Chapter 5

A Distortion Aware Novel Routing Protocol for Video Transmission in MANETs

Quality received at the receiver of a transmitted video over MANETs is crucial and challenging issue among the wireless network research societies. High packet loss or error probability due to noise and congestion cause unbearable distortion in transmitted video. Hence, to observe overall distortion and calculate the affect of packet error or packet loss on video quality a probabilistic model is necessary. In this chapter a combined approach is introduced to reduce the effect of congestion and noise in MANETs. To obtain improved QoS, we have provided the solution in two steps– formulation of probabilistic model for estimating distortion in presence of congestion and channel noise, and design of smart node initiated feedback based routing protocol to eliminate channel condition affected distortion. At first, a probabilistic model is conceptualized to estimate video distortion due to random congestion and noise. Based on this model, a feedback based routing strategy, Distortion Aware Routing (DAR), is developed in which mobile nodes are made smart for taking decisions after considering network-specific constraints like,
congestion and SNR. The experiment results are obtained and compared with contemporary rate-adaptive strategy. The results show improvement in video quality for most of the channel environments.

5.1 Introduction

Due to inherent property of MANET, congestion and noise are the two major issues to deal with QoE at the receiver. Packets are dropped due to congestion and do not reach at the receiver whereas noise corrupt the bit(s) if the packets are successfully delivered to the receiver. Therefore, video quality is perceived at the receiver is more destroyed in presence of both of these channel parameters as compared to video quality perceived at the receiver in presence of any of them.

The works surveyed for eg. CRP, ECARP, CADV, MDC.RA-MDC etc. have studied the effects of congestion or noise separately on video quality, but none of the solutions has considered random nature of congestion and noise jointly. Most of them have used techniques based on classical routing protocols. To model the time varying nature of congestion and noise in MANET, a distortion model is needed. Motivated with this a novel routing protocol is proposed in this chapter based on estimating distortion at intermediate node caused by packet loss due to either congestion or errorful packets due to noise or both and then decide its routing strategies. The objective of the proposed work is to design a distortion model in video transmission based on packet loss probabilities caused due to channel conditions like congestion and noise and study the effect of distortion on video quality if they exist either in isolation or together.

Rest of the chapter is structured as follows: Section 5.2 introduces the packet state modeling. The packet state based distortion estimation is performed in Section 5.3. A distortion aware routing protocol is proposed in Section 5.4. Section 5.5 presents simulated results for the proposed protocol and it is concluded in Section 5.6.
5.2 Packet State Modeling

The distortion, independently or jointly, is a function of lost and lossy packets. The packets may be delivered successfully to the destination either in one attempt or in multiple attempts. In the latter case, they may be dropped and retransmitted several times before finally delivered or lost; in both the cases they are in transition. The behavior of packets differs with congestion and noise. In this work the packets, for the purpose of analysis and to obtain appropriate solution, are considered to be in good state, fail state or bad state. The concept is described in the following paragraphs.

Let there are $Z$ number of packets generated from the source and $P$ be the packet space set containing all $Z$ packets lying in any one of the three states (Fig. 5.1) represented as $P = \{P_{good}, P_{fail}, P_{bad}\}$ where the packets satisfying the conditions of good, fail and bad states are the elements of sets $P_{good}$, $P_{fail}$, and $P_{bad}$, respectively, that is, $P_{good} = \{P_{g1}, P_{g2}, \ldots, P_{gu}\}$, $P_{fail} = \{P_{f1}, P_{f2}, \ldots, P_{fu}\}$, and $P_{bad} = \{P_{b1}, P_{b2}, \ldots, P_{bw}\}$. The figure also shows possibility of transition from one state to another; the intersecting states are good candidates for a change of state. A packet in good state may be changed to bad state. Similarly, a bad state packet may become a fail state packet. Fail state never converts to good or bad until unless retransmission takes place, a solution proposed in our work.
A packet transmitted from source node reaches successfully to the destination node via intermediate nodes is assumed to be in **good** state. A packet dropped but not rejected at any intermediate node due to congestion or becomes lossy due to channel noise is considered to be in **bad** state and it may change its state to either **good** (or **fail**) depending upon its success (or failure) of delivery at the destination node.

The existing routing protocols retransmit packets dropped at intermediate nodes $N_{re}$ times but not more than a predetermined value $N_{n}^{\text{max}}$ forcing transition of one state to another state. The possible state transitions are represented in Table 5.1.

**Table 5.1: Packet State Transition Table**

<table>
<thead>
<tr>
<th>Present State</th>
<th>Number of Retransmission</th>
<th>Outcome</th>
<th>Next State</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>good</strong></td>
<td>$0 \leq N_{re} \leq N_{n}^{\text{max}}$</td>
<td>Transmitted Successfully</td>
<td><strong>good</strong></td>
</tr>
<tr>
<td><strong>bad</strong></td>
<td>$0 &lt; N_{re} \leq N_{n}^{\text{max}}$</td>
<td>Transmitted Successfully</td>
<td><strong>good</strong></td>
</tr>
<tr>
<td><strong>bad</strong></td>
<td>$0 &lt; N_{re} &gt; N_{n}^{\text{max}}$</td>
<td>Transmitted Unsuccessfully</td>
<td><strong>fail</strong></td>
</tr>
</tbody>
</table>
Flow of Packets in *good, bad, and fail* States

Flow of packets between source and destination nodes with respect to time is shown in Fig. 5.2. A source node ‘S’ transmits a sequence of video packets $P_1, P_2, P_3, \ldots$ via intermediate nodes to a destination node ‘D’. The packets $P_1$ and $P_2$ are successfully received at the node ‘D’, therefore, these packets are in *good* state, for every such packet the destination node generates *ACK* packets which are received at the source node. Packet $P_3$ is dropped at an intermediate node either due to high congestion or noise at some instant of time, say, ‘$t$’ and being in retransmission is in *bad* state. The state of $P3$ changes to *fail* as it exhausts all its retransmission attempts and a route error (*RERR*) is sent from the intermediate node to the source node. When a *RERR* packet reaches at the source node, it restarts the route discovery process by sending *RREQ* packet. Packets $P_4, P_5, \ldots$ following $P_3$ are also in *bad* state as they are generated after $P_3$ and may be lost for the same reason(s).
5.3 Distortion Estimation Based on Packet States

The *fail* packet states create distortion in the video reproduction. The states are governed by congestion and noise in random fashion; probability of packet loss under aforesaid states is studied in following subsections.

**Packet Loss Probability due to Congestion**

A relationship between probability of $n^{th}$ packet loss and packet state probability is established in the subsequent paragraphs. The packet states are not equally likely because of the dynamic channel conditions affected by congestion. The latter also affects transition from one state to other. At any instant of time a packet may be in any of the three states and may get lost. Therefore, the probability, $\rho(n)$ of loss of $n^{th}$ packet, in general, is a conditional probability and is given by Eq. (5.1)

$$\rho(n) = [\rho_g \cdot p_g(n) + \rho_f \cdot p_f(n) + \rho_b \cdot p_b(n)]$$  \hspace{1cm} (5.1)

where $\rho_g$, $\rho_f$, and $\rho_b$ are the probabilities of packet loss of $n^{th}$ packet in *good*, *fail* and *bad* states, respectively. Similarly, $p_g(n)$, $p_f(n)$, and $p_b(n)$ are the probabilities of being $n^{th}$ packet in corresponding states. But by definition, $\rho_g = 0$, $\rho_f = 1$ and $0 \leq \rho_b \leq 1$. Substituting these values in Eq. (5.1), the equation becomes:

$$\rho(n) = p_f(n) + \rho_b \cdot p_b(n)$$  \hspace{1cm} (5.2)

**Packet State Probability under Congestion and Noise**

Congestion (*cng*) is said to occur if the packet queue length exceeds the buffer size ($B$) and depends on packet arrival rate ($\lambda$) and packet forward rate ($\mu$). Let $P(cng)$ be the probability of congestion occurrence and $E_0 \equiv N_{re} > N_{n}^{max}$ indicating the event of retransmission. Then probability of a packet being in *good*, *bad*, or *fail* state is given by Eq. (5.3), eq. (5.4), or eq. (5.5), respectively.
Thus, in terms of $\lambda$ and $\mu$, the probability, $P(cng)$ from queuing theory, becomes as in Eq. (5.6).

$$P(cng) = P \left( \frac{\lambda \left[ 1 + \theta \left( \frac{\mu}{\theta} \right)^{S+1} - (\theta + 1) \left( \frac{\mu}{\theta} \right)^{S} \right]}{(\mu - \lambda) \left[ 1 - (\frac{\mu}{\theta})^{S+1} \right]} > B \right)$$  \hspace{1cm} (5.6)

In a similar fashion, the effect of channel noise is obtained and explained next. A channel is said to be noisy, if its signal strength is affected due to fading and interference experienced by nodes. Signal to noise ratio (SNR) is used as a parameter to calculate level of noise in the channels. Packet loss probability $\rho'(n)$ of $n^{th}$ packet and its probability of being in one of the three states are given by Eq. (5.7), Eq. (5.8), Eq. (5.9) and Eq. (5.10) respectively:

$$\rho'(n) = p_f'(n) + p_b'(n)$$  \hspace{1cm} (5.7)

$$p_g'(n) = [1 - P(noise)]$$  \hspace{1cm} (5.8)

$$p_f'(n) = P[E_0|noise]$$  \hspace{1cm} (5.9)

$$p_b'(n) = P(noise)$$  \hspace{1cm} (5.10)

where $\rho_g'$, $\rho_f'$, $\rho_b'$ represent the packet loss probabilities of $n^{th}$ packet in good, fail and bad state due to noise, respectively, and $p_g'(n)$, $p_f'(n)$, and $p_b'(n)$ are the probabilities of being $n^{th}$ packet in good, fail and bad state, respectively.
Packet Loss Probability

Proposed routing method uses aforesaid packet states to make routing decisions. In this Section the effect of congestion and noise on packet loss is formulated in terms of packet loss probabilities, $\rho_g$, $\rho_f$, $\rho_b$, $\rho'_g$, $\rho'_f$, and $\rho'_b$ which are based on number of retransmission attempts completed by a packet.

a. Calculation of $\rho_f$ and $\rho_b$

Each time a packet fails to complete its transmission, either it is not delivered successfully to the destination or the corresponding ACK packet has failed to reach the source node. Further, in the proposed routing algorithm a packet in bad state is retransmitted till it is delivered successfully, converting to good state, or get lost after $N_n^{max}$ attempts – the condition of the fail state. The routing algorithm synchronizes retransmissions involved in bad and fail states with following events:

I. The ‘attempts of retransmission’ of $n^{th}$ packet in bad state is termed as event, $E_{data}$ and

II. ‘completion of $N_n^{max}$ retransmission attempts’, leading to fail state of the packet, represents event, $E_{re}$.

III. Events $E_{data}$ and $E_{re}$ jointly confirm failure of the packet under consideration, which is termed as event $E_0$.

The probabilities $\rho_f$ and $\rho'_f$ of packet loss in fail state obtained in Eq. (5.2) earlier are therefore, dependent on these events.

From (I), probability of $N_n^{max}$ retransmissions, assuming equal probability of packet transmission failure, $P_{data}$, the probability of event, $E_{data}$ is expressed as in Eq. (5.11)
Chapter 5  A Distortion Aware Novel Routing Protocol for Video Transmission in MANETs

\[ P(E_{data}) = P_{data} \times P_{data} \ldots \text{upto } N^{max}_n \text{ times } = (P_{data})^{N^{max}_n} \]  \hspace{1cm} (5.11)

Also from above, the probability \( P(E_{re}) \) is a function of either transmission failure or ACK failure, and maximum number of attempts, \( N^{max}_i \), that is,

\[ P(E_{re}) = [P_{data} + (1 - P_{data}) \cdot P_{ACK}]^{N^{max}_n} \]  \hspace{1cm} (5.12)

where \( P_{ACK} \) is the probability of failure of ACK packet.

And

\[ P(E_0) = P(E_{data} | E_{re}) \]  \hspace{1cm} (5.13)

Using Eq. (5.11) and Eq. (5.12), \( P(E_0) \) is represented as

\[ P(E_0) = \frac{(P_{data})^{N^{max}_n}}{[P_{data} + (1 - P_{data}) \cdot P_{ACK}]^{N^{max}_n}} \]  \hspace{1cm} (5.14)

The probability of loss of \( n^{th} \) packet in \textit{fail} state, is the conditional probability of packet loss under congestion, using Bayes’ rule, is given as

\[ \rho_f = P(E_0 | cng) = \frac{p(cng|E_0)P(E_0)}{P(cng)} \]  \hspace{1cm} (5.15)

Equations eq. (5.6), eq. (5.14), and eq. (5.15) yield eq. (5.16).

\[ \rho_f = \frac{(P_{data})^{N^{max}_n}}{P_{data} + (1 - P_{data}) \cdot P_{ACK}} \]  \hspace{1cm} (5.16)

From definition of \textit{bad} state and \( E_{data} \), Eq. (5.17) is obtained.

\[ \rho_b = P(E_{data}) \]  \hspace{1cm} (5.17)
The probability of $n^{th}$ packet loss under congestion in Eq. (5.2), is expressed in Eq. (5.18) using Eq. (5.4), Eq. (5.5), Eq. (5.16), and Eq. (5.17)

$$P(n) = \left[ \frac{(P_{data})^{N_{n}^{max}}}{\left[ \frac{\lambda_j + B\left(\frac{\lambda_j}{\mu_j} \right)^{B+1} - (B+1)\left(\frac{\lambda_j}{\mu_j} \right)^B}{\mu_j} \right] > B} \right] \cdot P(N_r > N_{n}^{max}) +$$

$$\left[ \frac{(P_{data})^{N_{n}^{max}}}{\left(\mu_j - \lambda_j \right)^{B+1}} \cdot \left( \frac{\lambda_j + B\left(\frac{\lambda_j}{\mu_j} \right)^{B+1} - (B+1)\left(\frac{\lambda_j}{\mu_j} \right)^B}{\mu_j} \right) > B \right) \right]$$

(5.18)

b. Calculation of $\rho_f^{' \beta}$ and $\rho_b^{' \beta}$

In the same way as in congestion, the probabilities, $\rho_f^{' \beta}$ and $\rho_b^{' \beta}$ in noisy channels are obtained and shown in Eq. (5.19) and Eq. (5.20), respectively.

$$\rho_f^{' \beta} = \frac{(P_{data})^{N_{n}^{max}}}{P(SNR_{ij} > SNR_{thresh})}$$

(5.19)

$$\rho_b^{' \beta} = (P_{data})^{N_{n}^{max}}$$

(5.20)

Using Eq. (5.8), Eq. (5.9), Eq. (5.19), and Eq. (5.20), Eq. (5.6) becomes as follows:

$$P'(n) = \left[ \frac{(P_{data})^{N_{n}^{max}}}{P(SNR_{ij} > SNR_{thresh})} \right] + (P_{data})^{N_{n}^{max}} \left[ P(SNR_{ij} > SNR_{thresh}) \right]$$

(5.21)

The distortion model is developed to evaluate video quality in the noisy and congested channels from the loss probabilities formulated in the preceding section.
Distortion Modeling

Video quality in MANETs is dependent on number of good state packets reaching at the destination node and level of error in them. As packets go to fail state due to channel conditions, the number of packets in good state decreases producing distortion in the video. An analysis is made for congestion induced distortion and noise induced distortion based on the packet loss probability. For which three possible cases of network parameters - congestion and noise, based on their absence and presence, are considered:

Table 5.2: Three Cases of Channel Parameters

<table>
<thead>
<tr>
<th>Cases</th>
<th>Congestion</th>
<th>Noise</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.</td>
<td>0 (absent)</td>
<td>1 (present)</td>
</tr>
<tr>
<td>2.</td>
<td>1 (present)</td>
<td>0 (absent)</td>
</tr>
<tr>
<td>3.</td>
<td>1 (present)</td>
<td>1 (present)</td>
</tr>
</tbody>
</table>

Primarily distortion is caused by packet drops due to congestion or / and packet errors because of noise in the channel as defined by the cases given in Table 5.2. Therefore, the end to end distortion \( D(n) \) is summation of congestion induced distortion, \( D_{cng} \) and noise induced distortion, \( D_{noise} \) as given in Eq. (5.22).

\[
D(n) = D_{cng} + D_{noise} \tag{5.22}
\]

In addition, \( D_{cng} \) can be viewed as the summation of the distortion caused due to a packet’s failure and success of its dependent packets to reach at the destination:

\[
D_{cng} = P(n)(D_n^d - D_n^*) + \sum_{d=1}^{z}(1 - P(n)).D_d \tag{5.23}
\]

The first term of Eq. (5.23) specifies the impact of distortion caused due to loss of \( n^{th} \) packet by congestion. The second term quantifies the effect of error propagation over its dependent packets. Where \( D_n^d \) is the distortion of the video reconstructed
from all packets except both packet ‘n’ and its dependent packets, $D_n^*$ is the distortion of the video reconstructed from all packets except only the dependents of $n^{th}$ packet, $D_d$ is the loss-distortion of packet ‘n’ due to preceding packet (say $(n-1)^{th}$ packet), respectively.

Similarly, $D_{\text{noise}}$ can be calculated as:

$$D_{\text{noise}} = P'(n)(D_n^d - D_n^*) + \sum_{d=1}^{k}(1 - P'(n)).D_d$$  (5.24)

Using (5.23), and (5.24), the channel distortion is obtained and given as in (25):

$$D(n) = \left\{P(n)(D_n^d - D_n^*) + \sum_{d=1}^{k}(1 - P(n)).D_d\right\} + \left\{P'(n)(D_n^d - D_n^*) + \right.$$  

$$\left. d=1 \right. \left. k1 \right. - P'n.Dd$$  

$$\left. d=1 \right. \left. k \right. \left. \left. 1 \right. \right. \left. 1 \right. - P'(n))\right]\right\}$$  (5.25)

$$= [(D_n^d - D_n^*)\left\{P(n) + P'(n)\right\} + D_d(\sum_{d=1}^{k}(1 - P(n)) + \sum_{d=1}^{k}(1 - P'(n)))]$$ (5.26)

A routing protocol DAR is proposed using Eq. (5.26) wherein channel induced distortion is determined at the intermediate nodes and decision making is done at the time of route discovery.

### 5.4 Propose Routing Protocol

Detection of dynamic channel conditions in MANETs has always been a challenging issue. Moreover, protocols reported for solving the associated problems do not provide required QoS. Periodic broadcasting of beacon packets [11] and modification of packet structure are used in most of the traditional mechanisms of gathering network parameters that make an already congested network more congested. The proposed cross-layer based protocol detects channel conditions in route discovery phase (Procedure 5.1) and chooses a route which supports better QoS. In our routing protocol instead of broadcasting beacon packets the mobile nodes are assumed to be
smart enough to take decisions based on the distortion criteria to forward or discard a \textit{RREQ} packet without increasing the payload. Thus, it relieves the source node from the responsibility as well as avoids additional control packet flow and packet payload. A source node (\textit{s\_node}) willing to transmit video data to the destination node (\textit{d\_node}) starts with broadcasting of Route Request (\textit{RREQ}) packets to its neighboring nodes (\textit{n\_node}). On receiving \textit{RREQ} packet each \textit{n\_node} (or \textit{d\_node}) calculates distortion $D(n)$ using (5.26) and compares it with a distortion threshold value say, $D_{th}$. If the value of $D(n)$ is greater than or equal to the $D_{th}$ then that node sends a Route Error Packet (\textit{RERR}) to \textit{s\_node} and terminates route discovery process otherwise; \textit{n\_node} establishes a reverse route to \textit{s\_node} and then checks for a forward route to \textit{d\_node}. If the forwarding route exists then the node responds with Route Reply (\textit{RREP}) packet, on the other hand, it forwards \textit{RREQ} packet to its neighbors. This process continues until \textit{RREQ} packets reach \textit{d\_node}. On receiving \textit{RREQ} packets, \textit{d\_node} sends back \textit{RREP} packets to \textit{s\_node} via loop-free and disjoint reverse routes contained in the \textit{RREQ} packets. In worst situation, \textit{s\_node} may get \textit{RERR} packets from all its \textit{n\_node}'s, then \textit{s\_node} terminates route discovery process and starts a fresh one after some time. For successful \textit{RREP} packets, \textit{s\_node} selects a route with least distortion among all discovered routes. On establishing the route, \textit{s\_node} starts sending video packets. If a route breaks due to node mobility, the upstream node at which route break has occurred sends a \textit{RERR} packet to \textit{s\_node}. On receiving \textit{RERR} packet, \textit{s\_node} restarts route discovery process. The pseudo code of the procedure is presented next.
Chapter 5  A Distortion Aware Novel Routing Protocol for Video Transmission in MANETs

Procedure 5.1: Route Discovery Procedure of Distortion Aware Novel Routing Protocol

1. while no path exists to d_node do
2. s_node broadcasts RREQ packet to its n_nodes // route discovery process initiation
3. while n_node != d_node do
4. calculate D(n) using (5.26)
5. if \( D(n) \geq D_{th} \) then // compare channel distortion with threshold value
6. terminate route discovery process and send RERR packet to the s_node
7. else n_node sets a reverse path to the source
8. if a forward path to d_node exists then
9. send RREP packet to the s_node through reverse path
10. else forward copy of RREQ packets to all its n_node(s)
11. endif
12. endif
13. end while
14. d_node locates disjoint and loop free reverse path(s) to s_node
15. d_node sends RREP packets to its n_node(s)
16. while n_node != s_node do
17. n_node forwards RREQ packet
18. end while
19. If route is broken    //Route Maintenance
20. upstream node sends a RERR packet to s_node
21. s_node restarts route discovery process
22. endif

5.5 Results and Discussions

In order to study the effect of dynamics of channel environment on the video quality, transmission of QCIF “foreman” video over MANET is simulated. The corresponding input parameters are listed in Table 5.3. MyEvalVid is used to simulate video transmission and to generate random packet errors in NS2.

Observations on distortion when network is operated using the proposed routing algorithm and contemporary rate adaptive protocol are recorded to compare network
performance. The video distortion is estimated using Eq. (5.26). Discussion on the results for the three cases is presented in the subsequent sections.

### Table 5.3: Simulation Parameters used in DAR

<table>
<thead>
<tr>
<th>S. No.</th>
<th>Parameters</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>SNR Threshold (Th)</td>
<td>5 dB</td>
</tr>
<tr>
<td>2</td>
<td>Distortion Threshold ($\geq D_{th}$)</td>
<td>25dB</td>
</tr>
<tr>
<td>3</td>
<td>Buffer Size (B)</td>
<td>128 packets</td>
</tr>
<tr>
<td>4</td>
<td>Packet Size</td>
<td>512 Bytes CBR</td>
</tr>
<tr>
<td>5.</td>
<td>Packet Rate</td>
<td>10 Packets/s-70 Packets/s</td>
</tr>
<tr>
<td>6.</td>
<td>BER</td>
<td>$3 \times 10^{-5}$-1 $\times 10^{-3}$</td>
</tr>
<tr>
<td>7.</td>
<td>Transmission Range</td>
<td>250 m</td>
</tr>
<tr>
<td>8.</td>
<td>Link Capacity</td>
<td>1 Mbps</td>
</tr>
</tbody>
</table>

**Presence of Congestion only (case 1)**

The packets are transmitted at fixed SNR of 9 dB and varying rates 100 packets/s-700 packets/s to generate different levels of congestion. The effects of congestion in the channel on packets are shown in Fig. 5.3a and Fig. 5.3b, video quality in terms of PSNR is represented in Fig. 5.3c. The nature of packet and its impact on PSNR is examined as follows:

Sufficient bandwidth is available for packet transmission up to $\lambda = 200$ packets/s; channel is congestion free and no packet is found in *fail* state. In this range, PSNR of the transmitted video is observed greater than 40 dB in both the protocols. For packet arrival rate range greater than 200 packets/s but less than 400 packets/s, the packets are in *fail* state either due to congestion (exceeding time limit) or link breaks and lie between 1% and 2% (Fig. 5.3a) of the transmitted packets in our protocol and 8%-13% (Fig. 5.3b) in rate-adaption. The improvement in number of packets in *fail* state compared to rate-adaption technique is attributed to selection of low congestion path.
improving the video quality to 30 dB-37 dB (good as per ITU quality scale) against 24dB-30 dB (Fig. 5.3c).

When high number of packets is injected in the network the buffers at the nodes overflow and packets are dropped in large numbers (bad state) which are retransmitted for $\leq N_n^{max}$ times in our protocol. They change to either good or fail state as shown in Fig. 5.3a and Fig. 5.3b in terms of their percentage at 500 packets/s, 600 packets/s and 700 packets/s. In our protocol $\approx$7%, $\approx$11%, and $\approx$16% packets and $\approx$13%, 21%, and 46% packets in the existing protocol for the mentioned packet rates fail to reach to the destination within required time limit equivalent to 6%, 10%, and 30% conversion to good state. These packets contribute to better video quality generating PSNR difference of 5dB-7dB. The improvement in video quality establishes the superiority of our congestion-aware and controlled retransmission protocol over the rate-adaptive strategy.

![Packet State Vs. Packet Rate in DAR Protocol](image1.png)

![Packet State Vs. Packet Rate in Rate Adaptive Approach](image2.png)

**Presence of Noise only (case 2)**

It is a case of flow of packets in good state for all noise levels and with channel free of congestion. Fig. 5.4 compares the performance of our protocol with rate-adaptive strategy under this case. It is observed that both the protocols produce same video quality ($\approx$39dB-41dB) for SNR greater than 7 dB but the new protocol under
performs when noise is below 7 dB. The better results observed in the other protocol may be attributed to presence of its inherent error correction mechanism which has not been incorporated in our work.

**Congestion and Noise both are present (case 3)**

It is the most crucial case among all three cases. Packets in *fail* states are shown in Fig. 5.5a and Fig. 5.5b for the DAR protocol and rate adaptive strategy, respectively for both the channel parameters. On comparing these two figures, it is observed that percentage of packets lying in *fail* state in rate-adaptive protocol is high when packet rate is high. No packet is observed in *fail* state for packet rates lying between 100 packets/s and 400 packets/s and SNR > 3dB in our protocol and for packet rates between 100 packets/s and 200 packets/s and SNR > 3dB in rate-adaptive. The video quality is excellent for PSNR more than 37 dB at packet rates up to 400 packets/s and SNR > 7 dB in both the protocols. For packet rates more than 500 packets/s, deterioration of the order of 1dB-4dB is observed in the second protocol compared to ours, on the other hand, for SNR greater than 5dB distortion of 1dB-5dB is found in our protocol.
Our protocol provides improvement in PSNR between 2dB and 4dB at $\lambda$ greater than 500 packets/s for all SNR values under consideration. The rate-adaptive strategy presents better results ($\approx$5dB) in video quality at $\lambda$ less than 400 packets/s and SNR less than equal to 3dB. Both the protocols provide same video quality under low packet rate, less than 400 packets/s, and high SNR values, greater than or equal to 5 dB.

5.6. Conclusions

The MANET channels experience congestion and noise in a random fashion. Distortion of the video data has been studied for three cases of these channel parameters. A probabilistic distortion model is developed in this work for analyzing
the distortion and included in the route discovery phase in a novel way in the DAR protocol. The distortion model captures the affect of time varying noise and congestion on video quality. The protocol has also incorporated retransmission of packets in *bad* state in a controlled manner that has shown improvement in performance under high congestion. The effect of channel parameters – for three cases is analyzed and compared with an existing strategy. The results show that DAR protocol produces low distortion and an improvement of $\approx 5$dB in video quality when MANET is noise free but congested. The quality has improved by 2dB for packet rates greater than 500 packets/s and SNR below 5dB. Under low congestion i.e. less than 400 packets/s and high SNR, greater than 5dB, same video quality has been observed for both the protocols. However, poor results are obtained when channel noise is high and congestion is low since the protocol does not modify error correction mechanism inherently built in the simulator unlike the rate-adaptive strategy.