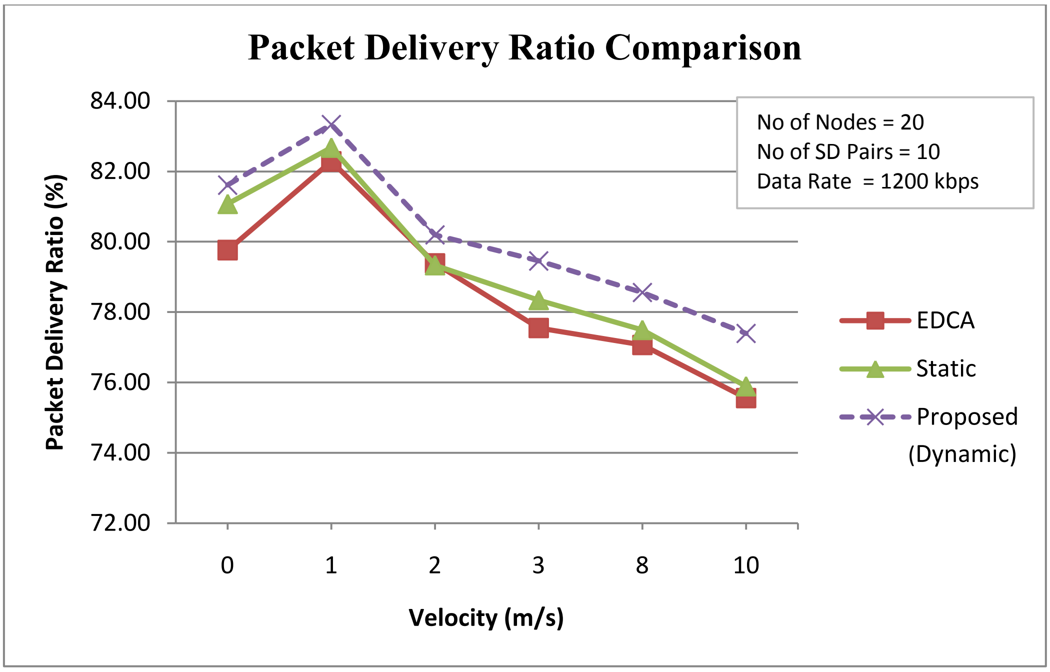

Figure 4.1: Packet Delivery Ratio Comparison (video only)

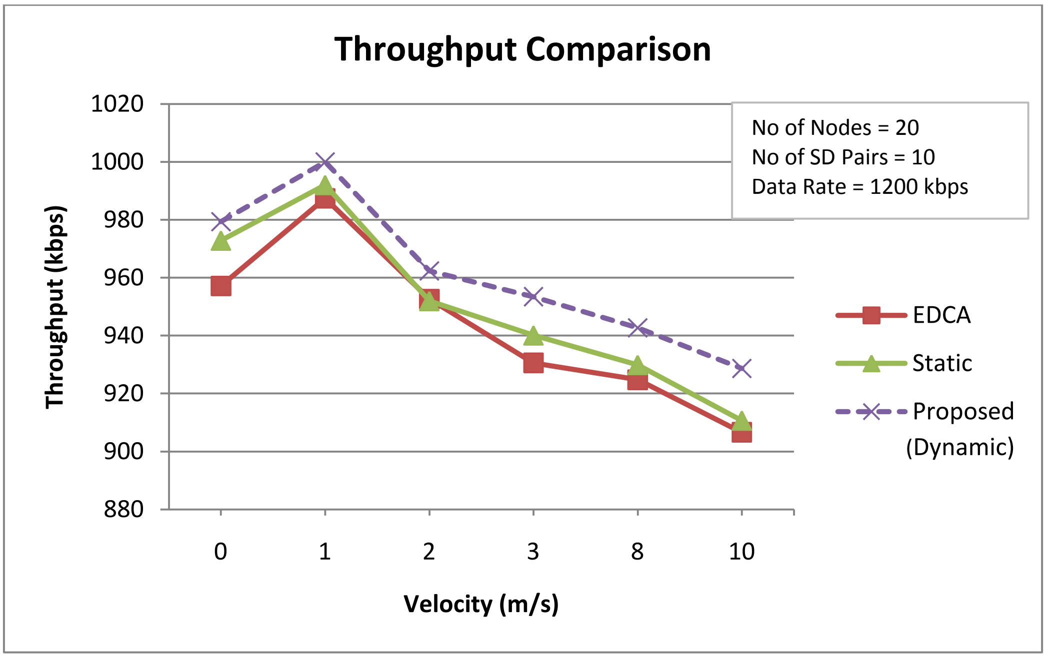

Figure 4.2: Throughput Comparison (video only)

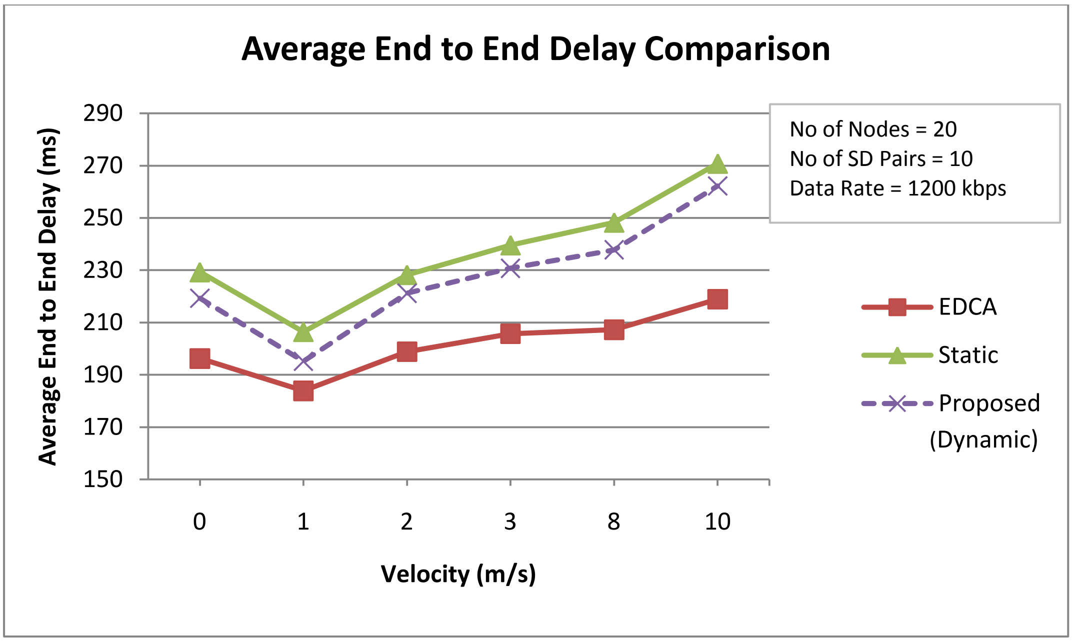

Figure 4.3: Average End to End Delay Comparison (video only)

Discussion

The increase in node mobility leads to the increase in link breakage which in turns defines more route failures. The increase in route failures requires more re route discoveries which creates more packet to drop decreasing the PDR and throughput. The impact of more re route discovery makes more packets to wait in a queue increasing average end to end delay. The PDR and throughput of the proposed approach is improved than EDCA and static mapping. The average end to end delay is more than that of EDCA but it is less than that of static.

a) Data Rate= 2400 kbps Per Total Connection (video Only)

In this case only video traffic is considered with a data rate as 2400 kbps per total connection.

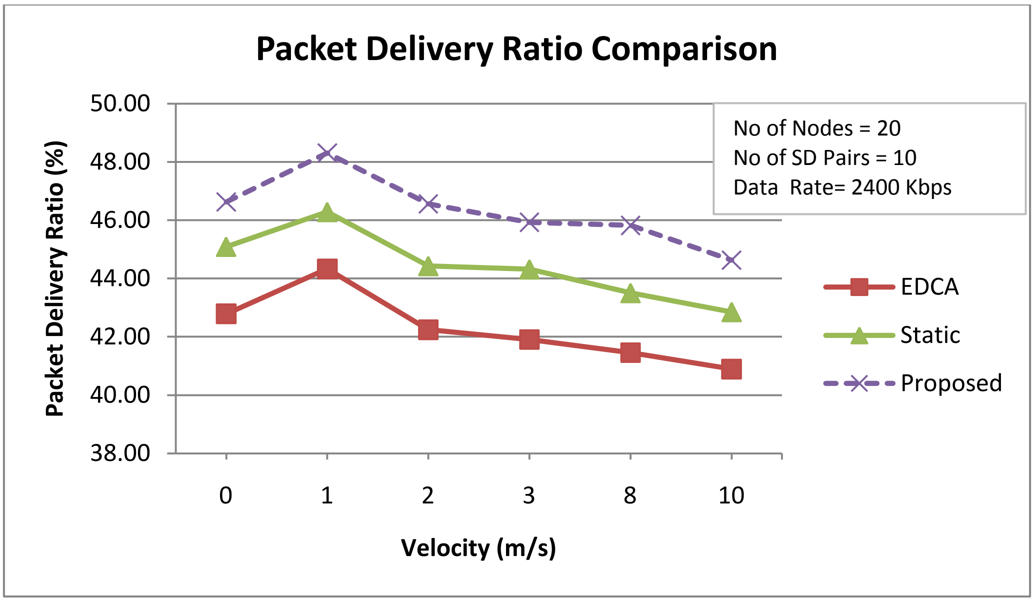

Figure 4.4: Packet Delivery Ratio Comparison (video only)

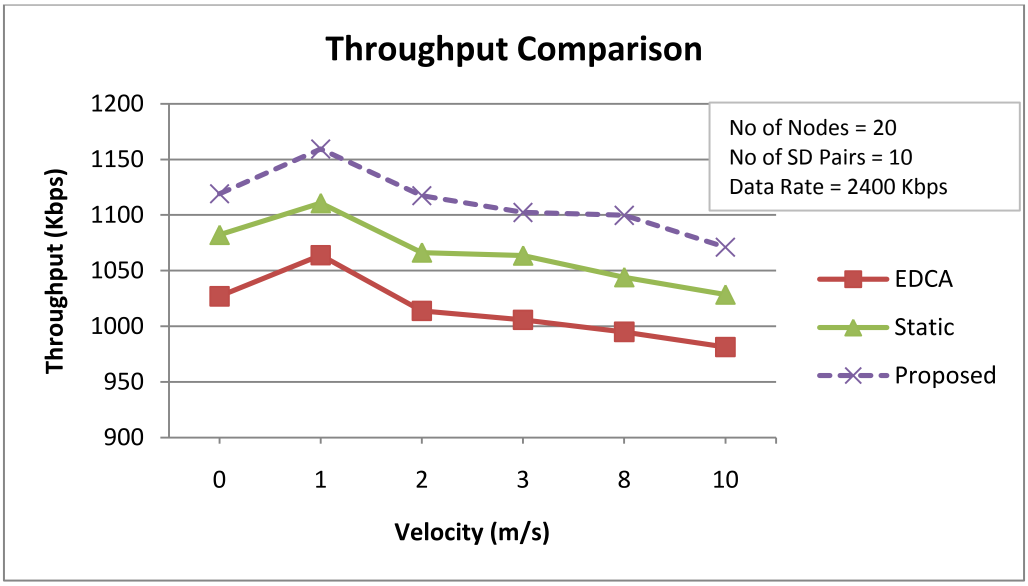

Figure 4.5: Throughput Comparison (video only)

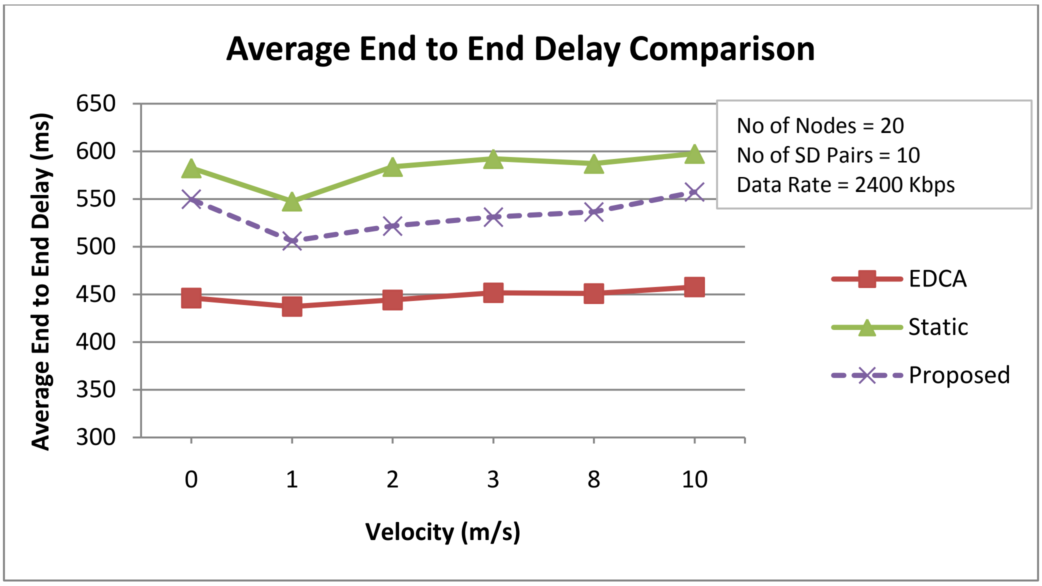

Figure 4.6: Average End to End Delay Comparison (video only)

Discussion

The increase in data rate makes more packet to arrive faster creating queue to overflow in short time span. This create packets to drop decreasing PDR. The rate of increased data rate leads throughput to increase rapidly. This enforces arriving packets to wait in a queue leading average end to end delay to increase. The quality of video degrades as packet loss increases. Hence, the increased data rate degrades the quality of video. The PDR and throughput of the proposed system is improved than other mapping approaches. But, the average end to end delay is quite higher than that of EDCA and lower than that of static approach.

4.2.2 Impact of Other Traffic Load in Video Streaming

a) Data Rate = 1200 kbps and 2400 kbps Per Total Connection (voice + video)

In this case (voice + video) traffic is considered. The Data Rate of video traffic is kept as 1200 kbps and 2400 kbps. The size of voice data is taken as 512 bytes with the data rate of 64 kbps.

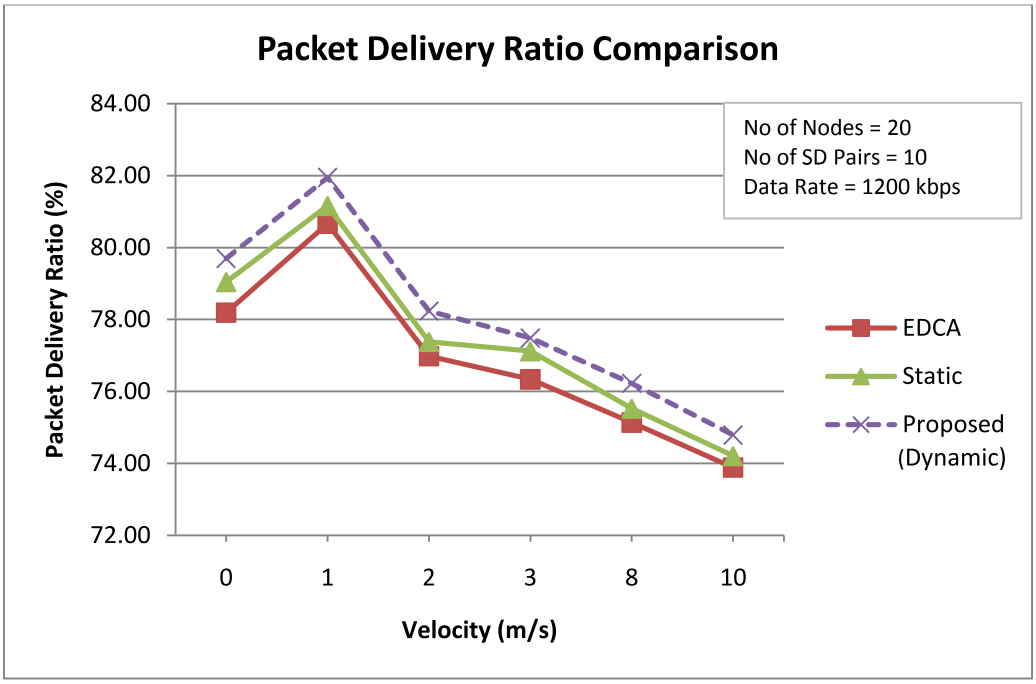

Figure 4.7: Packet Delivery Ratio Comparison (voice + video) in 1200 kbps

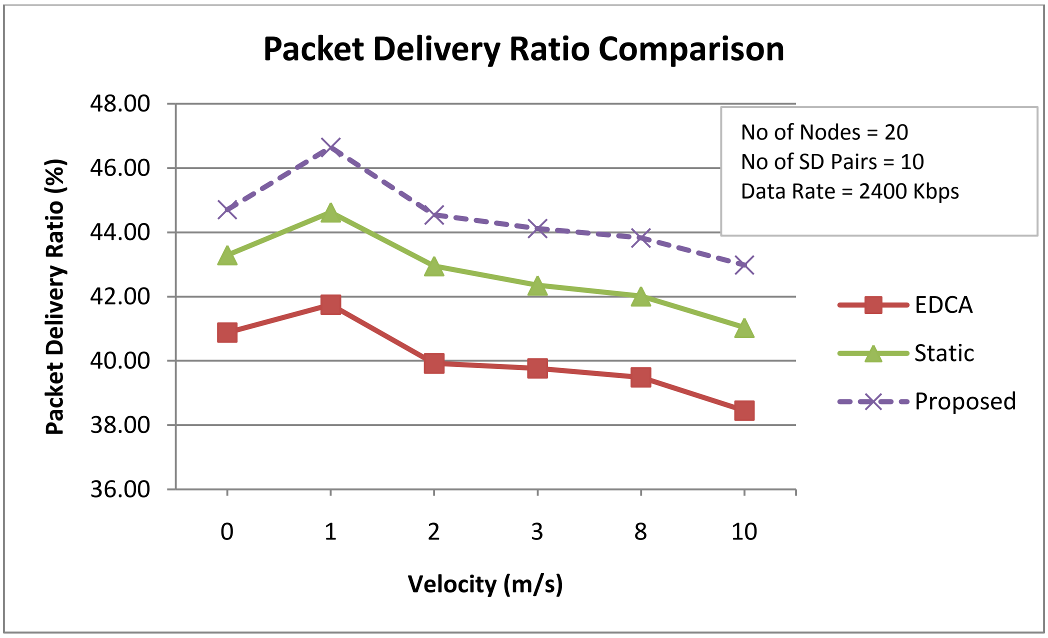

Figure 4.8: Packet Delivery Ratio Comparison (voice + video) in 2400 kbps

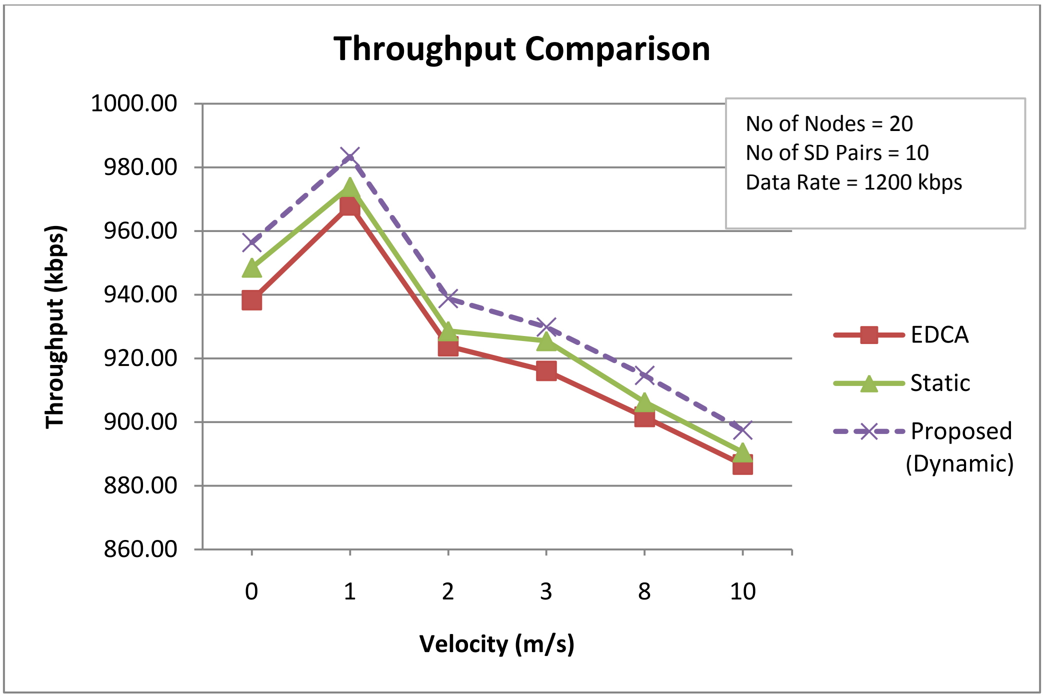

Figure 4.9: Throughput Comparison (voice + video) in 1200 kbps

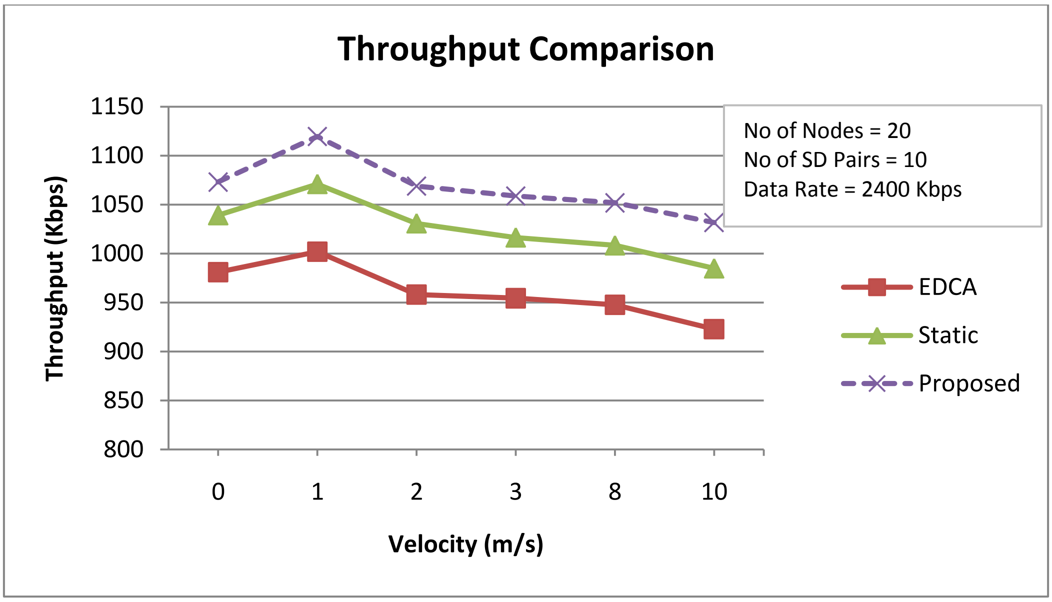

Figure 4.10: Throughput Comparison (voice + video) in 2400 kbps

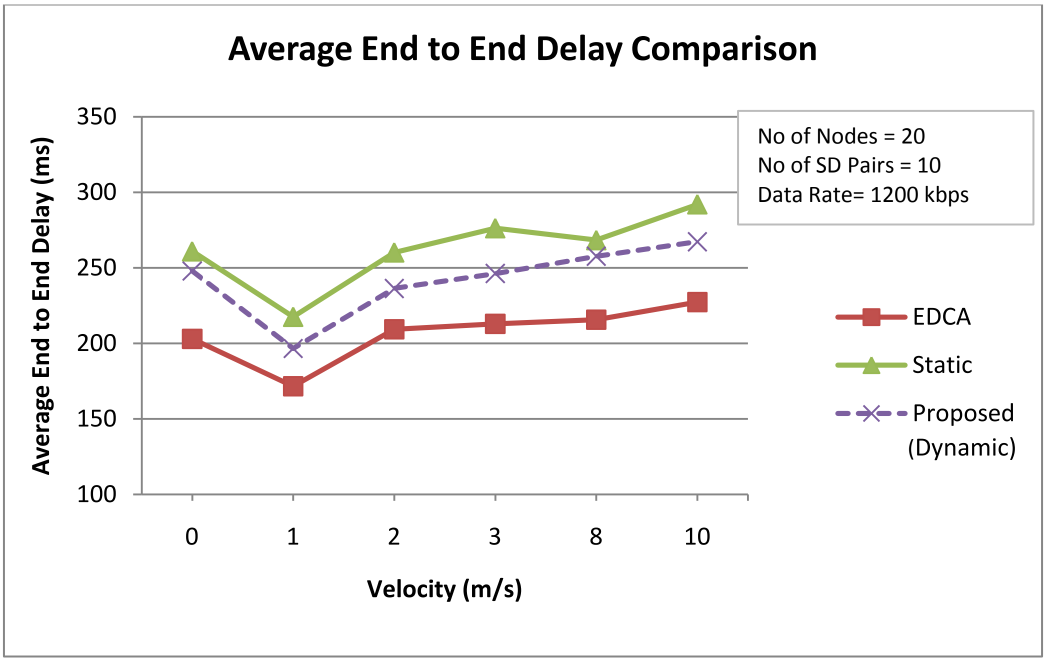

Figure 4.11: Average End to End Delay Comparison (voice + video) in 1200 kbps

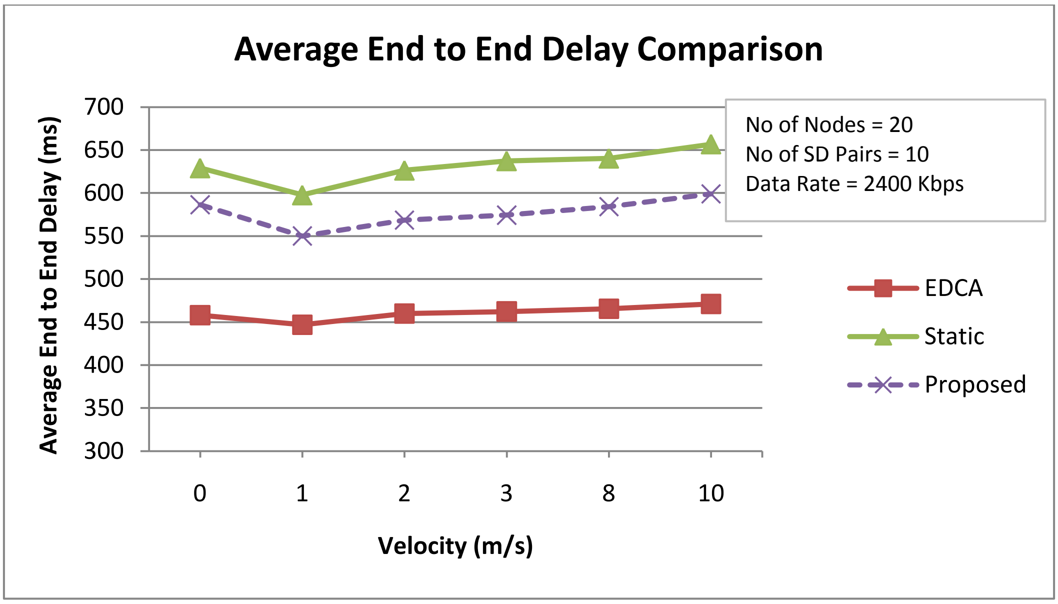

Figure 4.12: Average End to End Delay Comparison (voice + video) in 2400 kbps

Discussion

The increase in node mobility degrades the performance of (voice + video) flows. The increase in speed of mobile nodes create a lot of link failure providing direct impact on routing. This increases more re route discovery which in turn create more packets to be dropped causing PDR and throughput to be degraded. The more re route discovery enforce packet to wait in a queue increasing average end to end delay. The decrease in PDR leads to the loss of video frames degrading the quality of video. The PDR and throughput of the proposed system is improved. The average end to end delay is higher than that of EDCA but better than static approach.

The increase in data rate makes more packet to arrive faster creating queue to overflow in short time span. This create packets to drop decreasing PDR. The rate of increased data rate leads throughput to increase rapidly. This enforces arriving packets to wait in a queue leading average end to end delay to increase. The quality of video degrades as packet loss increases. Hence, the increased data rate degrades the quality of video.

b) Data Rate = 1200 kbps and 2400kbps Per Total Connection (voice + video + best effort)

In this case, (voice + video + best effort) traffic is considered. The offered load of video traffic is kept as 1200kbps. The size of voice data is taken as 512 bytes with the data rate of 64 kbps. The size of best effort data is taken as 256 bytes with the data rate of 128 kbps.

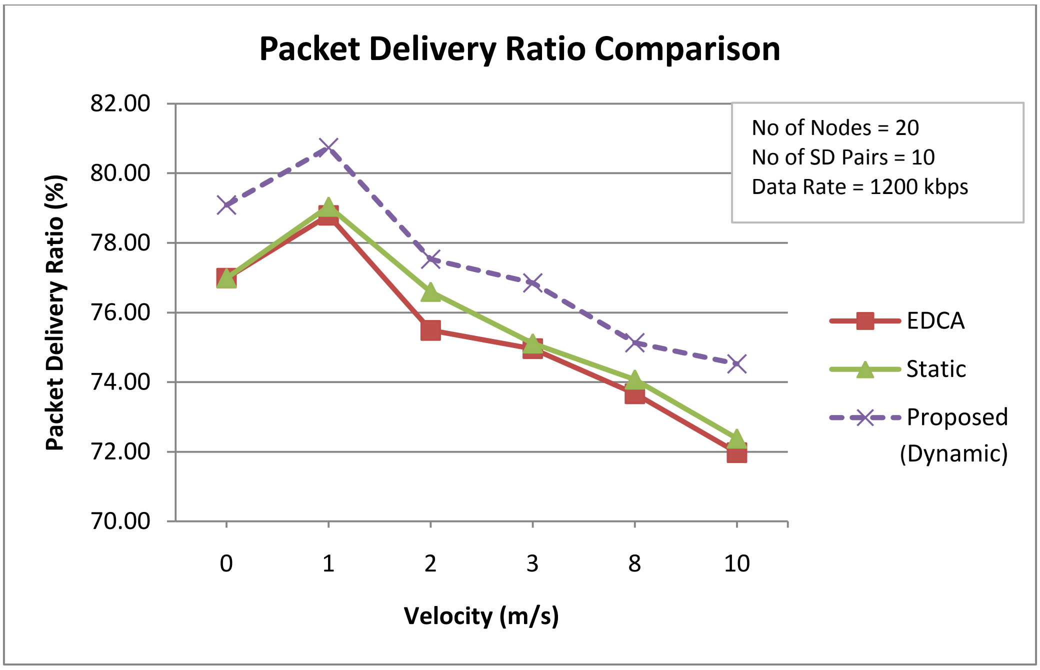

Figure 4.13: Packet Delivery Ratio Comparison in 1200 kbps

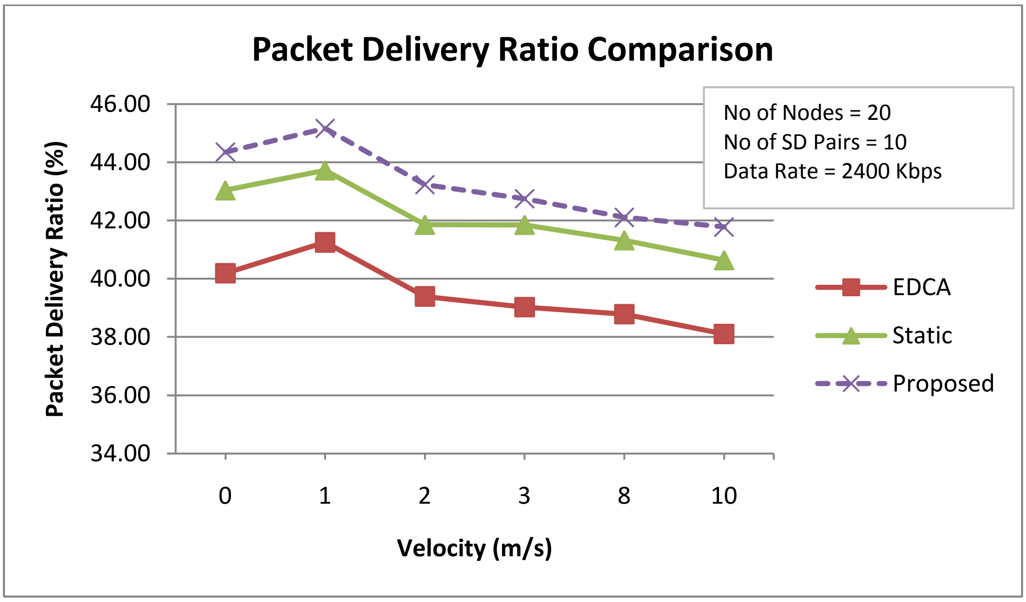

Figure 4.14: Packet Delivery Ratio Comparison in 2400 kbps

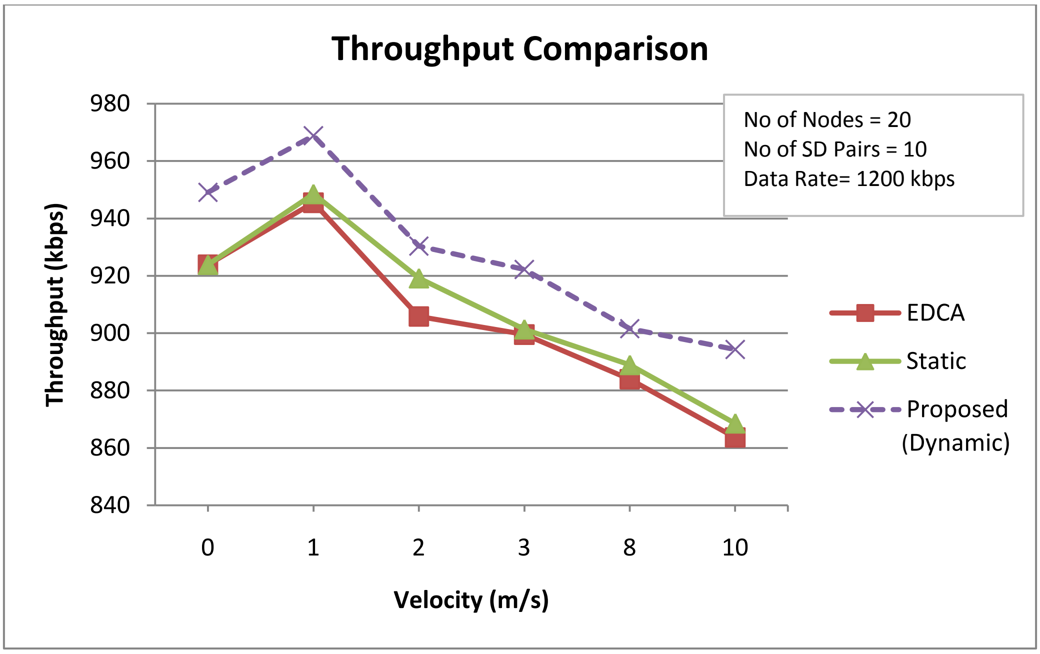

Figure 4.15: Throughput Comparison in 1200 kbps

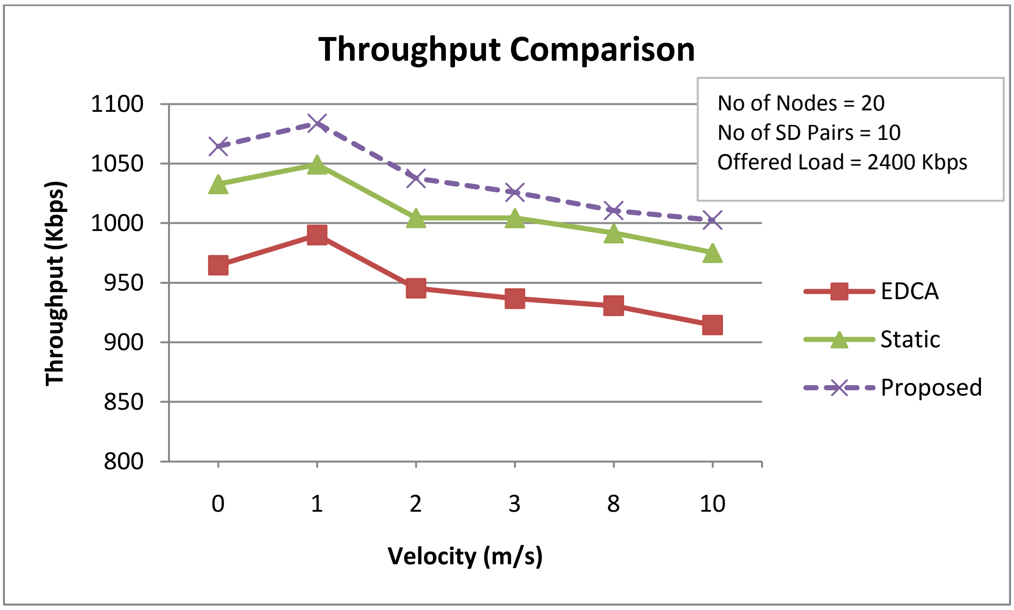

Figure 4.16: Throughput Comparison in 2400 kbps

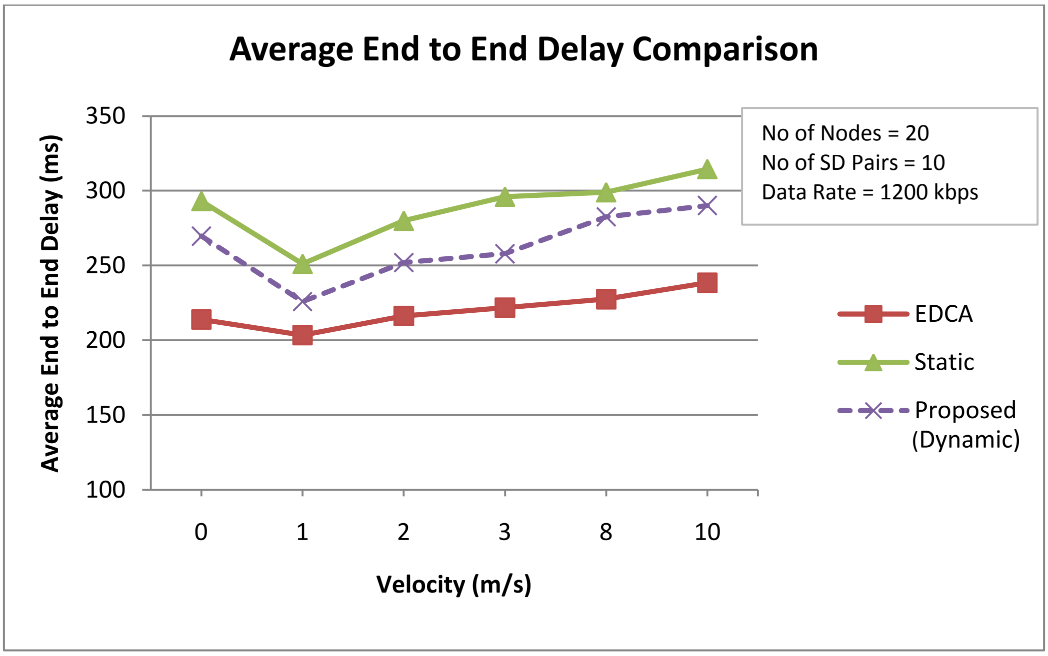

Figure 4.17: Average End to End Delay Comparison in 1200 kbps

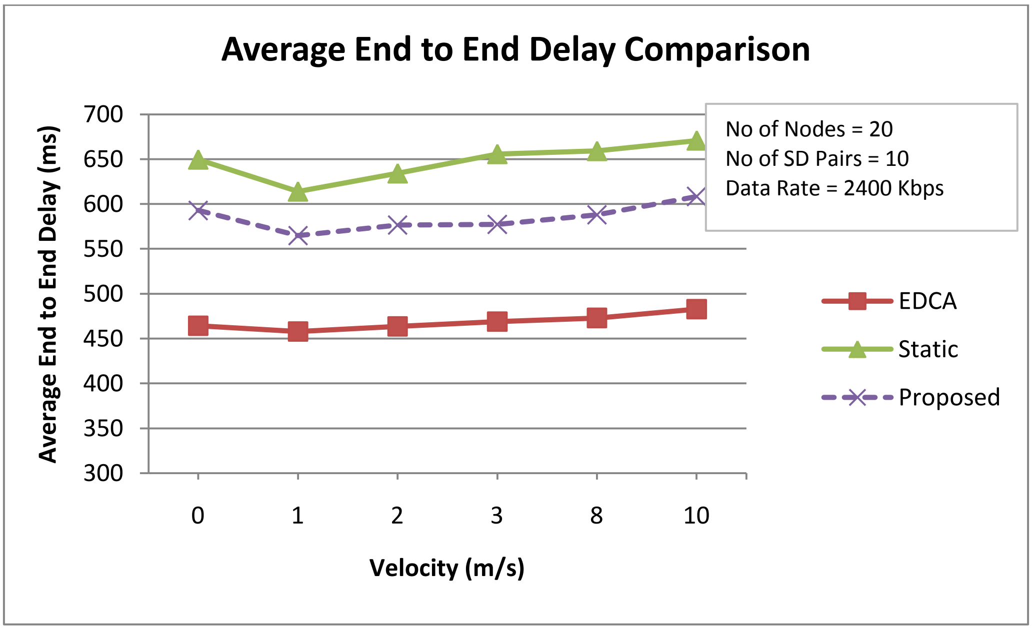

Figure 4.18: Average End to End Delay Comparison in 2400 kbps

Discussion

The performance of (voice + video + best effort) flows are decreased with the increase in velocity. This increase in velocity ensures more link failures providing more re route to be discovered. It creates PDR and throughput to decrease. Due to the route breakage, the re route discovery process enforce packets to wait in a queue increasing average end to end delay. The drop of more packets lead to the loss of video frames.

The increase in data rate makes more packet to arrive faster creating queue to overflow in short time span. This create packets to drop decreasing PDR. The rate of increased data rate leads throughput to increase rapidly. This enforces arriving packets to wait in a queue leading average end to end delay to increase. The quality of video degrades as packet loss increases. Hence, the increased data rate degrades the quality of video.

4.2.3 Effect of MAC aware Network

The MAC aware network is verified by implementing the mathematical function of congestion aware routing stated above in literature review. In this case only video data is taken into consideration. The video data rate is set to 1200 kbps. The mobility of node is taken from 0 m/s to 10 m/s.

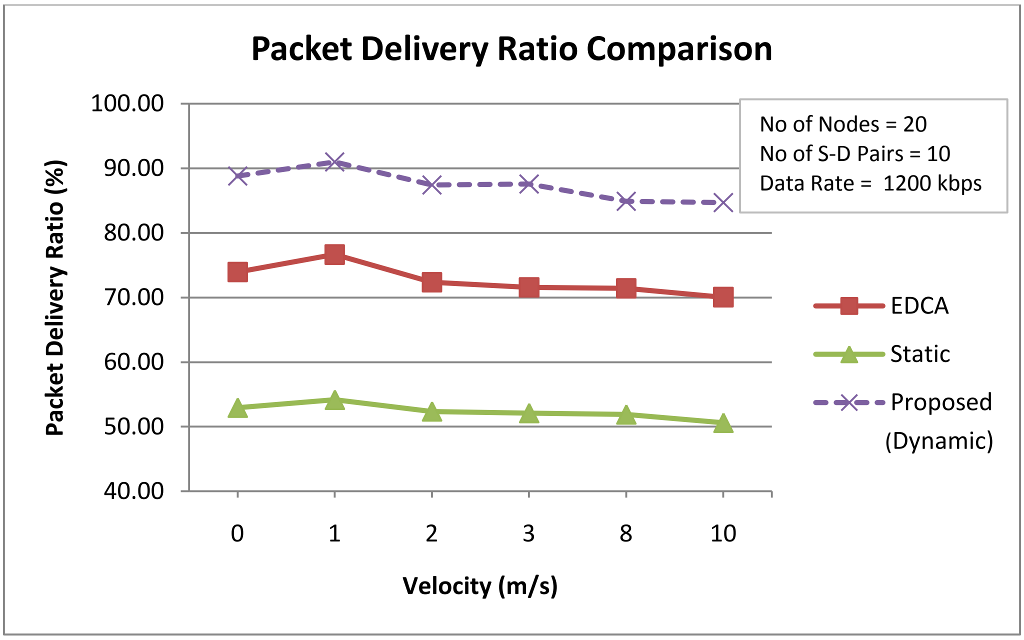

Figure 4.19: Packet Delivery Ratio Comparison

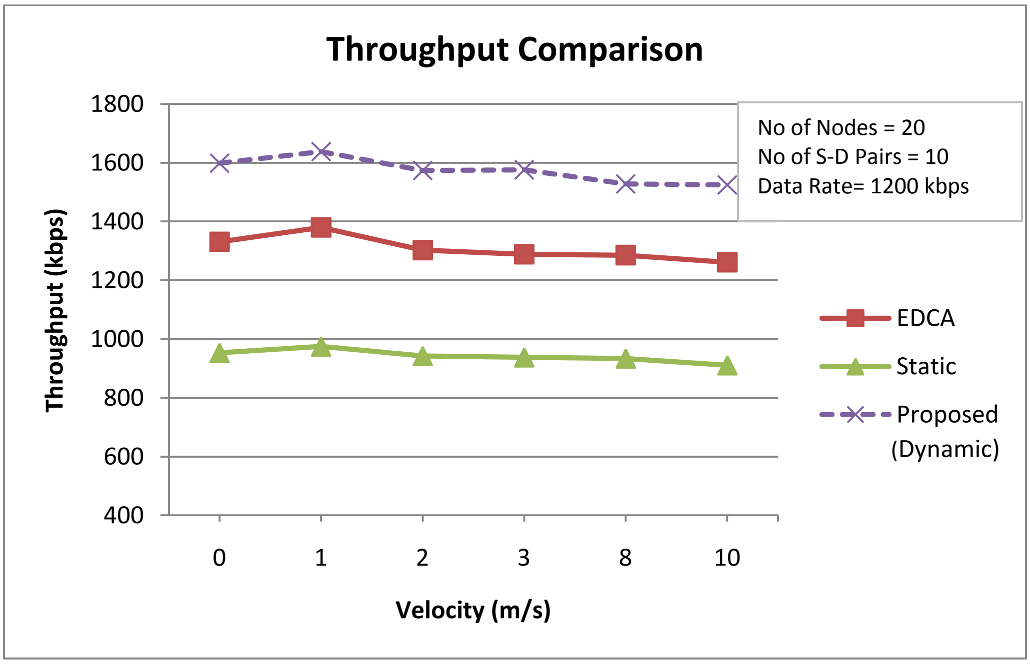

Figure 4.20: Throughput Comparison

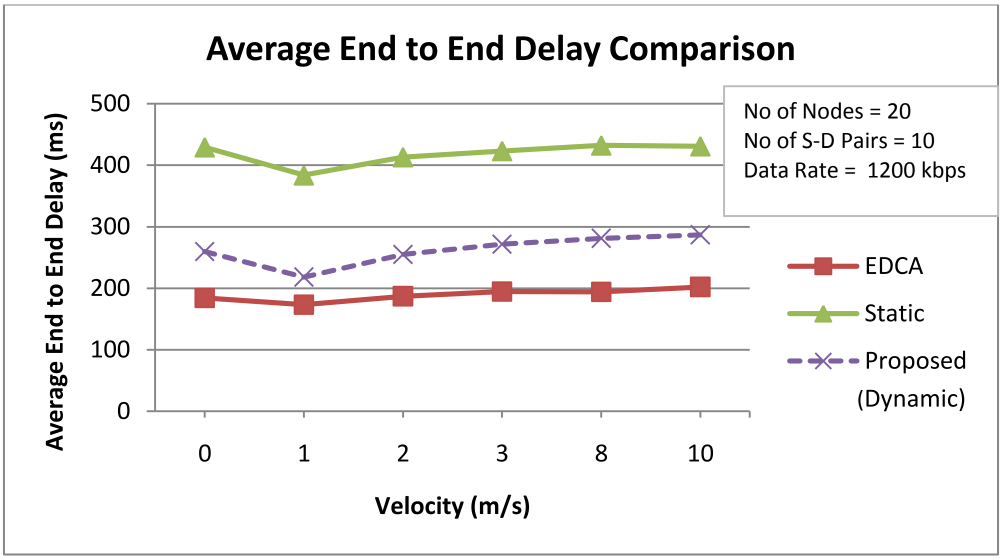

Figure 4.21: Average End to End Delay Comparison

Discussion

The above graphs show the impact of MAC aware network in mapping MPEG-4 video frames. The increase in mobility lead toward more link breakages. It enforces more re route discoveries. This creates more packet drop which lead towards decrement of PDR and throughput. The re route discovery process make packets to wait in a queue increasing average end to end delay. Due to the increase in packet drop, the video quality is degraded. The MAC aware network feature enhances the performance of dynamic mapping algorithm. The PDR, throughput and PSNR of the proposed system is improved. The average end to end delay is quite high than EDCA approach but less than that of static approach.

4.2.4 Summary

The analysis of our cross layer framework is entirely based on three mapping techniques which includes EDCA, static and our proposed dynamic mapping. It is done in three different aspects referred as effect of node mobility, effect of offered traffic load and effect of MAC aware transport. In each aspect three traffic scenarios are considered including video only, (voice + video) and (voice + video + best effort).

In the case of node mobility, the speed of mobile node is varied from 0 m/s to 1 m/s. The data rate is kept constant i.e. 1200 kbps or 2400 kbps per total connection. The increase in node mobility increases the rate of link failures. This link failure will enforce more re route to be discovered. It creates more packet to get dropped enabling PDR and throughput to decrease. The re route discovery makes packet to wait in a queue increasing average end to end delay. The loss of packet creates quality of video to be degraded. The PDR and throughput of the proposed system is improved than other mapping approaches. The average end to end delay of proposed system is higher than that of EDCA and lower than that of static. During the static velocity (0 m/s) , nodes will not move from its position. If the nodes are widely scattered then it might enforce packet to drop because of no route between source and destination. And, there is also no possibility that nodes will come nearer and route can be re discovered. This makes the performance of 0 m/s even worst than 1 m/s.

In the case data rate, the data rate of video traffic is evaluated with other traffic sources such as voice and best effort to study the impact of traffic loads. The increase in data rate makes more packet to arrive faster creating queue to overflow in short time span. This create packets to drop decreasing PDR. The rate of increased data leads throughput to increase rapidly. All the packets arriving at an increased rate can not be accommodated in a queue. This enforces arriving packets to wait in a queue leading average end to end delay to increase. The quality of video degrades as packet loss increases. Hence, the increased data rate degrades the quality of video. The PDR and throughput of the proposed system is improved than other mapping approaches. But, the average end to end delay is quite higher than that of EDCA and lower than that of static approach.

In order to verify MAC aware network, the speed of mobile node is varied from 0 m/s to 10 m/s. When the mapping techniques are applied, the PDR and throughput of the proposed system is improved than other approaches. The average end to end delay is higher than that of EDCA and lower than that of static approach. This MAC aware network enhances the performance of video streaming.

The impact study of other traffics is also carried out. It shows that the performance of video only flows is better than other flows. Finally it is found that the proposed dynamic approach is better in terms of PDR and throughput. The average end to end delay is higher than that of EDCA but lower than that of static. The addition of MAC aware network further enhanced the capability of dynamic mapping. Hence, the cost of delay is beared for the shake of quality. This states that our proposed system is only suitable for non real time video streaming applications.

CHAPTER 5
CONCLUSIONS AND RECOMMENDATIONS

5.1 Conclusions

Multimedia services in wireless ad hoc environment is gaining more popularity during the recent years. Many challenging problems exists in order to satisfy the QoS demand of an end users. Cross layer architecture is one of the solution to address these issues. The cross layer design is an innovative idea that provides knowledge sharing environment between layers to achieve the required QoS and to maintain the highest possible adaptivity among layers.

The main idea behind this research is to provide the cross layer framework between application and MAC and also between MAC and network layer. The cross layer between application and MAC aids to provide dynamic mapping of MPEG-4 video frames based on priority of video and network traffic load of an AC queue. Similarly, the cross layer between MAC and network helps to design network aware MAC. The performance parameters such as throughput, average end to end delay and PDR are considered to evaluate the system. The proposed algorithms are compared with EDCA and static mapping approaches. The PDR and throughput of the proposed system is improved than EDCA and static mapping approaches. The average end to end delay is higher than that of EDCA but less than that of static approach. The cost of delay is beared for the shake of video quality. It can also be concluded that our proposed system is not suited for real time video streaming but is best for non real time video streaming applications.

5.2 Future Recommendations

The future recommendations will help other researcher interested in the field of cross layer design to provide more better and optimized solutions for the video streaming. The key recommendations are enlisted below.

- The QoS based cross layer routing can be incorporated to make video streaming more efficient. This might help to enhance the performance of routing algorithm and can also aid to make video streaming more real time by minimizing the average end to end delay.

- The MAC aware network can be made more adaptive by incorporating physical layer parameters such as Signal to Noise Ratio(SNR).

- The dynamic mapping algorithm can be further enhanced by incorporating real time audio instead of constant bit rate traffic. This can help to study the impact of real time audio traffic in video streaming.

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