A PRIORITY AWARE BUFFER MANAGEMENT SCHEME FOR H.264 VIDEO OVER MANET

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DEDICATION

To God Almighty and my loving family.
Video is bandwidth hungry and delay-sensitive and the compressed video bit stream is very sensitive to packet losses. The transmission of compressed video over the mobile ad hoc networks (MANETs) is very challenging due to the limited bandwidth resources and error-prone channels. It is therefore important to reduce the packet loss by intelligently handling the video traffic.

In this thesis, we use a routing protocol, known as the Ad Hoc On Demand Multipath Distance Vector Routing Protocol with Dynamic Path Update (AOMDV-DPU) which is an extension of AOMDV. AOMDV-DPU emphasizes on maintaining the secondary paths to the destination along with the primary route so that a back-up route can be used in case of the failure of the primary route.

We propose a buffer management scheme, called Smart Queuing technique that intelligently sorts the H.264/AVC compressed video packets in the buffer according to their priority. As a result, the higher priority packets do not have to wait longer or expire in the queue to get the channel access. The purpose of the scheme is to serve the higher priority packets better than the lower priority packets thereby increasing the goodput of the network. The routing protocol is integrated with the Smart Queue to serve the higher priority packets and hence increase the quality of the video transmission.
# TABLE OF CONTENTS

<table>
<thead>
<tr>
<th>ABSTRACT</th>
<th>LIST OF TABLES</th>
<th>LIST OF FIGURES</th>
<th>CHAPTER</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td>1</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>INTRODUCTION</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1.1 Motivation of Thesis</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1.2 Research Contributions</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>1.3 Thesis Outline</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>INTRODUCTION TO H.264 VIDEO CODING STANDARD</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2.1 Introduction</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2.2 VCL</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2.3 Slices and Slice Groups</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>2.4 Flexible Macroblock Ordering (FMO)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>ROUTING PROTOCOLS</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3.1 Mobile Ad-hoc Networks (MANET)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3.1.1 Difficulties Faced in MANETs</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3.1.2 Routing in MANET</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3.2 Routing Protocols</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3.3 Ad-hoc On-Demand Distance Vector Routing (AODV)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3.4 Ad-hoc On-Demand Multipath Distance Vector Routing (AOMDV)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>3.5 Ad-hoc On-Demand Multipath Distance Vector Routing with Dynamic Path Upgrade (AOMDV-DPU)</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>4</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>BUFFER MANAGEMENT TECHNIQUES</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>4.1 Related Schemes</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>4.2 Prioritization of H.264 Video Packets</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>4.3 Video Packet Formation in NS-2</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>4.4 CMSE Computation/Prediction of H.264 Video Slices</td>
</tr>
</tbody>
</table>
4.5 Basic NS-2 Queuing Scheme ..............................................................14
  4.5.1 Definitions ..............................................................................14
  4.5.2 Interface Queue in NS-2 ..........................................................15
  4.5.3 Event Scheduler in NS-2 ...........................................................16
4.6 Buffer Overflow ...............................................................................17
4.7 Smart Queue Scheme for Video Traffic ...........................................18
  4.7.1 At the Enqueue .......................................................................18
    4.7.1.1 Enqueue the Packet ...........................................................19
    4.7.1.2 Pre-emptive Dropping of Packets .....................................19
  4.7.2 Sorting Algorithm ...................................................................20
    4.7.2.1 Frame Number Based Sorting ..........................................22
    4.7.2.2 Priority Based Sorting .......................................................22
    4.7.2.3 Time-stamp Based Sorting .................................................22
  4.7.3 Working of Sorting Technique .................................................23
  4.7.4 Adaptive Rate Control .............................................................23
  4.7.5 At the Dequeue .......................................................................24
4.8 IEEE 802.11e ...............................................................................25

5 RESULTS ............................................................................................27
  5.1 Network Parameters .................................................................27
  5.2 Video Traffic Parameters .............................................................27
  5.3 Average Packet Delivery Ratios of Different Schemes at Node Speed 5m/Sec ..........................................................28
  5.4 Average Packet Delivery Ratios of Different Schemes at Node Speed 10m/Sec: ..........................................................30
  5.5 Average Packet Delivery Ratios of Different Schemes at Node Speed 20m/sec ..........................................................31
  5.6 PSNR Plots ..................................................................................34
    5.6.1 Average PSNR of Different Schemes at Node Speed 5m/sec .....34
    5.6.2 Average PSNR of Different Schemes at Node Speed 10m/sec ....34
    5.6.3 Average PSNR of Different Schemes at Node Speed 20m/sec ....37
  5.7 Video Results ...............................................................................37
  5.8 Concept of Source Priority .........................................................41
    5.8.1 Results with Source Priority ................................................42
5.8.2 PDR and PSNR Plots with Source Priority for 2 Connections ...............42
5.8.3 PDR and PSNR Plots with Source Priority for 3 Connections ...............44
5.8.4 PDR and PSNR Plots with Source Priority for 4 Connections ...............44

6 CONCLUSION ............................................................................................................49

7 FUTURE WORK .........................................................................................................50

REFERENCES ........................................................................................................................51
## LIST OF TABLES

<table>
<thead>
<tr>
<th>Table</th>
<th>Title</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>Table 4.1</td>
<td>Contention Window values based on Packet Priority</td>
<td>26</td>
</tr>
<tr>
<td>Table 5.1</td>
<td>Network Parameters</td>
<td>27</td>
</tr>
<tr>
<td>Table 5.2</td>
<td>Video Traffic Parameters</td>
<td>27</td>
</tr>
<tr>
<td>Table 5.3</td>
<td>Different Schemes Used in the Simulation</td>
<td>28</td>
</tr>
<tr>
<td>Table 5.4</td>
<td>Combined Packet Priority with Source Priority</td>
<td>41</td>
</tr>
</tbody>
</table>
### LIST OF FIGURES

<table>
<thead>
<tr>
<th>Figure</th>
<th>Description</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>2.1</td>
<td>Structure of H.264/AVC Encoder</td>
<td>3</td>
</tr>
<tr>
<td>2.2</td>
<td>Division of picture into slices without the use of FMO</td>
<td>5</td>
</tr>
<tr>
<td>3.1</td>
<td>An example of MANET</td>
<td>7</td>
</tr>
<tr>
<td>3.2</td>
<td>Illustration of the Route discovery process in AODV</td>
<td>9</td>
</tr>
<tr>
<td>4.1</td>
<td>Basic queuing scheme in NS-2</td>
<td>15</td>
</tr>
<tr>
<td>4.2</td>
<td>Flowchart of the IFQ in NS-2</td>
<td>16</td>
</tr>
<tr>
<td>4.3</td>
<td>Event scheduler in NS-2</td>
<td>17</td>
</tr>
<tr>
<td>4.4</td>
<td>Flowchart of the FIFO queue</td>
<td>17</td>
</tr>
<tr>
<td>4.5</td>
<td>Flowchart of the smart queue technique</td>
<td>18</td>
</tr>
<tr>
<td>4.6</td>
<td>Flowchart of the en-queue in smart queue</td>
<td>19</td>
</tr>
<tr>
<td>4.7</td>
<td>Flowchart of the pre-emptive dropping scheme in smart queue</td>
<td>20</td>
</tr>
<tr>
<td>4.8</td>
<td>Flowchart of the sorting scheme in smart queue</td>
<td>21</td>
</tr>
<tr>
<td>4.9</td>
<td>Packets in the queue before sorting</td>
<td>23</td>
</tr>
<tr>
<td>4.10</td>
<td>Packets in the queue after sorting</td>
<td>24</td>
</tr>
<tr>
<td>4.11</td>
<td>GOP based rate control</td>
<td>25</td>
</tr>
<tr>
<td>4.12</td>
<td>De-queue function</td>
<td>25</td>
</tr>
<tr>
<td>5.1</td>
<td>Average packet deliver ratio of all the priority packets of the integrated scheme (scheme1) vs. the average PDR of scheme 3 for varying number of connections at node speed of 5m/sec</td>
<td>29</td>
</tr>
<tr>
<td>5.2</td>
<td>Average packet deliver ratio of priority-1 packets for different Schemes for varying number of connections at node speed of 5m/sec</td>
<td>29</td>
</tr>
<tr>
<td>5.3</td>
<td>Average packet deliver ratio of all priority packets for different schemes for four connections at node speed of 5m/sec</td>
<td>30</td>
</tr>
<tr>
<td>5.4</td>
<td>Average packet deliver ratio of all the priority packets of the integrated scheme (scheme1) vs. the average PDR of scheme 3 for varying number of connections at node speed of 10m/sec</td>
<td>30</td>
</tr>
<tr>
<td>5.5</td>
<td>Average packet deliver ratio of priority-1 packets for different schemes for varying number of connections at node speed of 10m/sec</td>
<td>31</td>
</tr>
<tr>
<td>5.6</td>
<td>Average packet deliver ratio of all priority packets for different schemes for four connections at node speed 10m/sec</td>
<td>32</td>
</tr>
</tbody>
</table>
Figure 5.7. Average packet deliver ratio of all the priority packets of the integrated scheme (scheme1) vs. the average PDR of scheme 3 for varying number of connections at node speed of 20m/sec. .................................................................32

Figure 5.8. Average packet deliver ratio of priority-1 packets for different schemes for varying number of connections at node speed of 20m/sec. .................................................................33

Figure 5.9. Average deliver ratio of all priority packets for different schemes for four connections at node speed of 10m/sec. ........................................................................33

Figure 5.10. Average PSNR gain in dB over AOMDV for varying number of connections at node speed of 5 m/sec. ........................................................................35

Figure 5.11. Average PSNR in dB for different Schemes for varying number of connections at node speed of 5 m/sec. ........................................................................35

Figure 5.12. Average PSNR gain in dB over AOMDV for varying number of connections at node speed of 10 m/sec. ........................................................................36

Figure 5.13. Average PSNR in dB for different schemes for varying number of connections at node speed of 10m/sec. ........................................................................36

Figure 5.14. Average PSNR gain in dB over AOMDV for varying number of connections at node speed of 20m/sec. ........................................................................37

Figure 5.15. Average PSNR in dB for different schemes for varying number of connections at node speed of 20m/sec. ........................................................................38

Figure 5.16. The 128th frame of the original foreman video. .......................................................38

Figure 5.17. The 128th frame of the foreman video received at the destination using scheme-4 (AOMDV + FIFO queue + IEEE802.11). ...................................................39

Figure 5.18. The 128th frame of the foreman video received at the destination using scheme-1 (AOMDV_DPU + Smart queue + IEEE802.11e). .........................................39

Figure 5.19. The 79th frame of the original foreman video. ..........................................................40

Figure 5.20. The 79th frame of the foreman video received at the destination using scheme-4 (AOMDV + FIFO queue + IEEE802.11). ...................................................40

Figure 5.21. The 79th frame of the foreman video received at the destination using scheme-1 (AOMDV_DPU + Smart queue + IEEE802.11e). .........................................41

Figure 5.22. Average packet deliver ratio for different users with source priority (user 1 > user 2) for the integrated scheme for 2 connections at node speed of 5 m/sec. ........................................................................42

Figure 5.23. Average packet deliver ratio for different users with source priority (user 1 > user 2) for the integrated scheme for 2 connections at node speed of 10 m/sec. ........................................................................43

Figure 5.24. Average packet deliver ratio for different users with source priority (user 1 > user 2) for the integrated scheme for 2 connections at node speed of 20 m/sec. ........................................................................43
Figure 5.25. Average PSNR in dB for different users with source priority (user 1 > user 2) for the integrated scheme for 2 connections at different node speeds. ..........44

Figure 5.26. Average packet deliver ratio for different users with source priority (user 1, user 3 > user 2) for the integrated scheme for 3 connections at node speed of 5 m/sec.................................................................................................................................................45

Figure 5.27. Average packet deliver ratio for different users with source priority (user 1, user 3 > user 2) for the integrated scheme for 3 connections at node speed of 10 m/sec..................................................................................................................................................45

Figure 5.28. Packet deliver ratio for different users with source priority (user 1, user 3 > user 2) for the integrated scheme for 3 connections at 20 m/sec.................................................46

Figure 5.29. Average PSNR in dB for different users with source priority (user 1 > user 2) for the integrated scheme for 3 connections at different node speeds..........46

Figure 5.30. Average packet deliver ratio for different users with source priority for 4 connections at node speed of 5 m/sec (user 1, user 3 > user 2, user 4). .................47

Figure 5.31. Average packet deliver ratio for different users with source priority for 4 connections at node speed of 10 m/sec (user 1, user 3 > user 2, user 4). ..................47

Figure 5.32. Average packet deliver ratio for different users with source priority for 4 connections at node speed of 20 m/sec (user 1, user 3 > user 2, user 4). ...................48

Figure 5.33. Average PSNR in dB for different users with source priority (user 1 > user 2) for the integrated scheme for 4 connections at different node speeds..........48
CHAPTER 1

INTRODUCTION

In Mobile Wireless Networks, the wireless nodes can change their location and configure themselves. The network can use a standard Wi-Fi connection or a cellular or satellite transmission. Some networks are restricted to a local area of wireless devices (such as a group of laptop computers), while others may be connected to the Internet. An example of such a network is a VANET (Vehicular Ad Hoc Network) that allows vehicles to communicate with roadside equipment.

The challenge faced by Mobile Ad Hoc Networks (MANET) used in this thesis is the transmission of real time multimedia data like the Audio and Video, while considering their QoS and low latency demands.

1.1 MOTIVATION OF THESIS

The transmission of video traffic over MANETs [1] is a challenge. H.264 compressed video codec is very sensitive to noise in the wireless channels. The reason for the sensitive nature of H.264 towards noisy channel is the high entropy of the compressed data. Hence, it is necessary to identify the important data packets. For instance, the I-frame is more important than the B-frame of the video. The importance is because the I-frame is used to decode other P-frames and B-frames. On the other hand, the loss of B-frame will not affect the decoding of other frames. By prioritizing the important video packets and serving them better, we can achieve better video quality over the MANET.

MANETs are prone to noisy channels and congestion in the network. As a result, we can expect packet losses in the network during data transmission. Hence, by providing a suitable QoS i.e. making sure that the higher priority packets are being served efficiently by utilizing the available bandwidth, we can increase the overall efficiency of the network.

In this thesis, we propose a Buffer Management scheme that considers the priority of the video packets for serving the packets. We make use of the channel access facilities of IEEE802.11e for providing channel access to the packets based on priority.
1.2 RESEARCH CONTRIBUTIONS

- Review of existing ad hoc routing protocols such as Ad-hoc On-Demand Distance Vector routing (AODV), Ad-hoc On-Demand Multipath Distance Vector Routing (AOMDV) and Ad-hoc On-Demand Multipath Distance Vector Routing with Dynamic Path Upgrade (AOMDV-DPU).
- Performance analysis of AOMDV and AOMDV-DPU in terms of Packet Delivery Ratio and average PSNR for video traffic using NS-2 simulator.
- Design and implementation of a Smart Queue scheme at the Interface Queue between the routing layer and the MAC layer for providing better Quality of Service for video traffic over ad hoc networks.
- Integration of the Smart Queue scheme with the routing protocol AOMDV-DPU at the network layer and the IEEE802.11e at the MAC layer. Performance analysis of the integrated scheme with the First-In-First-Out (FIFO) scheme.

1.3 THESIS OUTLINE

This thesis contains 7 chapters as discussed below:

- Chapter 1 gives a brief introduction to the Mobile Wireless networks and the challenges faced during data transmission through these networks.
- Chapter 2 gives an overview of the H.264 Video Codec that is used as the traffic in the Wireless Ad Hoc networks in our simulations.
- Chapter 3 introduces to MANETs and then gives a brief description of various Routing protocols AODV, AOMDV and AOMDV-DPU.
- Chapter 4 proposes a Smart Queue technique that provides overall improvement in the goodput of the network there by producing better video quality.
- Chapter 5 discusses the Simulation Results of the various schemes indicating the improvement provided by the routing protocol AOMDV-DPU and the Smart Queue scheme to the goodput of the network.
- Chapter 6 summarizes the research effort.
- Chapter 7 proposes the future work.
CHAPTER 2

INTRODUCTION TO H.264 VIDEO CODING STANDARD

2.1 INTRODUCTION

H.264/AVC is a state-of-the-art video coding standard developed by the Joint Video Team (JVT). Its enhanced compression performance and “network friendliness” make this standard very popular [2].

Conceptually, H.264 encoder is divided into Video Coding Layer (VCL) and Network Abstraction Layer (NAL). VCL is designed to efficiently represent the video content and NAL to format the VCL representation so as to be compatible with various transport streams or storage media.

Figure 2.1 illustrates the structure of an H.264/AVC video encoder. VCL generates the coded macroblocks (MB). These MBs are aggregated to form slices at the NAL by exploiting context adaptive coding. The H.264 slices consist of macroblocks processed in raster scan order when not using FMO. Slices formed at the NAL can be classified into 5 different priorities based on their properties. Each slice can be considered as a packet ready to be transmitted over a network.

![Figure 2.1. Structure of H.264/AVC Encoder.](image-url)
2.2 VCL

The VCL design follows the so-called block based hybrid video coding approach, in which each coded picture is represented in block-shaped units of associated luma and chroma samples called macroblocks. The basic source-coding algorithm is a hybrid of inter-picture prediction to exploit temporal statistical dependencies and transform coding of the prediction residual to exploit spatial statistical dependencies. There is no single coding element in the VCL that provides the majority of the significant improvement in compression efficiency in relation to prior video coding standards. It is rather a plurality of smaller improvements that add up to the significant gain [3].

2.3 SLICES AND SLICE GROUPS

Slices are a sequence of MBs which are processed in the order of a raster scan when not using FMO. A picture may be split into one or several slices as shown in Figure 2.2. A picture is therefore a collection of one or more slices in H.264/AVC. Slices are self-contained in the sense that given the active sequence and picture parameter sets, their syntax elements can be parsed from the bit stream and the values of the samples in the area of the picture that the slice represents can be correctly decoded without use of data from other slices provided that utilized reference pictures are identical at encoder and decoder. Some information from other slices may be needed to apply the de-blocking filter across slice boundaries.

Regardless of whether FMO is in use or not, each slice can be coded using different coding types as follows.

- **I slice**: A slice in which all macroblocks of the slice are coded using intra prediction.
- **P slice**: In addition to the coding types of I slice, some macroblocks of the P slice can also be coded using inter prediction with at most one motion-compensated prediction signal per prediction block.
- **B slice**: In addition to the coding types available in a P slice, some macroblocks of the B slice can also be coded using inter prediction with two motion-compensated prediction signals per prediction block.
2.4 Flexible Macroblock Ordering (FMO)

H.264 provides the possibility of increasing the resilience of a coded stream by means of FMO. Using FMO, each macroblock can be assigned freely to a specific slice group using a macroblock allocation map (MBA map). FMO provides great flexibility in terms of defining the coding order of macroblocks within a picture. H.264/AVC allows defining of a maximum of eight slice groups in one picture and the macroblocks within a certain slice group can further be subdivided into a number of slices. The case where there is only one slice group within a picture is similar to case of using no FMO. The power of FMO as an error resiliency tool lies within the way the macroblocks are ordered. For instance, slice groups can be constructed independently from each other and a picture can be decoded even if not all the slice groups are available or even if some slices are missing from one or more slice groups since it is possible to use information of the surrounding macroblocks. Another important quality of FMO-based error resiliency is that it is particularly well suited to real-time and low-delay applications like video conferencing.
CHAPTER 3

ROUTING PROTOCOLS

3.1 MOBILE AD-HOC NETWORKS (MANET)

A mobile ad hoc network (MANET) is a self-configuring network that has a dynamic topology due to the mobility of the nodes. The connection is wireless in the sense, there is no physical connection between the nodes. These nodes have a random movement and hence the route between them changes dynamically. As a result, each node must be capable of routing the data that is not meant for it. Thus, Routing plays an important role in MANETs.

Figure 3.1 represents a typical MANET. As we can see, the nodes can be any mobile device like a laptop, a phone etc. There exists no actual physical connection between these nodes. However, there exists a communication between them through wireless. MANETs are self-forming and there is no centralized handling of the communication between the nodes.

Some applications of MANET technology could include industrial and commercial applications involving cooperative mobile data exchange. In addition, mesh-based mobile networks can be operated as robust, inexpensive alternatives or enhancements to cell-based mobile network infrastructures. There are also existing and future military networking requirements for robust, IP-compliant data services within mobile wireless communication networks many of these networks consist of highly-dynamic autonomous topology segments.

3.1.1 Difficulties Faced in MANETs

- Since, the medium and the environment used are unreliable, there is a possibility of the packets arriving in different order or there may be error in the packets leading to transmission errors [4].
- The failure of a single node might lead to the loss of a lot of information.
- The dynamic environment coupled with Node failures can lead to the failures in links between the nodes.
- Addition of new nodes or loss of existing nodes can cause breakages in the routes or the routes may become stale.
3.1.2 Routing in MANET

The nodes in MANET are not aware of the network topology. Hence, it is necessary for the nodes to broadcast their presence to the neighboring nodes when they join the network. Similarly, it is important to determine the loss of any node in the network [5]. Each node in MANET acts as a router for forwarding packets to other mobile nodes in the network. Routing becomes an important part of a successful network.

Routers are small physical devices that join multiple networks together. Technically, a router is a Layer 3 gateway device, meaning that it connects two or more networks and that the router operates at the network layer of the OSI model.

By maintaining configuration information in a piece of storage called the routing table, wired or wireless routers also have the ability to filter traffic, either incoming or outgoing, based on the IP addresses of senders and receivers. Some routers allow a network administrator to update the routing table from a Web browser interface [6]. Broadband routers combine the functions of a router with those of a network switch and a firewall in a single unit.
3.2 ROUTING PROTOCOLS

Routing protocol is a protocol used by a router to determine the appropriate path over which data is transmitted. The routing protocol also specifies how routers in a network share information with each other and report changes. The routing protocol enables a network to make dynamic adjustments to its conditions, so routing decisions do not have to be predetermined and static.

3.3 AD-HOC ON-DEMAND DISTANCE VECTOR ROUTING (AODV)

Ad-hoc On-Demand Distance Vector routing (AODV) also called as pure on-demand route acquisition system works both on wired and wireless media. A node does not have to discover any routing information nor participate in any periodic routing table exchanges. The nodes broadcast route discovery packets only when necessary. They have to distinguish between local connectivity management and general topology maintenance. The changes in routing information are only transmitted to neighboring nodes that are likely to need the information.

Path discovery is initiated whenever a node needs to communicate with another node as shown in Figure 3.2. It is started by initializing two separate counters node sequence number and broadcast_id. The source (node A in the figure) initiates route discovery by broadcasting Route request packets (RREQ). RREQ contains the following:

< Source address; source sequence number; broadcast id; destination address; destination sequence number; hop count >

Source sequence number is used to maintain freshness information about the reverse route to the source. Destination Sequence number is used to specify how fresh a route to destination must be. An Intermediate node, if it has a route to the source (node D in the Figure 3.2) may reply to the source with a Route Reply packet (RREP) or forward the RREQ packet to the next available node after increasing the hop count.

- The Reverse path is set up through the RREQ packets. As the RREQ packet traverse from the source to the destination, it keeps a track of the path it follows. As a result, the reverse path is stored in the RREQ.
- When an RREQ packets arrive at a node which has a route to the destination. It checks for following conditions. If the Destination sequence number in RREQ is greater than Destination Sequence number recorded by intermediate node, then
Figure 3.2. Illustration of the Route discovery process in AODV.

the RREQ is forwarded to the neighboring node. If Destination sequence number in RREQ is less than Destination Sequence number recorded by intermediate node, a RREP (Route Reply) packet is unicast to its neighbor. Since the reverse path is already set up, the RREP packet travels to the source.

- If multiple RREPs are received at a node, the one with greater destination sequence number or with same destination sequence number are transmitted. All the remaining RREPs are suppressed by the intermediate nodes.

3.4 Ad-hoc On-Demand Multipath Distance Vector Routing (AOMDV)

Ad-hoc On-Demand Multipath Distance Vector Routing (AOMDV) is an extension of AODV to compute multiple disjoint and loop-free path routes. In AOMDV, the RREQ packets from the source establish multiple reverse paths both at the destination and at the intermediate nodes. Similarly, the RREP packets establish multiple forward paths both at the destination and at the intermediate nodes. The advantage of this protocol over AODV is that during path breakage, the source does not have to re-initiate route discovery, instead it can obtain the next best route in the routing table to forward the packets.

When an intermediate node obtains a reverse path via a RREQ copy, it checks whether there are one or more valid forward paths to the destination. If so, the node generates a RREP and sends it back to the source along the reverse path, the RREP includes a forward path that was not used in any previous RREPs for this route discovery. In this case, the
intermediate node does not propagate the RREQ further. Otherwise, the node re-broadcasts the RREQ copy if it has not previously forwarded any other copy of this RREQ.

3.5 Ad-hoc On-Demand Multipath Distance Vector Routing with Dynamic Path Upgrade (AOMDV-DPU)

Ad-hoc On-Demand Multipath Distance Vector Routing with Dynamic Path Upgrade (AOMDV-DPU) is applicable mainly for real time data like video transmission over MANET. AOMDV-DPU is an extension of AOMDV with the following improved features over AOMDV:

- Number of hops to the destination and Received Signal Strength Index (RSSI) are used as the routing metrics for route discovery.
- Local path update mechanism is used to avoid frequent route breakages.
- A Keep alive mechanism is used to maintain the secondary routes to reduce the route discovery frequency.
- Reduced usage of Hello messages and a package salvage mechanism is used to ensure packet delivery during route breakages.
CHAPTER 4

BUFFER MANAGEMENT TECHNIQUES

In this chapter, we discuss about various buffer management techniques for Mobile Ad-Hoc Network (MANET) and also propose a new scheme (Smart Queue). During congestion, buffer management plays an important role in improving the efficiency of a network in terms of throughput, latency and goodput. Based on the network conditions, buffer management helps in serving the packets better. Here, we focus on the buffer management techniques proposed to achieve QoS for video data over MANET. In our scheme, we emphasize on serving the prioritized video packets over mobile wireless networks [7].

4.1 RELATED SCHEMES

Seong-ryong et al., [8] proposed a video streaming framework that allows applications to mark packets with different priority and use multi-queue congestion control inside routers to effectively drop the less-important packets during buffer overflow. In [1] author proposed a buffer management scheme called Frame-Level Packet Discard with Dynamic Thresholds (FDDT), in which the packets are sorted/dropped based on the following conditions:

1. The first packet of B or P-frame is discarded when the buffer size reaches certain threshold levels.
2. An I-frame packet is only discarded when buffer is completely full.
3. If early packet of a frame is dropped, all the subsequent packets are dropped.
4. When two packets are competing, lower priority packets are dropped.
5. All incoming packets are dropped when the buffer is full. FDDT scheme showed results with great advantage for video quality, while having a small increase in computational complexity.

In [9], a Drop Dependency Based (DDB) scheme was proposed, where basic information of the packet priorities was provided in the packet header and a buffer management was done based on this information. An optimized strategy operates on the
Head of line (HOL: The packet which resides longest in the buffer) group of packets. By dropping the HOL packet with the lowest priority, a significant improvement in the video quality was achieved. This was extended in [10] to achieve an optimal combination of scheduler and drop strategy.

There have been some smart router based solutions such as Active networking [11]. In which routers play very important role of smartly discarding the packets based on a priori knowledge of the transmission in progress. But, having functionality and computational complexity in the routers can go against the general norm of just forwarding the packets to the next node. But, this feature can be of great advantage in MANETs where the mission/video information of the current communication can be transmitted in the header during session establishment. In [12] author introduces a new term called fairness of service, which means the fair allocation of available resources to achieve the expected quality. It means that more resources are allocated for higher QoS requirement streams. The lower QoS requirement stream packets are dropped while giving more access to important streams. This can lead to starvation of resources for the later.

In Van der Schaar [13], a joint APP and MAC adaptation scheme was proposed with the use of MPEG-4 and its Fine Granularity Scalability (FGS) extension. In this work, packets containing multimedia data are classified into different classes and in the light of poor network conditions only packets with high class value are transmitted. The network conditions are jointly measured by combining the information obtained by the retransmission number of lost MAC frames (ARQ) and the information provided by the RTCP protocol. Authors in [14] proposed a selective retransmission scheme for multimedia transmission over wireless networks. The idea was to retransmit only the important information in a video in order to achieve high quality of video streaming. Retransmission of important packets is a good idea, but since prevention is better than cure and we need to protect the high priority information and reduce the packet loss rate (PLR).

### 4.2 Prioritization of H.264 Video Packets

In our scheme, we use prioritized H.264/AVC video traffic over Mobile Wireless Ad hoc networks. It is well known that each slice in video has its own importance and contributes towards the quality of the video. Hence, the decoded video quality can be
improved by intelligently controlling the slice losses. For instance, the slices which contain the information of the motion in the video will have more significance than the slices having other information. Hence, when slices containing important information are lost, it has adverse effects on the video quality. In our scheme, the priority to the slices in the video is assigned based on the Cumulative Mean Square Error (CMSE) contributed by their loss.

### 4.3 VIDEO PACKET FORMATION IN NS-2

We use the network simulator NS-2 for implementing our scheme. NS-2 has provision for Constant bit rate Data packets. Since, we use video traffic in our simulations; we use Evalvid to generate video packets. At the application layer, the video packets that are generated from the H.264 trace files using the CSME based prioritization is attached to the NS-2 agent using the Evalvid [15].

### 4.4 CMSE COMPUTATION/PREDICTION OF H.264 VIDEO SLICES

The H.264 slices are prioritized based on their distortion contribution to the received video quality [2]. The loss of a slice introduces error in the current reference frame and could propagate to other frames in the GOP. The total distortion is computed by using the CMSE introduced by a slice loss, since it takes into consideration the error propagation within the entire GOP.

The CMSE contributed by the loss of the slice is computed in Equation below as the sum of mean squared error (MSE) over the current and all the other frames in the GOP,

\[
CMSE = \sum_{i=\text{current frame with slice loss}}^{\text{last frame of GOP}} \left\{ \frac{1}{H \times W} \sum_{j=1}^{H} \sum_{k=1}^{W} \left( \hat{Pe}_{i,j,k} - \bar{Pe}_{i,j,k} \right)^2 \right\}
\]

Here,

- ‘t’ → is the temporal duration of a reference frame in the backward direction.
- \(H \times W\) → Height and Width of the number of pixels of a video frame.
- \(\hat{Pe}_{i,j,k}\) → The pixel energy value at location (j, k) in the reconstructed frame i at the encoder without the slice loss.
- \(\bar{Pe}_{i,j,k}\) → The corresponding pixel energy value in the same frame decoded at the receiver with the slice loss.
However, the computation of slice CMSE introduces high computational overhead as it requires decoding the entire GOP for every slice loss. The slice contributing the highest distortion is the most important slice (i.e., highest priority). This process defines the relative importance order for the slices in the GOP.

Figure 4.1 represents the basic Queue structure in ns-2. The Interface Queue (IFQ) is the buffer between the Link layer and the MAC that holds the packets when they do not get channel access during congestion in the network. The Link layer is responsible for addressing. It makes use of the Address Resolution Protocol (ARP) to decide the source and destination address of a packet. MAC layer is responsible for the channel access and scheduling. It continuously monitors the channel and then schedules the packet transfer based on the availability of the channel.

4.5 BASIC NS-2 QUEUING SCHEME

Once the address of a packet is resolved by the Link layer, it passes it to the MAC for transmission. In case of congestion, the MAC may not be able to schedule the packets immediately. Hence, these packets get stored in the buffer (IFQ). The packets may get lost depending on the length of the Queue due to buffer overflow or they might expire in the Queue. By carefully managing the buffer between the Link layer and the MAC, it is possible to achieve higher successful transmission rates during congestion in the network.

4.5.1 Definitions

1. **Time To Live (TTL):** The time within which the Packet must travel from the Source to the Destination is called the Time to Live. We include a field in the packet header defining the time to live of that particular packet. The difference in the time of generation of the packet and the time at which the packet is received gives the delay encountered by the packet to reach the destination. If the delay encountered by the packet exceeds the TTL defined for it, the packet is expired and hence dropped. TTL helps in eliminating the existence of stale packets in the network and helps in reducing the congestion due to the undelivered packets by increasing the bandwidth. The importance of TTL is seen in delay sensitive and real time applications. We use the concept of TTL in our scheme to discard the stale packets making way for the packets that are alive.
2. **First In First Out (FIFO):** FIFO is a technique in which the packets in the queue are processed in the order in which they appeared. The packet which comes first in the queue is served before the other packets irrespective of the priority and the importance of the packet. This technique is purely based on the time stamp of the packet. The major disadvantage of this technique is that the higher priority packets which are of greater importance may not be processed because of the lower priority packets that stay in the queue ahead of the higher priority packets without getting the channel access. The basic queuing scheme in ns-2 is FIFO and we compare our scheme with the FIFO.

**4.5.2 Interface Queue in NS-2**

Interface Queue (IFQ) is the queue between the Link Layer and the MAC layer. The packets from the Link layer are queued until the MAC layer assigns the channel access to the packets. There are two types of packets that can appear at the IFQ:

1. **Control Packets** - These include the Route Request (RREQ), Route Reply (RREP) and Acknowledgement (ACK) packets.
2. Data Packets - These include the actual data to be transferred. It can be text, images or video.

By default, the Control packets are the higher priority packets and the Data packets are the lower priority packets. In the IFQ, when a Control packet appears, it is placed at the front of the Queue ahead of the Data packets.

In the basic NS-2 Queue, the queuing mechanism has three major functions (see Figure 4.2):

- **Enqueue**: here, the TTL of the incoming packet is checked. If the packet is expired, it is dropped.
- **Sort**: here, the packets are enqueued in the queue based on FIFO.
- **Dequeue**: here, the TTL of the packet is checked before dequeuing onto the MAC layer.

![Figure 4.2. Flowchart of the IFQ in NS-2.](image)

**4.5.3 Event Scheduler in NS-2**

NS-2 is an event driven simulation. The main concept here is to put all the events on the simulation timeline, move along the timeline and take actions when confronting any event. An event indicates what happens and a handler indicates what to do. As shown in Figure 4.3, every event is marked by a Unique ID. NS-2 maintains a common pool of Unique IDs and then each event is assigned an ID Known as the UID.
4.6 BUFFER OVERFLOW

Buffer overflow is one of the major challenges in wireless networks. The major cause of buffer overflow is the congestion in the network. Generally, the routers use the FIFO or the Drop-Tail Queue Management Scheme. Here, the packets as they arrive are en-queued at the tail of the queue and de-queued at the head as shown in Figure 4.4.

Buffer overflow occurs when a new packet arrives while the queue is full to and hence can’t be stored in the queue. In this situation, the packet is dropped irrespective of its importance. This packet drop can have adverse effects on the QoS especially in real time applications. For instance, the packet dropped due to buffer overflow maybe an IDR frame of
the video sequence. When such a packet is dropped, it will significantly affect the quality of the video. Hence, it becomes necessary to manage the queue and intelligently drop the packets that are of low importance [16].

4.7 SMART QUEUE SCHEME FOR VIDEO TRAFFIC

Based on the drawback discussed about the FIFO/drop tail queuing scheme, we have designed a Smart Queue technique. The purpose is to improve the QoS of the network by dropping the video packets based on their importance (i.e., priority) and flushing the stale packets (whose deadline has expired) [17]. Based on the CMSE, each video packet is given a priority, indicating its importance. In our simulations, we have used four different packet priorities (P1, P2, P3 and P4). The basic idea of our scheme is shown in the flowchart in the Figure 4.5.

![Flowchart of the smart queue technique.](image)

4.7.1 At the Enqueue

As discussed earlier, Time-To-Live (TTL) is the time within which a packet must travel from its source node to the destination node in the network. Based on the priority of the packet, each packet is assigned a pre-determined value of TTL. Generally, a higher priority packet has larger TTL value.
4.7.1.1 **Enqueue the Packet**

The TTL values of the packets that arrive at the router are checked before enqueuing the packets on to the queue. If the packet has not expired, it is enqueued otherwise the packet is dropped. The delay encountered by the packet to travel to the node is calculated as,

\[
\text{Delay} = \text{Current time} - \text{Time at which the packet was generated}
\]

The purpose of checking the TTL is to delete the stale packets in the network. There is a possibility that these stale packets can block the other important packets from accessing the channel, thereby reducing the overall performance of the network. The process of checking the TTL of the packets before en-queuing is shown below in Figure 4.6.

![Flowchart](image)

Figure 4.6. Flowchart of the en-queue in smart queue.

4.7.1.2 **Pre-emptive Dropping of Packets**

One way of avoiding the buffer overflow is by periodically checking the buffer fullness. In order to avoid the loss of important packets, we intelligently drop the lower priority packets as follows (also shown in Figure 4.7):

- If the current queue length is >= 75% of the total queue size, we drop all the priority four (P4) packets in the second half of the queue.
- If the current queue length is >= 85% of the total queue size, we drop all the priority four (P4) packets from the entire queue.
- If the current queue length is 95% of the total queue size, we drop all the priority four (P4) packets and half of the priority three (P3) packets of the queue.
The idea here is to serve the higher priority packets in unfavorable network conditions. This helps in increasing the network goodput, thereby improving the quality of the video by helping the receiver to decode the video. The procedure of pre-emptive dropping is shown in Figure 4.7.

**4.7.2 Sorting Algorithm**

Once the packets are en-queued in the queue, it is important for us to make sure that the higher packets are not blocked in the queue for a longer time. We use a sorting technique to arrange the packets of a frame that are en-queued in the queue based on their priority. This allows the higher priority packets to be served first and given channel access before they expire in the queue. The packet sorting starts from the head of the queue and traverse through the entire queue. This Sorting algorithm used is similar to the bubble sorting technique.
Since we have used video traffic in our simulations, we perform the sorting in the following order as shown in Figure 4.8 and discussed below:

- Frame Number based sorting.
- Priority based sorting.
- Time-stamp based sorting.

Figure 4.8. Flowchart of the sorting scheme in smart queue.
4.7.2.1 Frame Number Based Sorting

As we know, the video sequence is divided into frames. Hence, in order for the decoder to decode the video appropriately it is necessary for us to make sure that the video frames reach the decoder in the order they were transmitted. Each frame is given a number (Frame Number) based on the order of transmission. At the queue, the packets may arrive from various nodes in different order. We arrange these packets based on their frame number so that all the packets with a lower frame number are stored together and get access to the channel first.

4.7.2.2 Priority Based Sorting

We use the concept of time-to-live in our network, where the packets are given a time within which they must be delivered to the destination. During congestion in the network, it may happen that the packets may wait longer than expected in the queue without getting the channel access.

As a result of this, it is possible for some of the packets near the tail of the queue to expire and thus be dropped. In this process, we might lose some higher priority packets of a frame if they are behind the lower priority packets in the queue. In order to eliminate this, we sort all the packets of a frame based on their priority within the queue thereby, giving access to the higher priority packets earlier than the lower priority packets.

4.7.2.3 Time-stamp Based Sorting

Every packet in NS-2 is stamped with the current time at the time of generation. This time stamp is helpful in calculating the delay it experiences in the network. It is possible that at an intermediate node, two packets having the same frame number and priority have arrived (this is possible only when you consider packets from two different video sources). Suppose, one of these packets has travelled through some distance and the other has just begun its journey. It may happen that the packet which has just begun might be placed ahead of the older packet in the queue blocking it and eventually causing the packet to expire. In order to eliminate this issue, we arrange the packets based on the time-stamp at which it was generated i.e. the packet with earlier time-stamp is ahead in the queue.
4.7.3 Working of Sorting Technique

Let us consider a queue having the following packets as shown in the Figure 4.9.

Suppose a new packet P – (F2, p3, t1) arrives at the queue, the following mechanism is used to store it in the queue:

- The TTL of the packet is checked to see if it has expired.
- If the packet is not expired, the frame number of the packet is compared with the frame numbers of the packets that are in the queue.
- If the packet has the same frame number as that in the queue, the priority of the packet is compared.
- Finally, the time-stamp of the packets is compared and the packet is inserted in the queue appropriately.

The Queue after Sorting is shown in Figure 4.10.

4.7.4 Adaptive Rate Control

If the network is unable to match the packet injection rate, we drop the packets in the queue as a preemptive measure. This is done by calculating the number of packets served by the network in a unit time. Based on this information, we drop the lower priority packets. This technique is well suited for Scalable Video Coding.
Rate control is applicable when the bandwidth of the network available is less than the data to be transferred [18]. Video packets are transmitted in chunks of Group of Pictures (GOP). Each GOP is given a unit time in the queue within which they are expected to be served. In case the given GOP could not be served within a unit time, the remaining packets of the GOP are dropped. This method mainly concentrates on flushing of the stale packets giving other important packets a chance to be served thereby increasing the goodput of the network. Higher goodput implies higher PSNR and hence better video quality.

The concept of rate control helps in making sure that the queue does not fill and to avoid the buffer overflow. In our simulations, we assign 20 frames as a GOP. Hence, every 20 frames are given a unit time to be served as shown in Figure 4.11. Generally, the lower priority packets are present at the back end of the GOP and hence dropping these packets will not affect the video quality by much.

The advantage of rate control protocol is it produces a better quality-to-space ratio. The available data packets are used flexibly to encode the video data more accurately. If the video encoding rate is greater than the data rate supported by the network, the remaining packets of the GOP in the queue are dropped otherwise the transmission is continued.

### 4.7.5 At the Dequeue

Before dequeueing a packet from the queue, its TTL is checked. If the packet is expired, it is dropped. The dropping of a packet at the Dequeue is shown in Figure 4.12. This
ensures that the packet which is to be de-queued to the MAC layer [19] has enough time to be transmitted over the network.

**4.8 IEEE 802.11E**

IEEE 802.11e is an approved amendment to the IEEE 802.11 standard that defines a set of QoS enhancements for wireless LAN applications through modifications to the MAC layer. The standard is considered of critical importance for delay-sensitive applications, such as Voice over Wireless LAN and streaming multimedia.

Since, we use the common channel for all the nodes in the network, it is necessary to serve the users with higher priority packets first. We use the EDCA mode of the IEEE802.11e.
Hence, at the MAC layer we serve the packets based on priority by setting different contention window size, arbitrary inter-frame space (AIFS) and packet retransmission reitres limit. Based on the priority, the following values are set as shown in Table 4.1:

**Table 4.1. Contention Window Values Based on Packet Priority**

<table>
<thead>
<tr>
<th>Priority</th>
<th>CWMIN (slots)</th>
<th>CWMAX (slots)</th>
<th>AIFS (slots)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>3</td>
<td>7</td>
<td>2</td>
</tr>
<tr>
<td>2</td>
<td>7</td>
<td>15</td>
<td>3</td>
</tr>
<tr>
<td>3</td>
<td>15</td>
<td>31</td>
<td>5</td>
</tr>
<tr>
<td>4</td>
<td>31</td>
<td>1023</td>
<td>7</td>
</tr>
</tbody>
</table>

*Note. Each slot = 250 micro sec.*  
The contention window is selected randomly in the range: CWMIN < CW < CWMAX.
CHAPTER 5

RESULTS

In this chapter, we present the results of a wide range of simulations run using NS-2. Here, we compare our Smart Queue scheme with the basic NS-2 queue scheme (FIFO). We also compare the routing schemes AOMDV and AOMDV-DPU.

5.1 NETWORK PARAMETERS

The Network parameters used in the simulation are listed in Table 5.1.

Table 5.1. Network Parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Grid Size</td>
<td>1000m x 1000m</td>
</tr>
<tr>
<td>No of nodes</td>
<td>25</td>
</tr>
<tr>
<td>Channel capacity</td>
<td>2Mbps</td>
</tr>
<tr>
<td>TTL values</td>
<td>Pr 1 = 0.45 sec, Pr 2 = 0.4 sec, Pr 3 = 0.35 sec, Pr 4 = 0.25 sec</td>
</tr>
<tr>
<td>Transmission Range of each node</td>
<td>250m</td>
</tr>
<tr>
<td>Node Speed</td>
<td>5m/sec, 10m/sec and 20m/sec</td>
</tr>
<tr>
<td>Queue Size</td>
<td>100 packets</td>
</tr>
<tr>
<td>Number of iterations</td>
<td>25 (Monte Carlo simulation)</td>
</tr>
</tbody>
</table>

5.2 VIDEO TRAFFIC PARAMETERS

Video traffic parameters used in the simulation are listed in Table 5.2.

Table 5.2. Video Traffic Parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Video used</td>
<td>CIF Foreman video</td>
</tr>
<tr>
<td>Bit rate</td>
<td>256kbps for each source</td>
</tr>
<tr>
<td>Slice size</td>
<td>150 bytes</td>
</tr>
<tr>
<td>GOP size</td>
<td>20 frames</td>
</tr>
</tbody>
</table>
The mobility of the nodes is based on the Random Waypoint Mobility model. Random Waypoint Mobility Model is a model that includes pause times between changes in destination and speed. In our simulations, we have used zero pause time with changing direction and node speed. The different schemes used in the Simulations are listed in Table 5.3.

<table>
<thead>
<tr>
<th>Scheme</th>
<th>Routing Scheme</th>
<th>Smart Queue</th>
<th>MAC standard</th>
</tr>
</thead>
<tbody>
<tr>
<td>Scheme 1: AOMDV_DPU + Queue + 11e (integrated scheme)</td>
<td>AOMDV-DPU</td>
<td>Yes</td>
<td>IEEE 802.11e</td>
</tr>
<tr>
<td>Scheme 2: AOMDV_DPU + Queue</td>
<td>AOMDV-DPU</td>
<td>Yes</td>
<td>IEEE 802.11</td>
</tr>
<tr>
<td>Scheme 3: AOMDV_DPU</td>
<td>AOMDV-DPU</td>
<td>No</td>
<td>IEEE 802.11</td>
</tr>
<tr>
<td>Scheme 4: AOMDV</td>
<td>AOMDV</td>
<td>No</td>
<td>IEEE 802.11</td>
</tr>
</tbody>
</table>

### 5.3 AVERAGE PACKET DELIVERY RATIOS OF DIFFERENT SCHEMES AT NODE SPEED 5M/SEC

From Figure 5.1, we observe that the use of Smart Queue helps in improving the Packet Delivery Ratios of the priority-1 and priority-2 packets thereby improving the goodput of the network. With higher ratios of higher priority packets, the PSNR values obtained for the video is high and hence better video quality.

From the plot in Figure 5.2, we observe that the throughput of the network increases with the use AOMDV_DPU routing scheme. There is a significant gain in throughput with the addition of Smart Queue and IEEE802.11e as well.

From the plot in Figure 5.3, we observe that AOMDV_DPU gives a higher throughput over AOMDV and the addition of Smart Queue and IEEE802.11e increases the goodput by giving higher ratios for higher priority packets.
Figure 5.1. Average packet deliver ratio of all the priority packets of the integrated scheme (scheme1) vs. the average PDR of scheme 3 for varying number of connections at node speed of 5m/sec.

Figure 5.2. Average packet deliver ratio of priority-1 packets for different Schemes for varying number of connections at node speed of 5m/sec.
Figure 5.3. Average packet deliver ratio of all priority packets for different schemes for four connections at node speed of 5m/sec.

5.4 AVERAGE PACKET DELIVERY RATIOS OF DIFFERENT SCHEMES AT NODE SPEED 10M/SEC:

From Figure 5.4, we observe that the use of Smart Queue helps in improving the Packet Delivery Ratios of the priority-1 and priority-2 packets even at node speed of 10m/sec.

Figure 5.4. Average packet deliver ratio of all the priority packets of the integrated scheme (scheme1) vs. the average PDR of scheme 3 for varying number of connections at node speed of 10m/sec.
From the plot in Figure 5.5, we observe that the Average PDR of priority-1 packets is significantly high with the use of AOMDV_DPU along with Smart Queue and IEEE802.11e.

![Figure 5.5. Average packet deliver ratio of priority-1 packets for different schemes for varying number of connections at node speed of 10m/sec.](image)

From the plot in Figure 5.6, we observe that the Smart Queue and IEEE802.11e influences the increase in PDR of higher priority packets and a decrease in the lower priority packets. Clearly, there is an increase in goodput due to increase in PDR of higher priority packets.

5.5 AVERAGE PACKET DELIVERY RATIOS OF DIFFERENT SCHEMES AT NODE SPEED 20M/SEC

From the plot in Figure 5.7, we observe that even at high node speed (20m/sec), the average PDR of the higher priority packets is greater than the lower priority packets. Clearly, the integrated scheme performs well at high node speeds.

From the plot in Figure 5.8, we observe that at high node speed (20m/sec), the combination of AOMDV_DPU with Smart Queue and IEEE802.11e gives a higher PDR.

From the plot in Figure 5.9, we observe that the AOMDV_DPU increases the Average PDR while the Smart Queue increases the goodput.
Figure 5.6. Average packet deliver ratio of all priority packets for different schemes for four connections at node of speed 10m/sec.

Figure 5.7. Average packet deliver ratio of all the priority packets of the integrated scheme (scheme1) vs. the average PDR of scheme 3 for varying number of connections at node speed of 20m/sec.
Figure 5.8. Average packet deliver ratio of priority-1 packets for different schemes for varying number of connections at node speed of 20m/sec.

Figure 5.9. Average deliver ratio of all priority packets for different schemes for four connections at node speed of 10m/sec.
5.6 PSNR Plots

The peak Signal-to-Noise Ratio, often abbreviated PSNR, is most commonly used as a measure of quality of reconstructed video. The signal in this case is the original data, and the noise is the error introduced by channel. The noise is defined via the mean squared error ($MSE$). Given a noise-free $m \times n$ monochrome image $I$ and its noisy approximation $K$, $MSE$ is defined as:

$$MSE = \frac{1}{mn} \sum_{i=0}^{n-1} \sum_{j=0}^{m-1} [I(i,j) - K(i,j)]^2$$

The PSNR is defined as:

$$PSNR = 10 \cdot \log_{10} \left( \frac{MAX^2}{MSE} \right) = 20 \cdot \log_{10} \left( \frac{MAX_I}{\sqrt{MSE}} \right) = 20 \cdot \log_{10} (MAX_I) - 10 \cdot \log_{10} (MSE).$$

5.6.1 Average PSNR of Different Schemes at Node Speed 5m/sec

From the Figure 5.10, we observe that there is a gain in the average PSNR due to AOMDV_DPU, Smart Queue and IEEE802.11e. The gain in dB represented in the plot is with respect to AOMDV.

From the Figure 5.11, we observe that the average PSNR decreases as the number of connections in the network increases.

5.6.2 Average PSNR of Different Schemes at Node Speed 10m/sec

Figure 5.12 represents the gain in average PSNR produced by AOMDV_DPU, Smart Queue and IEEE802.11e over AOMDV at node speed of 10m/sec for different number of connections.

From the Figure 5.13, we observe that the average PSNR for all the Schemes is lesser at node speed of 10m/sec as compared to that at node speed of 5m/sec.
Figure 5.10. Average PSNR gain in dB over AOMDV for varying number of connections at node speed of 5 m/sec.

Figure 5.11. Average PSNR in dB for different Schemes for varying number of connections at node speed of 5 m/sec.
Figure 5.12. Average PSNR gain in dB over AOMDV for varying number of connections at node speed of 10 m/sec.

Figure 5.13. Average PSNR in dB for different schemes for varying number of connections at node speed of 10m/sec.
5.6.3 Average PSNR of Different Schemes at Node Speed 20m/sec

Figure 5.14 represents the gain in average PSNR produced by AOMDV_DPU, Smart Queue and IEEE802.11e over AOMDV at node speed of 10m/sec for different number of connections.

Figure 5.15 represents the average PSNR of different schemes for different number of connections at node speed of 20m/sec.

5.7 VIDEO RESULTS

The following snapshots in Figures 5.16–5.21 represent the Foreman_CIF video clip for different schemes. We can understand the improvement in video quality with the help of a better routing technique (i.e., AOMDV-DPU) and better buffer management technique (i.e., smart queue). The gain in average PSNR discussed above is also seen in the visual video quality. The purpose is to evaluate the adhoc routing protocols in terms of QoS [20].

Sample Snapshots:

Figure 5.14. Average PSNR gain in dB over AOMDV for varying number of connections at node speed of 20m/sec.
Figure 5.15. Average PSNR in dB for different schemes for varying number of connections at node speed of 20m/sec.

Figure 5.16. The 128th frame of the original foreman video.
Figure 5.17. The 128th frame of the foreman video received at the destination using scheme-4 (AOMDV + FIFO queue + IEEE802.11).

Figure 5.18. The 128th frame of the foreman video received at the destination using scheme-1 (AOMDV_DPU + Smart queue + IEEE802.11e).
Figure 5.19. The 79th frame of the original foreman video.

Figure 5.20. The 79th frame of the foreman video received at the destination using scheme-4 (AOMDV + FIFO queue + IEEE802.11).
So far in our discussion, we have considered only the priority of the packets of a video sequence. We discuss below the use of source priority along with the packet priority. The consideration of source priority is useful when users are given different priority in the network.

In this case, two sources are considered: Source 1 with higher priority and Source 2 with lower priority. Their packets are assigned a combined priority as shown in Table 5.4.

**Table 5.4. Combined Packet Priority with Source Priority**

<table>
<thead>
<tr>
<th>Actual Packet Priority</th>
<th>Combined Packet Priority</th>
</tr>
</thead>
<tbody>
<tr>
<td>Source1, priority 1</td>
<td>Priority 1</td>
</tr>
<tr>
<td>Source1, priority 2; Source2, priority 1</td>
<td>Priority 2</td>
</tr>
<tr>
<td>Source1, priority 3; Source2, priority 2</td>
<td>Priority 3</td>
</tr>
<tr>
<td>Source1, priority 4; Source2, priority 3; Source2, priority 4</td>
<td>Priority 4</td>
</tr>
</tbody>
</table>
5.8.1 Results with Source Priority

Simulation Parameters:

- Video bit rate is 256kbps for each source:
- Slice size: 150 bytes, GOP size: 20.
- No of nodes: 25, Channel capacity: 2Mbps.
- TTL values: Pr 1 = 0.45 sec, Pr 2 = 0.4 sec, Pr 3 = 0.35 sec and Pr 4 = 0.25 sec.
- Source Priorities:
  - High Priority – User 1 and User 3.

5.8.2 PDR and PSNR Plots with Source Priority for 2 Connections

From the Figure 5.22, we observe that the introduction of Source priority in the network allows higher priority Source to achieve better average PDR.

From Figures 5.23 and 5.24, we observe that the average PDR for higher priority Source is greater than the lower priority Source at high node speeds (10m/sec and 20m/sec). Figure 5.25 represents the plot of average PSNR for the high and normal priority source.

![Figure 5.22. Average packet deliver ratio for different users with source priority (user 1 > user 2) for the integrated scheme for 2 connections at node speed of 5 m/sec.](image-url)
Figure 5.23. Average packet deliver ratio for different users with source priority (user 1 > user 2) for the integrated scheme for 2 connections at node speed of 10 m/sec.

Figure 5.24. Average packet deliver ratio for different users with source priority (user 1 > user 2) for the integrated scheme for 2 connections at node speed of 20 m/sec.
Figure 5.25. Average PSNR in dB for different users with source priority (user 1 > user 2) for the integrated scheme for 2 connections at different node speeds.

5.8.3 PDR and PSNR Plots with Source Priority for 3 Connections

From Figures 5.26 to 5.29, User 1 and User 3 have a higher PDR than User 2. Hence, shifting of priority of the packet helps in providing better service to the higher priority source.

5.8.4 PDR and PSNR Plots with Source Priority for 4 Connections

From the above graphs and PSNR plots (Figures 5.30 to 5.33), we clearly see that the Packet Delivery Ratio (PDR) of the higher priority users is greater than the normal priority users. In case of 2 connections, User 1 has a higher PDR than the User 2. In case of 4 connections, User 1 and User 3 have higher PDR than User 2 and User 4.
Figure 5.26. Average packet deliver ratio for different users with source priority (user 1, user 3 > user 2) for the integrated scheme for 3 connections at node speed of 5 m/sec.

Figure 5.27. Average packet deliver ratio for different users with source priority (user 1, user 3 > user 2) for the integrated scheme for 3 connections at node speed of 10 m/sec.
Figure 5.28. Packet deliver ratio for different users with source priority (user 1, user 3 > user 2) for the integrated scheme for 3 connections at 20 m/sec.

Figure 5.29. Average PSNR in dB for different users with source priority (user 1 > user 2) for the integrated scheme for 3 connections at different node speeds.
Figure 5.30. Average packet deliver ratio for different users with source priority for 4 connections at node speed of 5 m/sec (user 1, user 3 > user 2, user 4).

Figure 5.31. Average packet deliver ratio for different users with source priority for 4 connections at node speed of 10 m/sec (user 1, user 3 > user 2, user 4).
Figure 5.32. Average packet deliver ratio for different users with source priority for 4 connections at node speed of 20 m/sec (user 1, user 3 > user 2, user 4).

Figure 5.33. Average PSNR in dB for different users with source priority (user 1 > user 2) for the integrated scheme for 4 connections at different node speeds.
CHAPTER 6

CONCLUSION

In this thesis, we implemented a Smart Queue technique in NS-2. We compared the results with the basic FIFO (First-In-First-Out) queue used in NS-2. For prioritized video traffic, we find that our scheme produces better throughput values for higher priority packets. This implies higher goodput, which is seen in the increase in PSNR values resulting in better video quality.

Also, we integrated the Smart Queue scheme with the AOMDV-DPU routing. We compared the performance of the integrated scheme with AOMDV. We find better results both in throughput and PSNR. It is clearly observed that, by using the Smart Queue Scheme along with IEEE802.11e, the more important information packets can be given more channel access to improve the quality of video streaming.

From the results discussed in Section 5.6, we find that there is a gain in the average PSNR contributed by AOMDV_DPU, Smart Queue and IEEE802.11e individually. AOMDV_DPU gives 3dB to 5dB gain in average PSNR based on the node speed and number of connections in the network. The combination of Smart Queue and IEEE802.11e improves the gain in average PSNR by 2dB to 3 dB based on the node speed and number of connections used. From Section 5.7, we observe the improvement in the video quality (lower distortion) with the inclusion of AOMDV_DPU and Smart Queue.

The introduction of Source priority gives an improved performance of the network. As seen in Section 5.8.1, Source priority yields better throughput and PSNR values for the higher priority source than the lower priority source.
CHAPTER 7

FUTURE WORK

This section presents some directions for future work. Possible avenues in which this work can be extended include the following:

1. The route used by the packets considered for the experiment in this thesis is common irrespective of their priority. Different routes can be used for different packets based on their priority [21].

2. The scenarios considered for the experiments in this thesis are based on RWP Mobility Model. It would be interesting to see how the algorithm behaves for different scenarios using different Mobility model.

3. The Smart Queue is integrated with routing protocols AOMDV and AOMDV_DPU. Routing algorithms like OLSR, DSR and DSDV [22] can be used for testing the performance of Smart Queue.

4. We have considered only video traffic for our simulations. It will be interesting to know the performance of our scheme when there are other types of data such as Audio and Text.

5. We have used two different priorities for the Source (High and Normal). More number of priorities can be assigned to the Source. We can have nodes having different type of data and they can be assigned priority based on the data they contain for transmission.

6. In real time applications, it is difficult to have a completely AdHoc network. Hence, we can consider transmission of data using dedicated resources (bandwidth).

7. It will be interesting to consider the dropping of packets based on the delay estimation or probability of the packet expiring within the queue.
REFERENCES


