ADDIS ABABA UNIVERSITY
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SCHOOL OF ELECTRICAL AND COMPUTER ENGINEERING

Comparison and Enhancement of Routing Protocols for Video Transmission in Mobile Ad Hoc Networks

Thesis submitted to the School of Graduate Studies of Addis Ababa University in partial fulfillment of the requirements for the Degree of Master of Science in Electrical Engineering (Computer Engineering).

BY

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ABSTRACT

Due to bandwidth limitation and highly dynamic topology in Mobile Ad hoc Network systems, one of the major challenges is implementation of end-to-end quality of service support mechanisms. Transporting time-sensitive data over this network seriously affected if quality of service support mechanisms do not exist in the system. Hence, to transmit a video data to the end users with acceptable quality it is important to enhance the routing protocol to be able to select a route with the required metrics, such as bandwidth and delay metrics.

In first section of this thesis, performance comparison was performed on the existing routing protocols to select suitable protocol for multimedia or video application. The selection was performed by comparing mainly the results of average end to end delay and packet delivery ratio. In the second section, the selected routing protocol is enhanced to incorporate quality of service mechanism on route selection process. Packet differentiation is enabled at MAC layer and also data packet life span controlling mechanism is implemented.

For performance analysis of the protocols and proposed enhancement, we simulate network using Network simulator-2 and for pre and post processing (for encoding, decoding and testing the received video data) we apply Evalvid.

Simulation results show that the proposed protocol performs well in both end to end delay and packet delivery ratio parameters. In general, we can conclude that adding quality of service metrics to routing protocol is mandatory and important to receive better video quality."

Keywords-Quality-of service, routing protocols, MANETs, NS-2, Evalvid

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ACRONYMS

AAC	Access Admission Control
AC	Access Categories
ACK	Acknowledgement
AIFS	Arbitration IFS
AODV	Ad Hoc on-Demand Distance Vector Routing Protocol
AODV-D	Delay Based Ad Hoc on-Demand Distance Vector Routing Protocol
BcastID	Broadcast Identifier
BE	Best effort
BW	Bandwidth
CAC	Call Admission Control
CBR	Constant Bit Rate
CC	Contention Count
CP	Contention Period
CFP	Contention Free Period
CSMA/CA	Carrier Sensing Multiple Access/Collision Avoidance
CSMA/CD	Carrier Sensing Multiple Access/Collision Detection
CTS	Clear To Send
CW	Contention Window
CWmax	maximum contention window size
CWmin	minimum contention window size
DCF	Distributed Coordinate Function
DestID	Destination Identifier
DestSeqNum	Destination Sequence Number
DiffServ	Differentiated Services
DSCP	Differentiated Service Code Point
DIFS	DCF inter-frame spacing
DSDV	Destination Sequenced Distance-Vector routing protocol
DSR	Dynamic Source Routing
DSSS	Direct-Sequence Spread-Spectrum

EDCF	Enhanced Distributed Coordinate Function
FQMM	Flexible QoS Model for MANET
HCF	Hybrid Coordination Function
IFS	Inter-frame Spacing
IntServ	Integrated Service
IP	Internet Protocol
ISP	Internet Service Provider
MAC	Medium Access Control
MANET	Mobile Ad hoc Networks
MPEG	Moving Picture Experts Group
NAV	Network Allocation Vector
NOL	Normalized Overhead Load
NS2	Network Simulator 2
OSI	Open System Interconnection
OTcl	Object-Oriented Tool Command Language
PCF	Point Coordinate Function
PDR	Packet Delivery Ratio
QoS	Quality of Service
RERR	Route Error
RREP	Route Reply
RREQ	Route Request
RSVP	Resource Reservation Protocol
RTS	Request To Send
SLA	Service Level Agreement
SIFS	Short inter-frame spacing
SrcID	Source identifier
T_P	Observing Time Period

1. INTRODUCTION

1.1 Background

A Mobile Ad Hoc Networks (MANETs) is an autonomous collection of distributed mobile users [1]. Every host in a MANET works as a source and a sink, and also relays packets for other hosts and is thus a router as well. MANETs have similar characteristics to other wireless communication networks, which are mainly attributed to the wireless channel's properties. A wireless channel is error-prone, which means that link bandwidth and packet delay are unpredictable due to interference and fading. Besides this common characteristic, MANETs have their own features: they are infrastructure less; they utilize multi-hop routing; they support a dynamic network topology; the nodes are energy constrained; the bandwidth is limited; and self-administering [1]. Therefore, many of the widely used network protocols cannot directly be applied to MANETs.

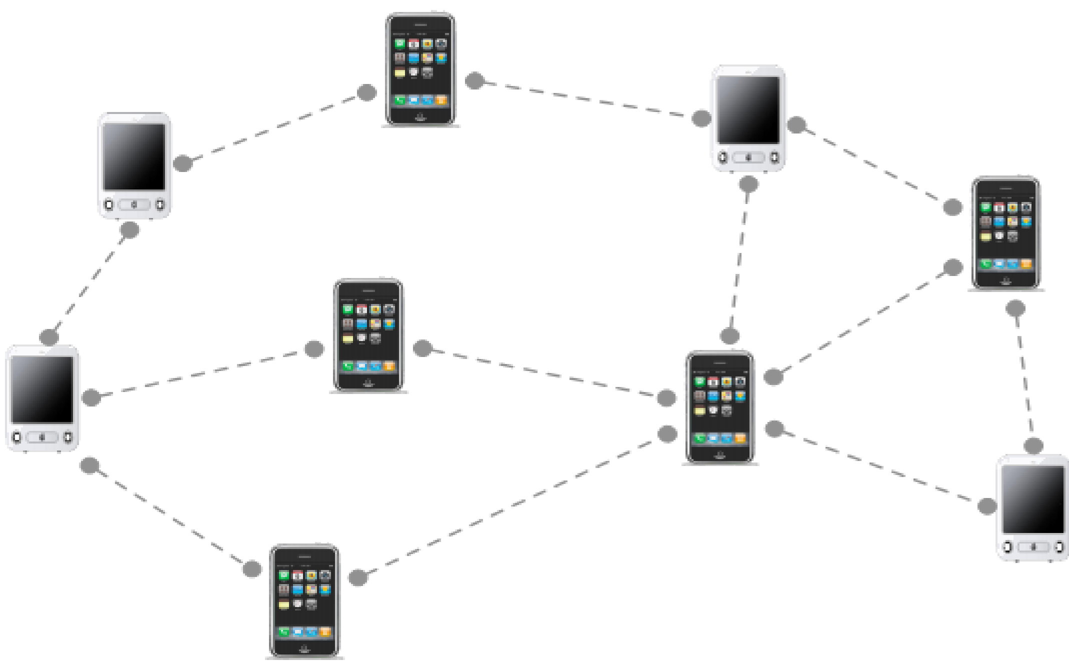

Figure 1.1 Mobile Ad Hoc Networks [1].

MANET is a distributed network that does not require centralized control, and every host works not only as a source and a sink but also as a router. This type of dynamic network is especially useful for military communications or emergency search and-rescue operations, where an infrastructure cannot be supported. Furthermore, the simplicity of building an ad hoc network enables sharing data in a meeting or in unwelcoming landscape conveniently. But enabling multimedia applications as video and audio communication in MANETs requires quality of service (QoS) support, since these data requires enough bandwidth and minimum delay [2].

The challenges of supporting QoS in ad hoc networks are how to reserve bandwidth and how to guarantee the specified delay for real-time application data flows. For wireless transmissions, the channel is shared among neighbors. Therefore, the available bandwidth depends on the neighboring traffic status, as does the delay. Due to this characteristic, supporting QoS cannot be done by the host itself, but cooperation from the hosts within a node’s interference range is needed. This requires an innovative strategy to coordinate the communication among the neighbors in order to support QoS in MANETs [2].

1.2 Statement of the Problem

Unlike wired communication, in mobile ad hoc networks the resources are limited and links are created and broken dynamically due to mobility. In addition to the above mentioned characteristics, to transmit video data from source node to destination with acceptable video quality the path should be stable and need to have enough bandwidth.

To fulfill these requirements the routing protocol needs to incorporate a QoS mechanisms to select a route with required bandwidth.

1.3 OBJECTIVE

1.3.1 General Objective

To compare the existing routing protocols, select the best protocol and enhance the protocol to incorporate QoS parameter during route discovery.

1.3.2 Specific Objectives:

- To compare the existing routing protocols and to select the appropriate one for video application. In our architecture Ad hoc On-Demand Distance Vector protocol (AODV) is selected.
- To modify the selected AODV routing protocol to incorporate a QoS metrics in route discovery phase. Route selection method is changed from shortest path (number of hops) metrics to available node’s bandwidth (minimum route bandwidth).
- To modify the original MAC 802.11b to enable packet differentiation at MAC layer, to allow the channel access depending on priority level of packets.
- To modify the link layer to control the life span of data packets not to exceed 200ms, since packets that arrived later than this have no effect on the received video quality.

2. ROUTING PROTOCOLS AND RELATED WORK

2.1 Introduction to Routing

Routing is the process in which a route from a source to a destination node is identified [3]. In order to facilitate communication within MANET, a routing protocol is used to discover routes between nodes. The primary goal of such a routing protocol is to ensure correct and efficient route establishment between a pair of nodes so that messages are delivered in a timely manner. The conventional routing protocols for mobile ad hoc networks have been designed to find a path, between two communicating nodes, without taking into account its quality. These protocols discover and maintain paths that are generally established according to the minimum hop-count metric [3].

2.2 Classification of Routing Protocols

Routing protocols for mobile ad hoc network categorized on the basis of how routing information is acquired and updated by mobile nodes and can be classified into three main categories: Proactive routing or Table driven routing, Reactive routing or On demand routing and Hybrid routing.

In proactive routing protocol, nodes of ad hoc network continuously evaluate routes to all reachable nodes and attempt to maintain consistent, up-to-date routing information. Therefore, a source node can get a routing path immediately if it needs one and all nodes need to maintain a consistent view of the network topology. When a network topology changes, respective updates must be propagated throughout the network to notify the change and mobile nodes proactively update network state and maintain a route regardless of whether data traffic exists or not, the overhead to maintain up-to-date network topology information is very high [3].

In a reactive routing protocol, routing paths are searched only when needed. The discovery procedure terminates either when a route has been found or no route available after examination for all route permutations. In comparison to proactive or table driven routing protocols this routing strategy has very low computational and memory requirements and hence are very much suitable for ad hoc networks.

Hybrid routing protocols are proposed to combine the qualities of both proactive and reactive routing protocols and overcome their drawbacks. For instance, using a reactive protocol, where the route discovery is done only when a communication is requested, and caching the available routes in a case of a link failure, which is an aspect of proactive routing. Hybrid protocols take advantage of both reactive and proactive protocols, but may require additional hardware, such as GPS, integrated into the communication device [4].

	REACTIVE	**PROACTIVE**
Overhead	Low	High
Memory requirement	Low	High
Cope with mobility	Good	Bad
Purpose	Relatively high mobility	Low mobility

Table 2.1: Reactive versus Proactive protocols [4]

In this paper three most popular routing protocols are used for performance evaluation, AODV, DSDV and DSR. AODV and DSR are Reactive (On demand) whereas DSDV is Proactive (Table driven) Routing protocol. The detailed route discovery operation of these protocols is described below.

2.2.1 Dynamic Source Routing (DSR)

The Dynamic Source Routing (DSR) is a reactive routing protocol which utilizes source routing algorithms. In source routing algorithm, each data packet contains complete routing information to reach its target. Additionally, in DSR each node uses caching technology to maintain route information that it has accumulated [4]. Every Reactive routing scheme follows two phases mainly Route Discovery and Route Maintenance in which a route is discovered dynamically and then due to frequent topology changes the route has to be maintained in order to ensure uninterrupted communication.

2.2.1.1 The Route Discovery Phase

In this phase when a source node wishes to send a packet, it firstly consults its route cache. If the required route is available, the source node includes the routing information inside the data packet and sends it to the destination. If the required route is not available, the source node initiates a route discovery operation by broadcasting Route Request (RREQ) packets. RREQ packet contains addresses of the source, the target and a unique number to identify the request. Receiving a route request packet, a node checks its route cache. If the node doesn't have routing information for the requested destination, it appends its own address to the route record field of the RREQ packet. Then, the request packet is forwarded to its neighbors. If the RREQ packet reaches the destination or an intermediate node has routing information to the destination, RREP packet is generated. When the RREP packet is generated by the destination, it contains addresses of nodes that have been traversed by the RREQ packet. Otherwise, the route reply packet includes the addresses of nodes the route request packet has traversed concatenated with the route in the intermediate node's route cache [4].

When route discovery is initiated, the sending node saves a copy of the original packet in the local buffer referred to as Send buffer. The send buffer contains a copy of each packet that cannot be transmitted by the node due to the absence of route to the packet's destination. Each packet in this buffer is stamped with a time to indicate the time when it was placed and is discarded from this buffer after a certain time interval in order to prevent the send buffer from overflowing.

When packets sit in this buffer the node should occasionally initiate a new route discovery for the packet's destination. The node must limit the rate of retries because there is a chance that the destination may no longer be in the transmission range. Each new route discovery contributes to a large number of route request packets which contributes to the overhead in the network. Hence there is a retry limit beyond which the packets are buffered in the send buffer until a route reply is received [3-4].

2.2.1.2 The Route Maintenance Phase

Each source or intermediate node forwarding a packet is responsible for its receipt at the next hop along the source route. Each packet is retransmitted up to a certain maximum number of attempts until its receipt is confirmed. This acknowledgement is provided by the MAC protocol. When the packet retransmission reaches a certain maximum number of retries and receives no confirmation, the node starts sending a Route Error message to the source identifying the link over which the packet transmission failed. On receipt of this message the source first deletes this link from its cache and initiates a route discovery to the destination.

DSR has increased traffic overhead by containing complete routing information into each data packet deteriorating its routing performance [4]. DSR has increased traffic overhead by containing complete routing information into each data packet deteriorating its routing performance.

2.2.2 Destination-Sequenced Distance-Vector Routing (DSDV)

Destination-Sequenced Distance-Vector Routing (DSDV) is a table-driven routing protocol that used in adhoc mobile networks. In DSDV, each node maintains a next-hop table, which it exchanges with its neighbors. In each data packet sent during a next-hop table broadcast or incremental updating, the source node appends a sequence number. This sequence number is propagated by all nodes receiving the corresponding distance-vector updates, and is stored in the next-hop table entry of these nodes.

A node, after receiving a new next-hop table from its neighbor, updates its route to a destination only if the new sequence number is larger than the recorded one, or if the new sequence number is the same as the recorded one, but the new route is shorter. In order to further reduce the control message overhead, a settling time is estimated for each route. A node updates to its neighbors with a new route only if the settling time of the route has expired and the route remains optimal [4].

2.2.3 Ad Hoc on-Demand Distance Vector Routing Protocol

From the results of end to end delay and packet delivery ratio of section one, AODV is selected as a framework and here a detailed explanation is provided for this routing protocol. AODV uses the broadcast route discovery mechanism from DSR (Dynamic Source Routing) and uses the sequence number feature of DSDV (Destination Sequence Distance Vector) in order to maintain loop freedom and freshness of routes. The combination of these features leads to a protocol that uses bandwidth efficiently as well as adapts faster to dynamic topology changes.

2.2.3.1 Route Discovery

A route discovery is initiated each time a source node wants to communicate with another node for which it has no information in its routing table. The source starts a route discovery by broadcasting a route request message to all the nodes in its transmission range. This message contains fields which identify the source as well as the destination along with a hop count field. The combination of source address and broadcast id uniquely identify a route request. A new Route Request (RREQ) packets increments the broadcast id field. Each node in the network either satisfies the RREQ to send back a Route Reply (RREP) or just rebroadcasts the RREQ to its neighbors after incrementing the hop count. When an intermediate node receives an RREQ that it has seen before, then it is simply discarded without rebroadcasting.

SourceID	*DestID*	*SrcSeqNum*	*DestSeqNum*	*BroadcastID*	*Hop Count*

Figure 2.1 Format of the AODV RREQ packet

The Destination Sequence Number field contains the last Known sequence number provided by the destination, and the Originator Sequence Number the originator's own sequence number, typically incremented just before this packet is created.

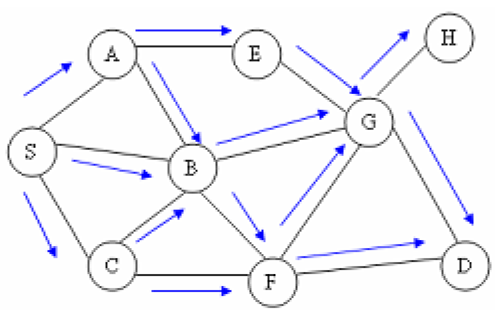

Figure 2.2 Flooding RREQ in AODV [4]

If the node receiving the RREQ has not heard this previously and it is not originated by itself, the node will set up a reverse route entry for the source node in its routing table. By storing this reverse route in the routing table, the node knows how to forward the RREP to the source if it received a RREP for this source node at a later time. If the entry for this source node already exists, this entry will be updated by refreshing the sequence number of the record. The sequence number is decided by choosing a larger one between the sequence number in RREQ and the one in the already existing route record. After processing with the reverse path, the node checks whether it is the destination node. If the node is an intermediate node and it may rebroadcast RREQ when it does not have a fresh enough route.

Intermediate node may send RREP to the source node if that the node has a valid route to the destination, the route is valid if the sequence number of the valid route in the route table is greater than or equal to the sequence number in the RREQ . The DestSeqNum is used to help find the latest route.

SourceID	DestID	DestSeqNum	Time out	Hop Count

Figure 2.3 Format of the AODV RREP packet

The destination node sends a RREP when it receives a RREQ. The highest Sequence number of the destination node is increased by 1 first. In the RREP message, the Destination Sequence Number field is filled by own highest Sequence No. of the destination node. The RREP is unicasted to the next hop towards the source of the RREQ. The hop count is incremented by one at each hop. In this way, when the RREP arrive the source, this hop count field is the distance in hops from the source to the destination.

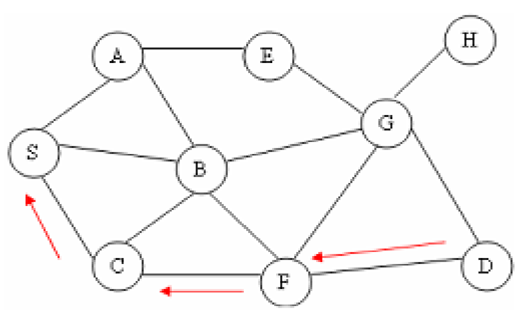

Figure 2.4: Route Reply in AODV [5].

2.2.3.2 Route Maintenance

When a node detects that a route to a neighbor no longer is valid, it will remove the routing entry and send a link failure message, Route Error (RERR) message to the neighbors that are actively using the route, informing them that this route is no longer valid. For this purpose AODV uses active neighbor list to keep track of the neighbors that are using a particular route. Then, these nodes propagate the RERR to their predecessor nodes. This procedure continues till the source node is reached. Once RERR is received by the source node, it will either stop sending the info or reinitiate the route discovery mechanism.

2.3 Quality of Service (QoS) in Ad Hoc Networks

To efficiently establish routes and deliver packets between mobile nodes with minimum communication overhead while ensuring high throughput and low end to-end delay in mobile ad hoc networks (MANETs), many ad hoc routing protocols have been deployed effectively for providing the shortest-path routing; thus packets are delivered with best-effort in a dynamically changing network topology. However, all routing protocols are designed without QoS considerations. Some applications may require a data delivery flow with QoS support, such as multimedia applications as voice or video communications with bandwidth and delay constraints. Quality of Service (QoS) that includes bandwidth and delay management is a set of service requirements to be met by the network while transporting a packet flow from source to destination [6].

The challenges of providing QoS are how to guarantee a flow of data packets along the same routing path and how to estimate available network resources for providing QoS requirements in the dynamic nature of an ad hoc wireless network. These are challenging because available network state information is naturally imprecise due to the node mobility and the limited bandwidth of a shared wireless channel. For instance, the available bandwidth of a link depends on the number of neighbor nodes in the limited wireless transmitting power range. The nodes are moving in and out of the range, which causes an unpredictable knowledge of the current link state and the established end-to-end path status [6].

The goals of QoS-aware routing differ from ad hoc routing protocols that don't provide QoS in two ways: (1) selecting a path in order to satisfy the required QoS by applications, and (2) allowing a node to be aware of the availability of network resources along the path in order to establish the routing connection based on the QoS requirements.

2.3.1 Limitations of Providing QoS in MANETs

Providing QoS guarantees are difficult in ad hoc wireless networks due to their unique characteristics as randomly distributed nodes that move completely independently. This means that providing QoS routing is problematic because it's difficult to be aware of information of the current network state. Constraints of providing QoS guarantees are due to the following characteristics of ad hoc wireless networks [7]:

- Dynamic network topology- Since the nodes in an adhoc wireless network are mobile, the network topology changes dynamically. However, this dynamic nature makes it difficult to provide QoS in an ad hoc wireless network. The network status is unpredictable because nodes may join, leave, and rejoin in an ad hoc wireless network, which means existing links may disappear and new links may be formed. Therefore, the dynamically changing network status is a major cause of the imprecise network state information (bandwidth, delay, loss rate, cost, etc.).
- Lack of centralized control – Unlike wireless LANs and cellular networks, ad hoc wireless networks do not have a central controller to coordinate the activity of nodes due to the randomly distributed nodes that perform as routers and hosts; these nodes can communicate with each other within their transmission range. However, the distributed nature of ad hoc wireless networks makes it difficult to provide QoS guarantees because network resources cannot be assigned in a predetermined way.
- Limited channel availability –Wireless channels may be unreliable due to the inherent error-proneness of wireless channels to such factors as interference from other transmissions, shadowing, and multipath fading effects. In addition, channel contention may exist due to the broadcast nature of wireless mediums without a central controller, which causes high packet collisions. These make it impossible for the providing QoS to guarantee a high packet delivery ratio or a link of available bandwidth.

2.4 Video Coding Concepts

Compression is the process of compacting data into a smaller number of bits. Video compression (video coding) is the process of compacting or condensing a digital video sequence into a smaller number of bits. Raw or uncompressed digital video typically requires a large bitrate and compression is necessary for practical storage and transmission of digital video. Data compression is achieved by removing redundancy, i.e. components that are not necessary for faithful reproduction of the data.

Video streaming places heavy requirement for bandwidth on a network based on its frame rate, resolution and bit rate. As these parameters increase, so does the bandwidth required to transmit the video file.

Moving Picture Experts Group version 4 (MPEG-4) is a standard method to transmit digital video and audio in a compressed format using less bandwidth than with the traditional analog method. MPEG-4 encoded video is formed by three types of frames: Intra-coded Pictures (I-Pictures), Predictive-coded Pictures (P-Pictures), Bidirectional -predictive-coded Pictures (B-Pictures) [6]. These three types of pictures are combined to form a GoP (Group of Pictures).

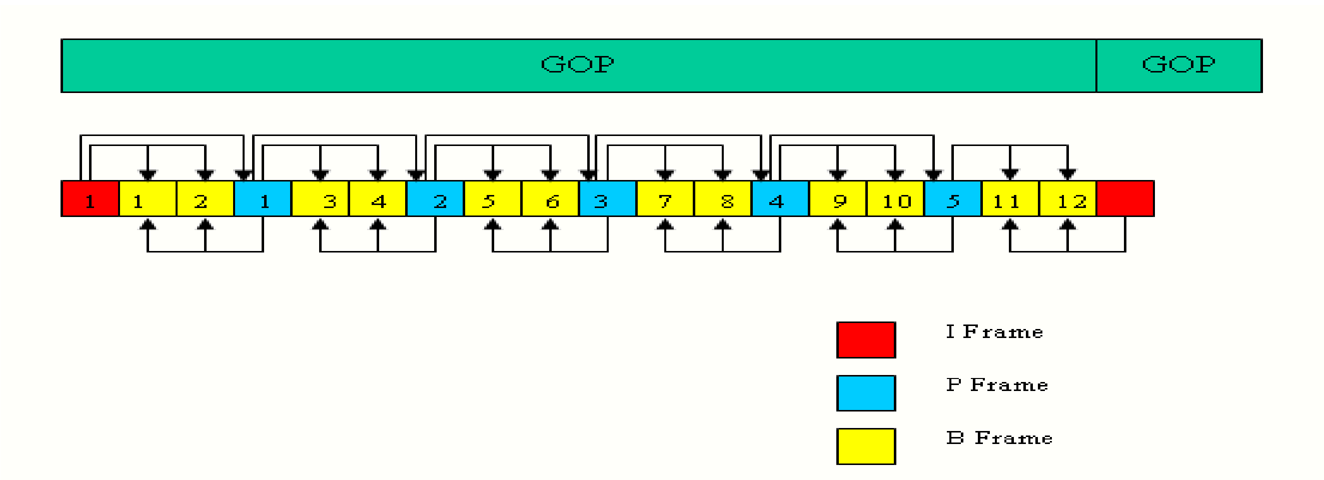

Figure 2.5 MPEG-4 frame structure [8].

I frames are the base layer and provide basic video quality. They carry the most important video information for the decoding process at the receiving side. GoPs could be decoded even if just I frames were present. The entire GoP would be lost if the corresponding I frame were not available at decoding time.

P and B frames provide enhancement layers, so that fine granularity and scalability can be achieved. They carry differential information from preceding or next frames. A typical GoP is formed by 15 frames. P frames and B frames might follow an I frame like [IBBPBBPBBPBB] to form a GoP.

2.4 Related Works

In this section we provided an assessment on the existing approaches in the provisioning of QoS in wireless ad-hoc networks. The existing approaches could be categorized mainly into the following groups: MAC QoS and QoS-aware routing.

- ### MAC QoS

The MAC approach provides QoS support at the media access control (MAC) layer. Radio channels are shared media, and can be shared differently to provide service differentiation for instance by assigning larger slots for higher priority packets. The 802.11 MAC protocol parameters, such as the Interframe spacing (IFS), Contention Window (CW), and Backoff Integer (BI) have been suggested for QoS support. Best-effort distributed MAC controllers are widely used in wireless ad-hoc networks. The IEEE 802.11 Distributed Coordination Function (DCF) is a good example of a best effort distributed MAC.

Recently, there have been a number of proposals to support service differentiation at the MAC layer using distributed control schemes. Enhanced Distributed Coordination Function (EDCF) is a growing IEEE 802.11 alternative that facilitates prioritized packet transmission [9].

In [10], Romdhaniet al. propose enhancements to the IEEE 802.11e technology to offer relative priorities by adjusting the size of the contention window (CW) of each traffic class, taking into account both applications requirements and network conditions. The performance enhancement obtained by IEEE 802.11E is experimentally approved in section 4.4.

- QoS-aware routing

QoS-aware routing considers the QoS dimension when performing route selection and packet scheduling. Embedding QoS in routing mechanisms can solve many of the problems faced during the QoS implementation on fixed wired networks. The QoS-aware routing approach is still in early research phase in the ad-hoc networks, and QoS routing is valuable in finding optimal QoS routes, opposite to other approaches.

In [11], an on demand delay based quality of service (QoS) routing protocol (AODV-D) protocol is proposed for QoS routing to ensure that delay does not exceed a maximum value for the MANETs. MAC layer channel contention information and number of packets in the interface queue are considered in addition to minimum hop criteria for route discovery in MANETs. AODV-D focuses on reducing the network delay but does not consider the bandwidth requirements of real time applications.

In [12] a QoS routing algorithm (QAODV) has been proposed as an extension to AODV-D. A weighted function of several parameters is used to select optimal routes and to provide support for QoS and fault tolerance. QAODV has also shown advantages in terms of throughput and delay.

In [13], Chen and Heinzelman modified the hello messages in the AODV routing protocol so that it carried bandwidth information of each node and its immediate neighbors. This information was then used to calculate the residual bandwidth due to second hop neighborhood interference.

In [14], another QoS routing protocol (MDSR-AODV) for MANETs with Link Stability Model is proposed. MDSR-AODV makes routing decisions by taking into account the characteristics of the channel. It finds paths with greater link stability factor in the route discovery phase. A long-lived path is preferred for data transfer among the available options by monitoring network topology changes through delay prediction the route maintenance is done.

In [15], Xue and Ganz propose a resource reservation-based routing and signaling algorithm (AQOR) that provides end-to-end QoS support in terms of bandwidth and delay.

Above-mentioned protocols used different techniques to increase the performance of the network. We have seen different load and delay reduction methods in the literature. Number of techniques for real time data transmission is proposed as well. Most interesting idea in QoS provision is by adding messages during the route discovery phase that carry QoS requirements. These message extensions are mostly about bandwidth requirements and delay threshold value. Almost every protocol in MANET is fully or partially adopting the technique of adding extension to RREQ message in order to provide QoS guarantees to the delay sensitive applications.

Even though the same route selection approach is followed in our protocol by adding extension to RREQ and RREP message in order to guarantee QoS metrics, our protocol integrated many additional features like packet differentiation, controlling packets life span in the network and real video data evaluation.

3. BANDWIDTH AWARE QOS-AODV ROUTING PROTOCOL

3.1 Architecture of the Proposed System

To transmit a video data and provide an acceptable video quality at the receiver the path need to acquire the required bandwidth for the data flow. So before transmitting the data to destination the application agent calculates the required bandwidth for the flow and broadcast RREQ with required bandwidth. To accomplish the proposed system first we have made an extensive experimental evaluation to select an appropriate routing protocol and we found that AODV is comparably best protocol to transmit real time data over MANET's due to its stability and lower end to end delay in case of high mobility and greater data rate. Detailed performance evaluation is explained in chapter four. The figure below shows the proposed routing protocol architecture and its main change for route selection, nodes bandwidth estimation and packet life span controlling.

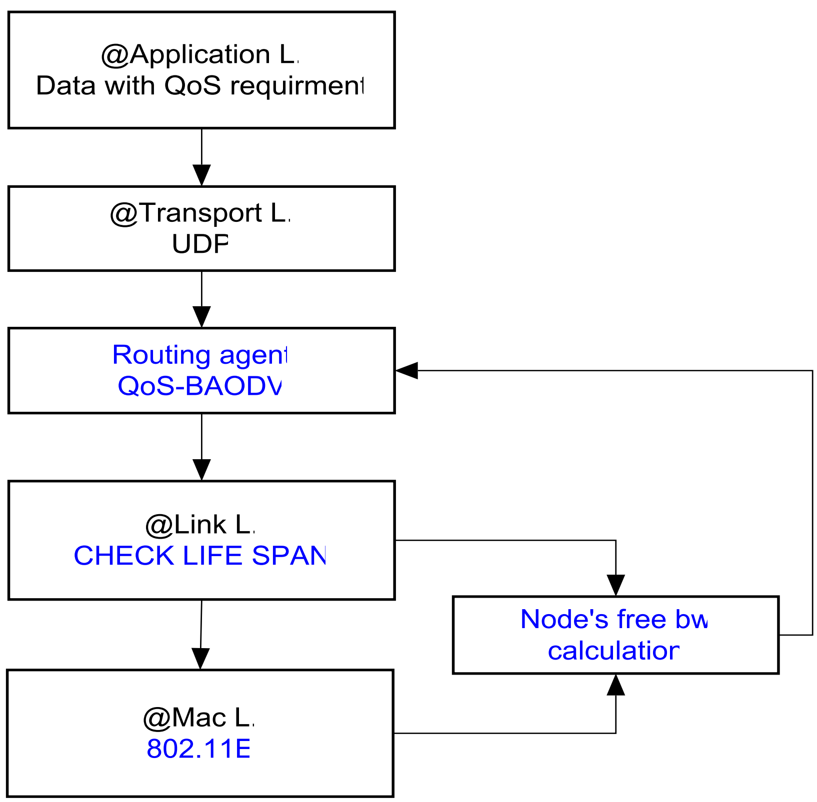

Fig. 3.1 General architecture of the proposed protocol

Then major changes on network, mac and data link layers have been made on the ordinary AODV routing protocol to incorporate QoS metrics. At the network layer route selection criteria is changed from hop count to node's available bandwidth. Data packet life span is checked at the LINK layer. The service differentiation should be completed in the MAC layer through a priority classifier. Estimation of a node's available bandwidth is performed in MAC layer by sensing the channel free and busy time.

3.2 Available Bandwidth Estimation

To offer QoS for the network architecture, the available end-to-end bandwidth along a route from the source to the destination must be known. The available end-to-end bandwidth is a concave metric, which determined by the bottleneck bandwidth of the intermediate nodes in the path. Therefore, estimating the end-to-end bandwidth can be simplified into finding the minimal residual bandwidth available among the nodes in that path. There are two standard methods for estimating bandwidth in MANETs.

The first one is channel 'Listen bandwidth estimation', in this method nodes listen to the channel and estimate the available bandwidth based on the ratio of free and busy times. The second method is called 'Hello bandwidth estimation' in this method every node to disseminate information about the bandwidth it is currently using in the Hello messages, and for a host to estimate its available bandwidth based on the bandwidth consumption indicated in the Hello messages from its neighbors [13].

Hello bandwidth estimation method incurs additional control message overhead to disseminate information to neighbors. This will consume additional bandwidth. Hence in our architecture to estimation node's locally available data rate we used channel listening method which is adopted from [13].

3.2.1 Listen Mode Bandwidth Estimation

In this method each node in the MANETs can determine its unconsumed or local bandwidth by passively listening network activities. This approach proposes to use the fraction of channel idle time based on the past history as an indication of local available bandwidth at a node. A node can observe the channel as either idle or busy.

The MAC detects that the channel is free if; network allocation vector (NAV) value is less than the current time, or Receive state is idle, or Send state is idle. The MAC claims that the channel is busy if; NAV sets a new value or Receive state changes from idle to any other state, or Send state changes from idle to any other state [16, 17].

The RTS and CTS frames contain a Duration/ID field that defines the period of time that the medium is to be reserved to transmit the actual data frame and the returning ACK frame. All nodes within the transmission range of either the source that transmits the RTS or the destination that transmits the CTS shall learn of the medium reservation and may save it in their NAV.

The NAV maintains a prediction of future traffic on the medium based on duration information that is announced in RTS/CTS frames [16, 17]. Figure 3.2 below illustrates the 802.11 RTS/CTS/DATA/ACK and NAV access mechanism.

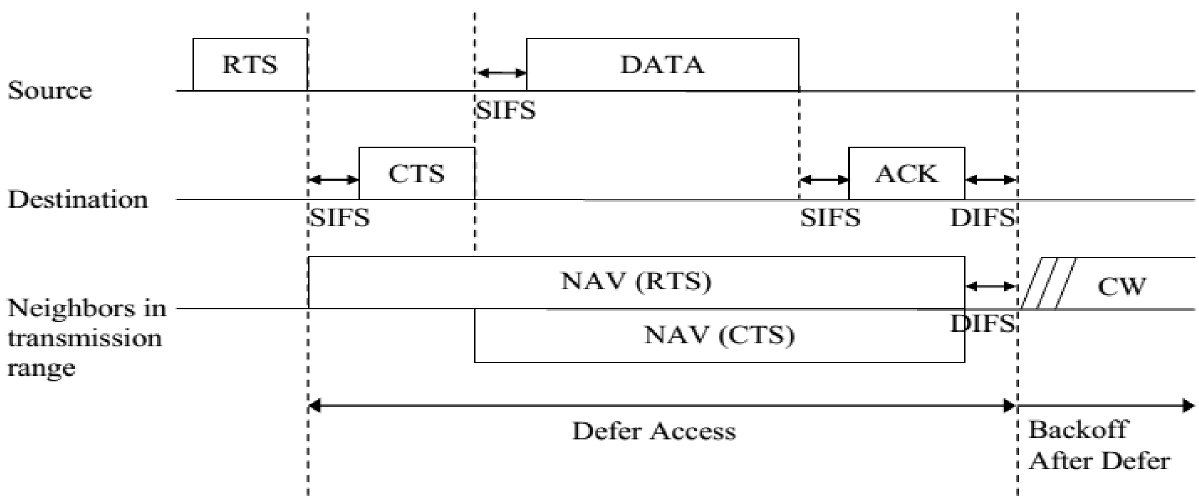

Figure 3.2 Illustration of the 802.11 RTS/CTS/DATA/ACK access mechanisms [17].

Idle time calculation requires estimation of channel busy time (BusyTime) within the specified time period (Tp) or observing time. Normally the medium is busy when transmitting or receiving the control messages (RTS, CTS, ACK) and the data frames [18].

$$\mathrm{BusyTime} = \mathrm{RcvTime} + \mathrm{SendTime} \ldots\ldots\ldots\ldots\ldots\ldots (3.1)$$

Where,

RcvTime – time consumed to receive control & data frame within T_P period.

SendTime - time consumed to transmit control & data frame within T_P period.

In both cases, either to transmit or receive a data frame, always control message is transmitted and received from source and destination stations. Hence the amount of time required for single data packet transmission is computed as [18]:

$$\mathrm{BusyTime} = \mathrm{Tc}_{\mathrm{msg}} + \mathrm{Tframe} \ldots\ldots\ldots\ldots\ldots\ldots\ldots (3.2)$$

Where,

T_{c_msg} - time consumed by the routing control messages like RTS, CTS, ACK.

T_{frame} - time needed for single data frame transmission.

Channel idle time (FreeTime) within the period Tp is calculated as shown in Equation below [18].

$$\mathrm{FreeTime} = \mathrm{Tp} - \mathrm{BusyTime} \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (3.3)$$

The Tp is the time that is used for observing the channel, in our simulation we used 1second. There is a tradeoff in the length of data rate calculation interval time. The longer the value of Tp, the more accurate the BW is. On the other hand, if we take smaller value for Tp, then the response for available data rate is faster [25]. By monitoring the amount of Free Time, during every period of time Tp, the free BW of a node is computed as:

$$\mathrm{FreeBandwidth} = \mathrm{FreeTime} \times \mathrm{ChannelCapacity} \ldots\ldots (3.4)$$

3.3 Modified Network Layer Packets Structures

Since the route selection metrics is changed from number of hops to nodes available bandwidth, on the network layer changes has been made to the packet structure of RREQ packet, RREP packet and also to routing table structure to incorporate the requested bandwidth for every data flow.

3.3.1 Modified Route Request Packet

When the source node receives a data to be transmitted, first it calculates the required bandwidth for the data flow and then calculates its own available bandwidth. Then if the source node has the required bandwidth it will broadcast RREQ packet with the QoS extension to its neighbors if not RREQ packet will not be generated. When a node receives a RREQ packet, it first checks if it has enough available bandwidth for the request. A node which does not satisfy the bandwidth will discard the RREQ packet. If it has the required bandwidth, a reverse route entry with required bandwidth is created and then rebroadcasts the RREQ packet, until the RREQ packet reaches the destination node.

Unlike the original AODV, if an intermediate node has a route to a destination, this node should not answer with a route reply to the sender, since the intermediate node does not know whether further nodes can accomplish the bandwidth criteria.

SourceID	*DestID*	*SrcSeqNum*	*DestSeqNum*	*BroadcastID*	*Hop Count*

Figure 3.3 shows the original RREQ packet format.

SourceID	*DestID*	*SrcSeqNum*	*DestSeqNum*	*BroadcastID*	*Hop Count*	**Req.BW**

Figure 3.4 shows the modified RREQ packet format.

3.3.2 Modified Route Replay Packet

When the RREQ packet arrives at the destination node, the node calculates its available bandwidth and compares it with the required bandwidth if satisfied; the modified RREP packet will be unicasted to the source. But before forwarding this RREP packet a final check procedure will be performed by the destination node to evaluate the maximum route capacity or route bottleneck.

SourceID	DestID	DestSeqNum	Time out	Hop Count

Figure 3.5 Original RREP packet format

SourceID	DestID	DestSeqNum	Time out	Hop Count	**Req.BW2**

Figure 3.6 Modified RREP packet format

RREP packet is only created by the destination node to ensure that all the nodes along the route satisfy the required bandwidth. RREP will be unicasted from the destination to source node only when the actual free bandwidth of the node is larger than total route consumed bandwidth (Req.BW2) for a given flow.

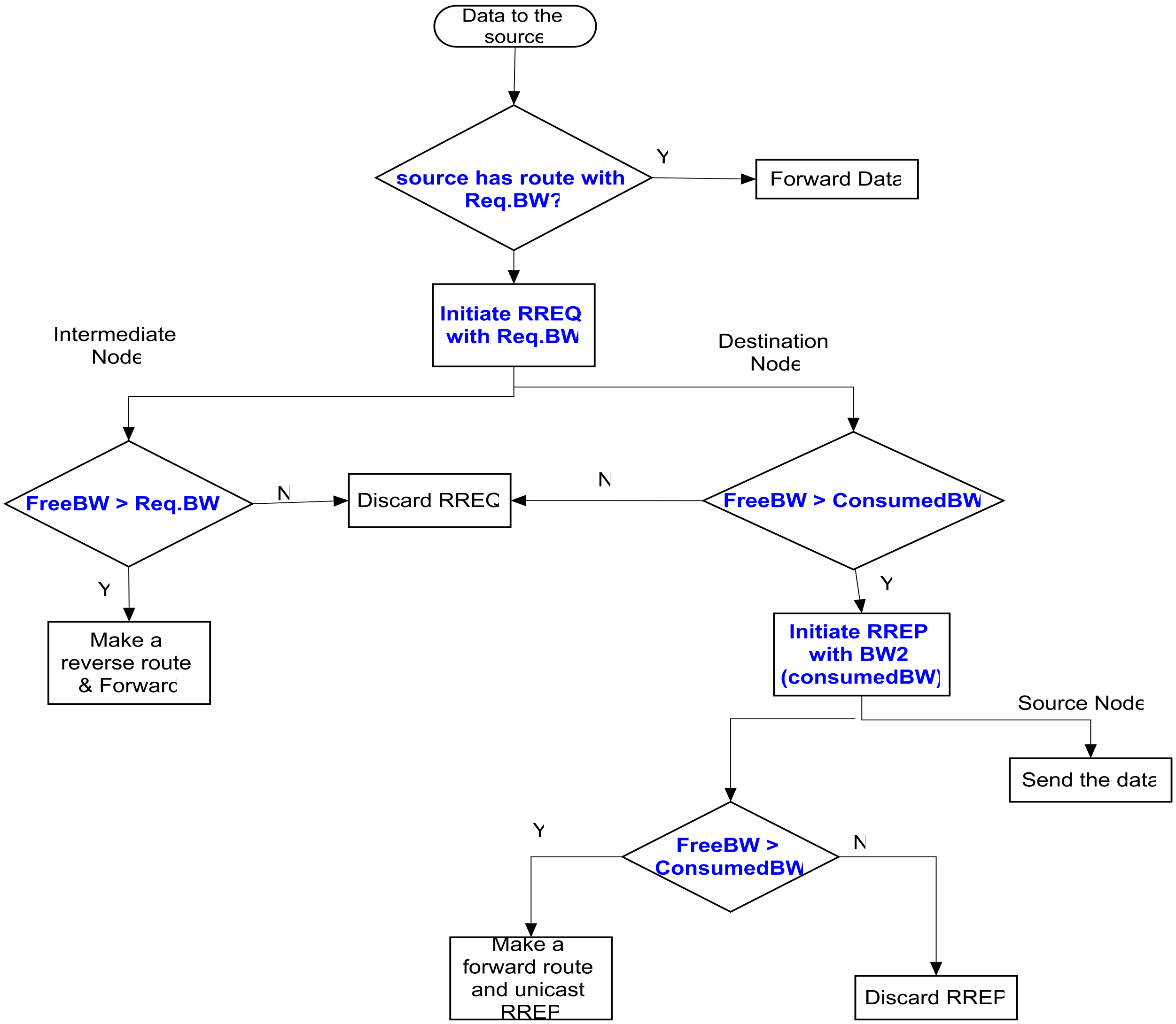

Figure 3.7 shows the RREQ process in the modified AODV (QoS-BAODV) protocol

3.4 Admission Control Mechanism

Obtaining greater bandwidth in the node is not sufficient to say that the current network can offer the required bandwidth for a given flow. The reason is that if the route is chosen, the chosen hosts will bring mutual interference into the network during transmission. To consider node's mutual interference in the route, one final check procedure is required by the destination node before sending the RREP packet back to the source node. The final check procedure is performed by comparing the destination node free bandwidth with the total route consumed bandwidth. The total route consumed bandwidth is the sum of flow bandwidth plus bandwidth due to node's mutual interference (intra flow contention).

Intra flow contention occurs when nodes along a multihop route contend among themselves for channel access to forward packets belonging to the same flow. To estimation the intra flow contention in the route, the Contention Count (CC) at each node must be calculated. The value of CC parameter will help to determine the actual required data rate at each node during an intra-flow transmission as studied in [19].

For good estimation, the carrier sensing range assumed to be more than twice of the transmission range. It means that the nodes which are one hop or two hops away from the transmitter will get the interference and cannot use the channel. Considering one flow which goes through multiple hops, the node has to consider the interference from one hop and two hops upstream and downstream nodes. As an example, assume there is a flow intended from Node 1 to Node 7. To understand the concepts of mutual flow interference across each node in the route finalized values of contention count values are indicated in the table below.

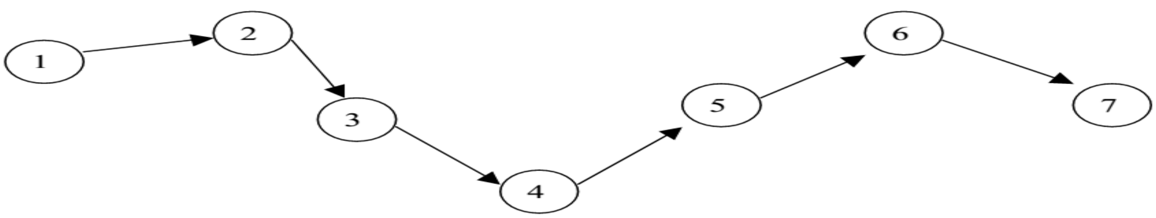

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Figure 3.8 Example of QoS with admission control [19].

	Node 1	Node 2	Node 3	Node 4	Node 5	Node 6	Node 7
Hop 1	Sender	Receiver	**CS**				
Hop 2	CS	Sender	**Receiver**	CS			
Hop 3	**CS**	**CS**	**Sender**	**Receiver**	**CS**		
Hop 4		CS	**CS**	Sender	Receiver	CS	
Hop 5			**CS**	CS	Sender	Receiver	CS
Hop 6				CS	CS	Sender	Receiver
CC	3	4	5	5	4	3	2

Table 3.2 QoS with admission [19]

CS means that the node is in the carrier sensing range of some other node that is transmitting. For example, at the 3rd hop, Node 3 is transmitting packets to Node 4, Node 1, 2 and 4 will get the interference and the channel for Node 1, 2 and 4 should be set as busy since they share channel with Node 3.

From the above example the minimum CC equals to 2, obtained when the destination of a traffic flow is only one hop away from the receiver, and hence the data rate will be the maximum. When the number of hop count is from 1 to 5 the values of CC increases and the data rate will decrease with the number of hop counts increases. When the number of hop counts for a flow is above 4, the data rate will remain the same regardless of the number of contention counts. As stated in the above example, there is a direct relation between CC value and number of hopes in the route. In other word, the route free bandwidth is directly related with number of hopes in the route.

As stated above the RREP packet will be unicasted from destination node to the source node only when the final check procedure is satisfied. In this thesis to check the minimum route capacity (bottleneck bandwidth), we directly related the end-to-end throughput with the number of hops in the route as studied in [13, 19]. The algorism applied for final free bandwidth calculation is as shown below.

```
if(rq->rq_hop_count==1){
        FreeBW=FreeBW;        }
    else{
        if(rq->rq_hop_count==2){
            FreeBW=FreeBW/2;        }
        else{
            if(rq->rq_hop_count==3){
                FreeBW=FreeBW/3;        }
            else{
                if(rq->rq_hop_count==4){
                    FreeBW=FreeBW/4;        }
                else{
                    FreeBW=FreeBW/5;            }
```

After the final check at the destination, the RREP will be unicasted towards the source node only when the free bandwidth of the node is larger or equal to number of hops multiplied by the required bandwidth.

3.5 Controlling Packets Life Span

To compensate the route delay evaluation metrics in such an application, a final extension has been made to AODV packet structure to include a field to measure life span of data packets, and then drop if their age is greater than the maximum accepted delay period, typically 150– 300 milliseconds for real-time traffic [20]. This field will have track the time stamp at which the packet has been created and the local time stamp at the transport layer will mark the time at which the packet has been received.

The difference between the time of creation and time of receipt will give the delay the packet has encountered so far. For delay sensitive applications it is critical that the packets reach the destination before the packets become out of date. So every received packet is checked for the delay with respect to the node it was transmitted and if the value exceeded it is dropped. Therefore, this will reduce the possibility that real-time packets are dropped in the packet queue when the network is congested. Thus, the delay of real-time application data packets can be reduced and the packet delivery ratio can be improved. Hence saving the bandwidth and improving decoding efficiency.

4. METHODOLOGY

4.1. Simulation Tools Used

4.1.1 Network Simulator - 2

Network Simulator-2 (NS-2) is used to simulate the flowing of video traffic in a multi-hop wireless network. It provides support for both wired and wireless networks. For example, NS2 supports protocols like TCP and UDP. It also supports the network layer routing protocols such as AODV, DSDV and DSR for wireless networks.

NS2 is widely used in wireless network research due to its flexibility and modular nature. Its simulation results are comparable among the different methods under the same simulation conditions. This simulator was chosen to be used in this thesis due to its versatility and the possibility to include easily in its code our own applications and modifications of the standard protocols [20].

The core of NS2 is compiled with C++ language, while the object-oriented tool command language (OTcl) is used to implement the simulation script with different parameters and configurations. The simulation is set up by injecting a Tcl simulation script into NS2, the script is read and configured by OTcl, and it is further preceded by C++ functions. The result is generally a text-based trace file which can be used for further analysis. In this work all simulations are performed on NS-2.34.

4.1.2 GNUPLOT

For post processing or to plot result data from the trace files the Gnuplot tool is used. It is a command-line program that can generate two- and three-dimensional plots of functions, data, and data fits. It is frequently used for publication-quality graphics as well as education. The program runs on all major computers and operating systems (Linux, Unix, Microsoft Windows, and others).

Gnuplot supports many types of plots in either 2D or 3D. It can draw using lines, points, boxes, contours, surfaces, and various associated text. It also supports various specialized plot types. Gnuplot supports many different types of output: interactive screen terminals, direct output to modern printers, and output to many file formats [21].

4.1.3 EvalVid - A Video Quality Evaluation Tool-set

EvalVid is a framework and tool-set for evaluation of the quality of video transmitted over a real or simulated communication network. Much research simulates the video traffic using constant bit rate (CBR) data in NS2, but CBR data is generated at a fixed rate, it is different from video traffic which is based on variable bit rates (VBR). The actual video data rate is always fluctuating. CBR data fits uniform packet distribution while actual video data fits exponential distribution. The different characteristics between CBR data and actual video data can cause large deviations in the simulation results. To overcome the problem of CBR data, the EvalVid module is installed to simulate real MPEG 4 video transmission.

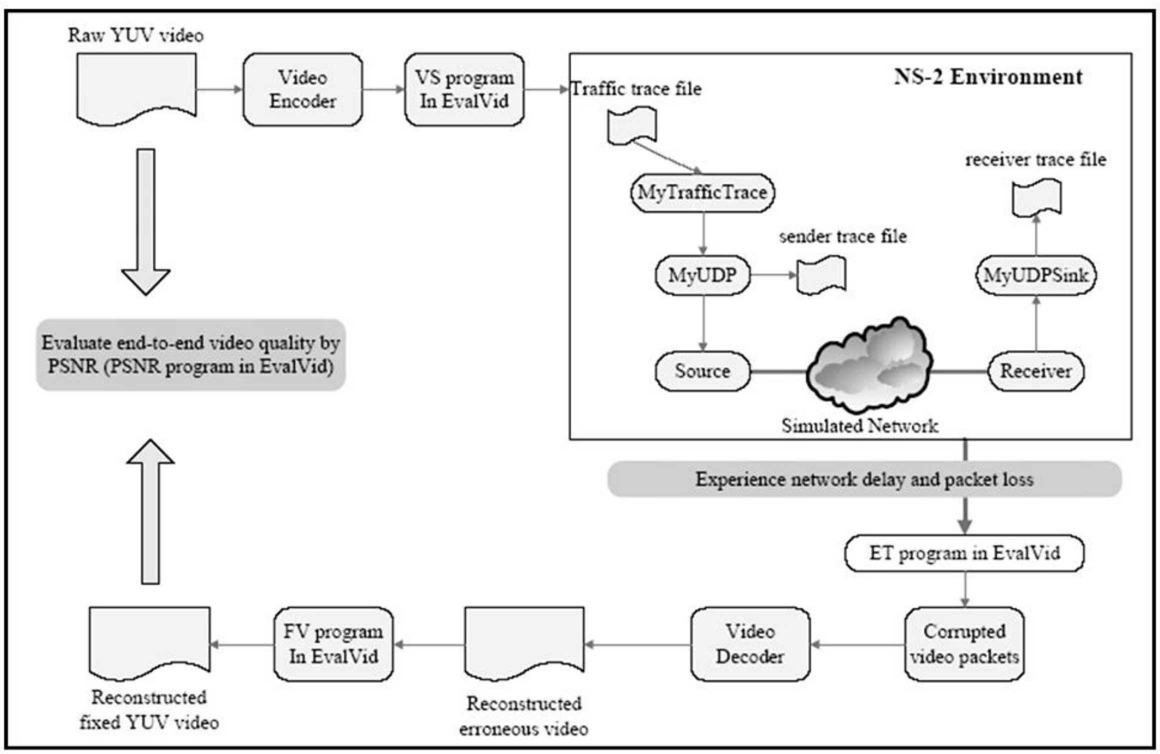

Figure 4.1: Interfaces between EvalVid and NS-2 [22].

The whole process of the EvalVid module is similar to real video streaming transmission over the wireless network. The raw uncompressed video is first compressed in to a reference video to the desired quality.

In our simulation, the raw video file Foreman.yuv is encoded into MPEG-4 format, the compressed video size is 176 x144 and the frame rate is 30 frames per second (fps). A source wireless device is used to send out the video stream and the destination receives the video stream. The source node monitors and records the packet outward time into the trace file. This trace file also records the frame types, frame size, number of packets and sending time [23].

The trace file is included in the simulation script. During the simulation period, the video packets are injected into NS2 source nodes according at the sending time in the trace file. All the video packets are transmitted via a wireless routing path which is discovered by a routing protocol to the destination node. The destination node receives the video packets and records the receiving time into the receiver's trace file. Once all the packets have arrived, they are re-assembled and decoded. The received playable video is then compared with the raw video to obtain the video quality after the transmission.

Three new connecting simulation agents namely MyTrafficTrace, MyUDP, and MyUDPSink, are implemented between NS2 and EvalVid. These interfaces are designed either to read the video trace file or to generate the data required to evaluate the video delivered quality [23, 24].

I. MyTrafficTrace agent: - is employed to extract the frame type and the frame size of the video trace file generated from the output of the VS component of EvalVid. Furthermore, this agent fragments the video frames into smaller segments and sends these segments to the lower UDP layer at the appropriate time according to the user settings specified in the simulation script file.

II. MyUDP: - is an extension of the UDP agent. This new agent allows users to specify the output file name of the sender trace file and it records the timestamp of each transmitted packet, the packet id, and the packet payload size.

III. MyUDPSink: - is the receiving agent for the fragmented video frame packets sent by MyUDP. This agent also records the timestamp, packet ID, and payload size of each received packet in the user specified file. The Integration of EvalVid in ns-2 is shown in the APPENDIX (A) section I.

4.2 Simulation Results and Conclusions

Two experimental evaluation sections are accomplished in this thesis work, the first experimental section is dedicated to select appropriate routing protocol from the existing routing protocol which is suitable for delay sensitive application. From the result obtained AODV protocol was selected due to its performance especially in high traffic and high mobility conditions. The second evaluation section was performed to see the performance gained by the proposed protocol. We evaluated the performance for both sections by measuring the following parameters: data packet delivery ratio, throughput and end-to-end delay of data packet [26].

- Data packet delivery ratio: The data packet delivery ratio is obtained by comparing the number of packets originating at the sources to the number of packets received by the destinations. This is the efficiency of delivering data within the network. This metric is important because it reflects the maximum throughput that the network can support.

- Average End-to-End Delay: - calculates the delay of the packet which is successfully transmitted from the source to the destination. End-to-end delay: This delay not only includes the delay in transmitting data packets through the wireless channel, but also the delay in the network interface queue due to network congestion. End-to-end delay is a measure of routing protocol effectiveness.

- Throughput: - the measure of how fast can we send packets through network or the total amount of data a receiver actually receives from the sender divided by the time it takes for receiver to get the last packet.
- Packets Loss- Some of the packets generated by the source will get dropped in the network due to high mobility of the nodes, congestion of the network etc.

4.2.1 SECTION – I: Comparison of AODV, DSDV & DSR Routing Protocols

To compare and select the better protocol from the existing routing protocols, three basic and mostly used protocols AODV, DSDV and DSR were selected for evaluation. Then exhaustive experimental evaluation was performed on these routing protocols against traffic load conditions and number of nodes.

4.2.1.1 Effect of Traffic Load

In this subdivision, the performance of the protocols was studied against the applied traffic load in the network. The simulation area is 500x500 meters and there are 50 mobile nodes. The maximum moving speed is 20m/sec and the pause time is set to 2seconds. For each connection, 4packet per second data traffic is generated at a constant bit rate, using packets of 512 bytes. The following table summarizes the simulation network parameters.

PARAMETER	**VALUE**
MAC Type	802.11b
Simulation Time	100 sec
Channel Type	Wireless channel
Routing Protocol	AODV, DSDV, DSR
Simulation Area	500 * 500
Traffic Type	CBR
Data Payload	512 bytes/packet
Rate	4 packets/sec / 16.384kbps
Radio Propagation Model	Two Ray Ground
Interface Queue Length	50
Number of nodes	50
Pause Time	2 sec
Mobility Model	Random Way point Mobility
Max speed	20m/s
Number of connections	10, 20, 30, 40

Table: 4.1Simulation Scenario-1

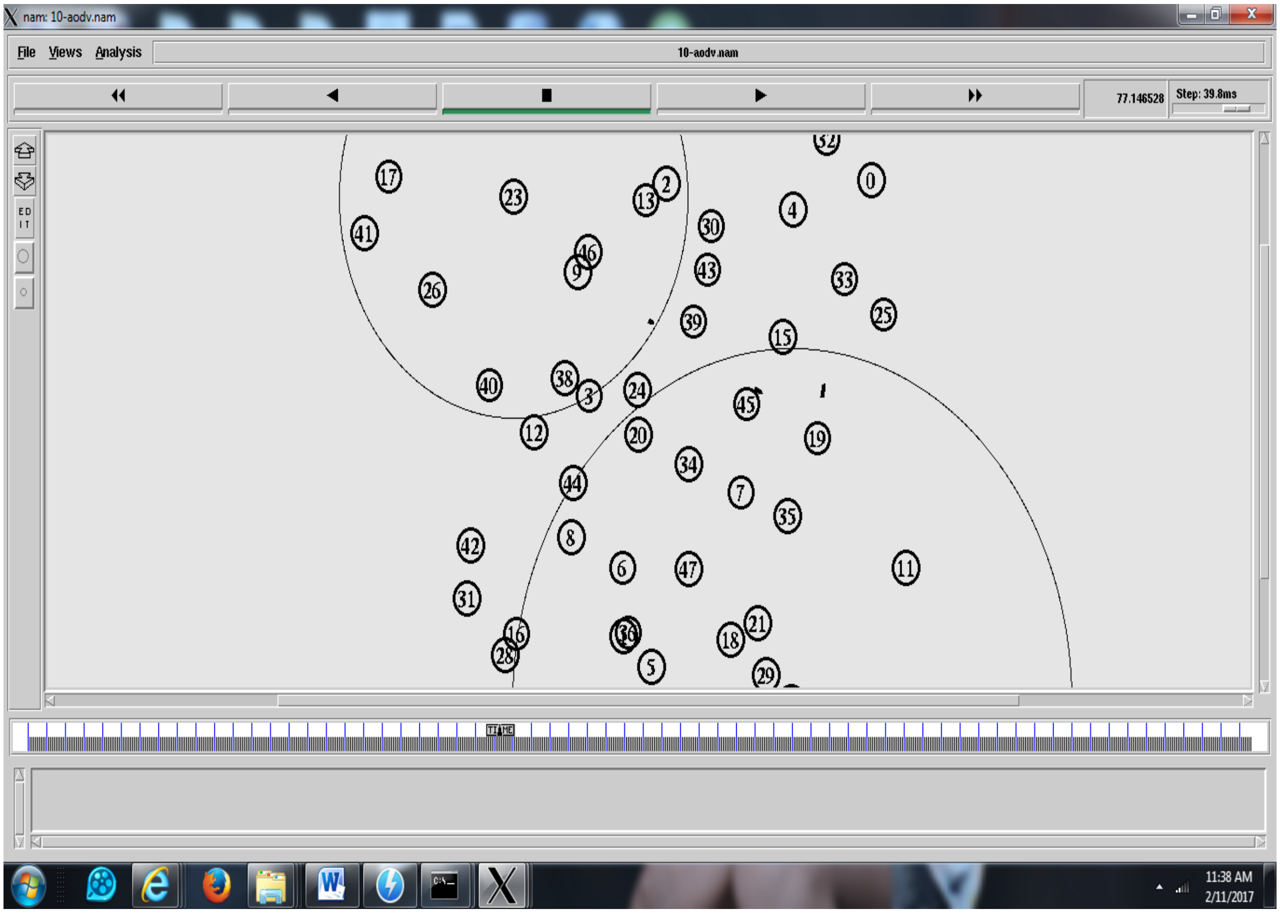

Figure 4.2 Nam snapshot for 50 nodes network

NO. OF CONNC.	THROUGHPUT	DELAY	PDR	PACKET LOSS
10	22.60	0.0169	0.9879	19
20	34.53	0.0237	0.9862	29
30	46.13	0.0578	0.9708	52
40	46.27	0.0932	0.9362	270

Table 4.2 AODV metrics result vs. number of connections.

NO. OF CONNC.	THROUGHPUT	DELAY	PDR	PACKET LOSS
10	16.51	0.0098	0.7185	622
20	25.07	0.0275	0.7174	996
30	30.91	0.0218	0.6545	1129
40	32.96	0.0325	0.6710	1234

Table 4.3 DSDV metrics result vs. number of connections.

NO. OF ONNEC.	THROUGHPUT	DELAY	PDR	PACKET LOSS
10	22.67	0.0228	0.9924	48
20	34.49	0.0161	0.9903	36
30	46.43	0.0649	0.9788	70
40	47.61	0.1416	0.9604	119

Table 4.4 DSR metrics result vs. number of connections

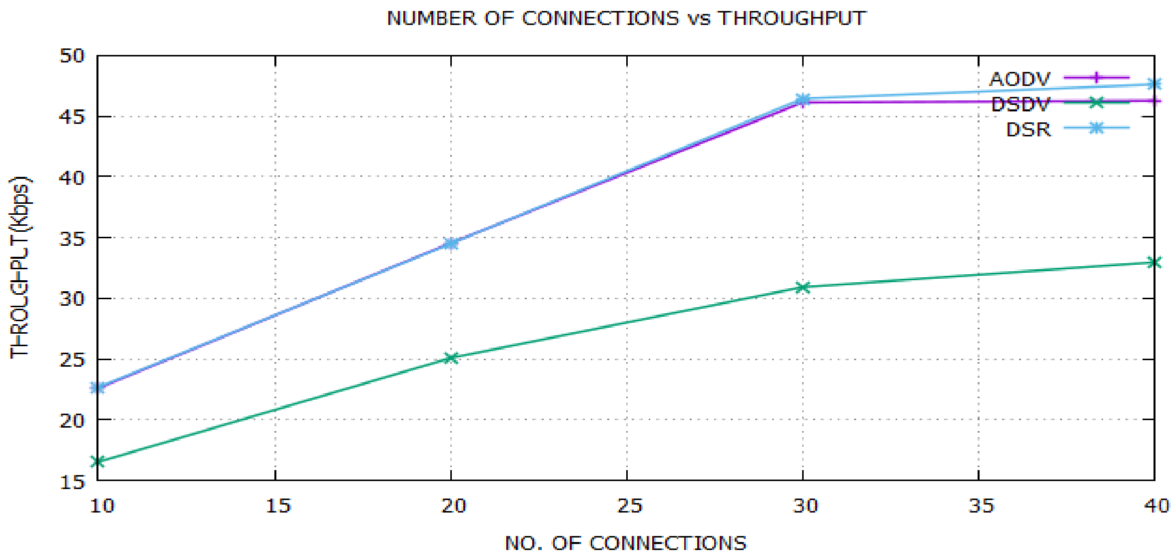

Figure 4.3 (a): Number of Connections vs. Throughput

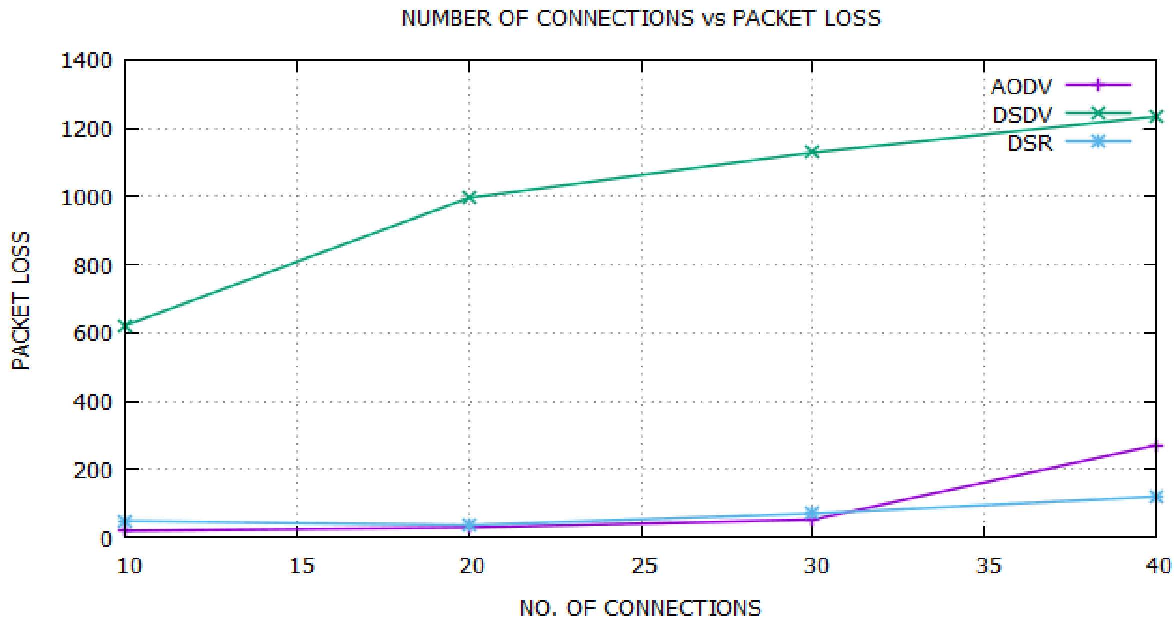

Figure 4.3 (b): Number of Connections vs. Packet Losses

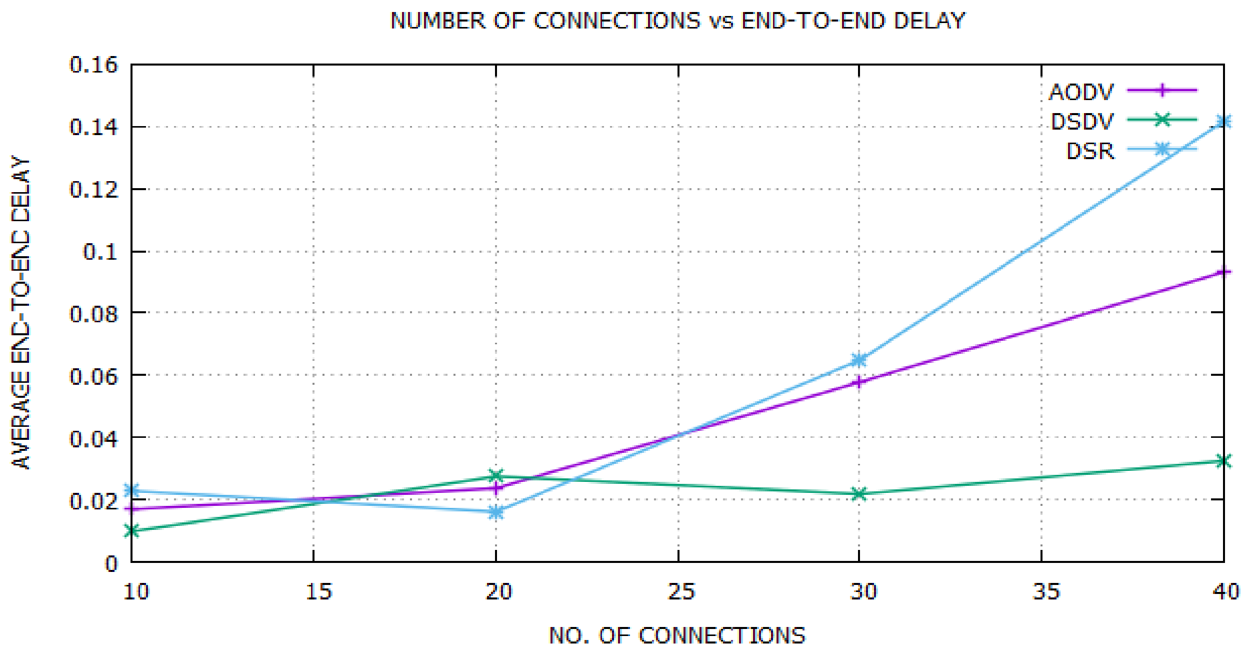

Figure 4.3 (c): Number of Connections vs. delay

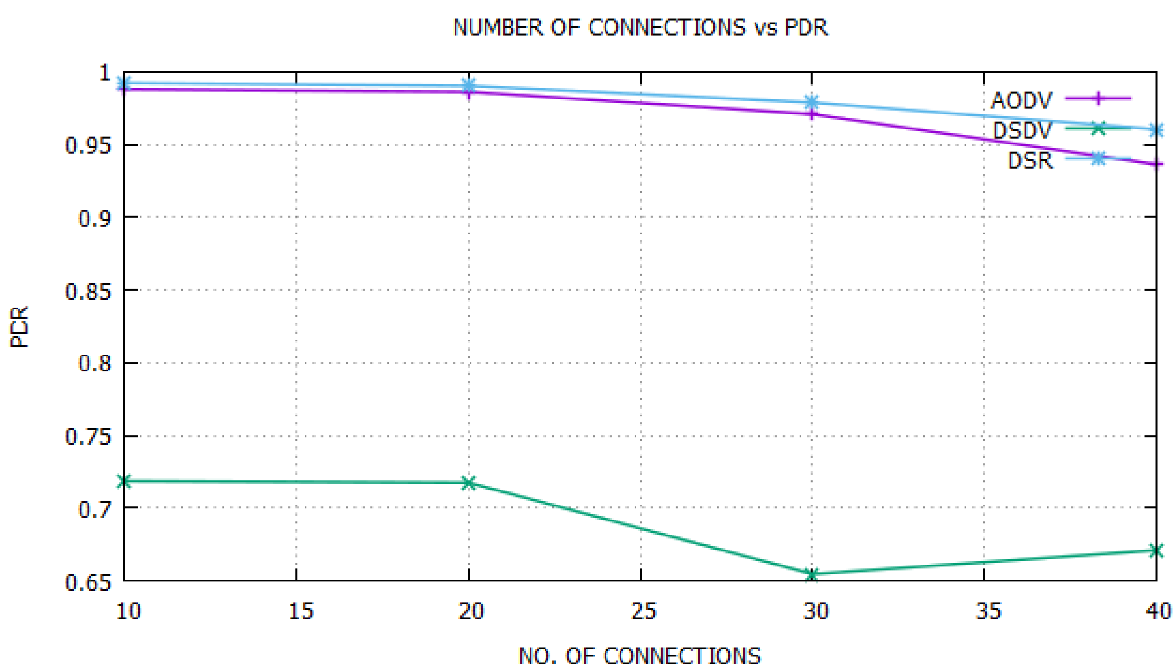

Figure 4.3 (d): Number of Connections vs. PDR

From the above four result figures, the characteristics of the stated three routing protocols with respect to traffic load have been evaluated and the protocol's performance in related to throughput, end to end delay, packet delivery ratio and packet loss are concluded as follow:

- THROUGHPUT - figure 4.3(a) shows the two on-demand routing protocols AODV and DSR perform well almost equivalently whereas the performance of DSDV is very low. Here the characteristic of reactive protocols, staying packets in their mac when the route gets lost favors packets to be delivered if the route gets recovered or if other route is discovered soon.

- PACKET DROP (From figure 4.3(b)) – Generally as the traffic load increases the bandwidth requirement increases and then packets dropped will also increase since the network capacity is limited. When we compare among the tested routing protocols DSR and AODV drops fewer packets compared to DSDV because, each packet that the MAC layer is unable to deliver is dropped in DSDV.

- END TO END DELAY: - Figure 4.3(c) shows DSDV has the lowest end-to-end delay, than the other protocols because DSDV always holds optimal paths to all other destinations in their routing tables and hence it can send data packets immediately. However high end-to-end delay is reasonable in DSR and AODV because they deliver more packets at the destination at the cost of delay. This metric is very essential when transmitting multimedia data as it affects the quality of the streaming video. For real-time multimedia services, the accepted threshold of delay can be considered to be approximately 150 -300 milliseconds [20]. In the result figure we observe that DSR has higher end-to-end delay after 30 connections and this shows that it is note stable like AODV when the traffic load get increased further.

- PDR: - from figure 4.3(d) we can observe that the packet delivery ratio decreases when increasing the transmission sessions. AODV and DSR present almost identical performance while DSDV has the lowest performance. In the case of multimedia transmission, the DSDV does not seem to be suitable, as the packet delivery ratio is very low even when having small number of connections. However, the reactive protocols present an acceptable ratio for up to 40 connections in our network topology parameters.

4.2.1.2-Effect of Node Number

In this subsection the performance of the three routing protocols in related to number of nodes in the network has been evaluated. Our objective is to investigate the impact of node density on the protocol's performance. We use the same simulation area as in our previous simulations and gradually increase the number of nodes in the network. Table 4.3 below shows the simulation parameters.

Parameter	**VALUE**
MAC Type	802.11
Simulation Time	100
Channel Type	Wireless channel
Routing Protocol	AODV, DSDV, DSR
Simulation Area	500 * 500
Traffic Type	CBR
Data Payload	512 bytes/packet
Network Loads	4 packets/sec
Radio Propagation Model	Two Ray Ground
Interface Queue Length	50
Number of nodes	10,20,30,40,50,80,100
Pause Time	2 sec
Mobility Model	Random Way point Mobility
Max speed	20m/s
Max connection	10

Table 4.3: Simulation Scenario-2

NO. OFNODE	**THROUGHPUT**	**DELAY**	**PDR**
10	45.88	0.0137	0.8542
20	48.89	0.0155	0.9858
30	49.15	0.0193	0.9880
40	48.93	0.0166	0.9960
50	48.40	0.0842	0.9897
80	71.65	0.408	0.5797
100	58.58	0.597	0.4732

Table 4.4 AODV metrics result vs. number of nodes.

NO. OF NODE	THROUGHPUT	DELAY	PDR
10	40.98	0.0088	0.5779
20	38.68	0.0222	0.6877
30	40.37	0.0280	0.7216
40	38.64	0.0094	0.8398
50	35.67	0.0106	0.858
80	34.01	0.0847	0.6618
100	32.14	0.986	0. 5769

Table 4.5 DSDV metrics result vs. number of nodes.

NO. OF NODE	THROUGHPUT	DELAY	PDR
10	45.75	0.0149	0.8522
20	49.46	0.0261	0.9796
30	49.04	0.0192	0.9910
40	48.96	0.0099	0.9969
50	48.83	0.0137	0.9991
80	42.22	0.5577	0.6295
100	38.45	0.7506	0.5575

Table 4.6 DSR metrics result vs. number of nodes

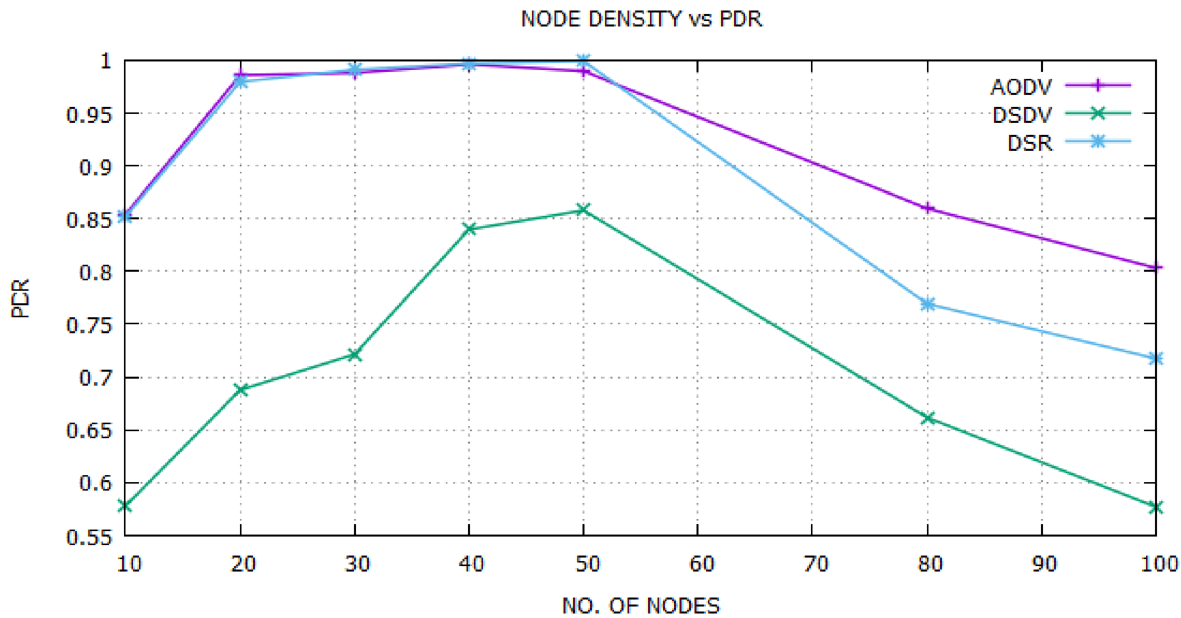

Figure 4.4(a): Number of Nodes vs. PDR

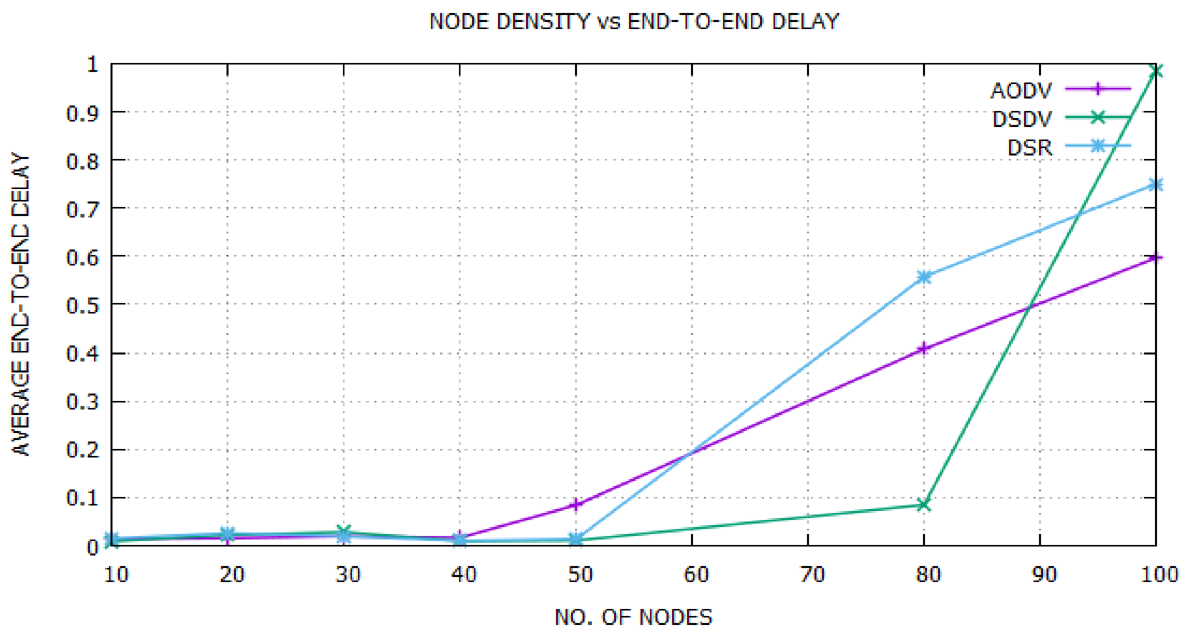

Figure 4.4(b): Number of Nodes vs. delay

The above result figures show the performance strength of AODV, DSDV and DSR routing protocols against the number of nodes in the network. As a general principle, a small number of nodes in a large simulation area will result in low connectivity due to the large distances between nodes. In contrast, a large number of nodes in a small simulation area will result in signal interference, as nodes are located very close to each other.

From the two result figures in terms of the PDR metrics AODV and DSR protocols performs equivalently good, whereas DSDV performs lower than the two other protocols. For the delay metrics the average end to end delay in DSDV is lower than AODV and DSR the reason would be, once a congestion or route failure occurred the entire packet in the middle of the route will be dropped since DSDV is incapable of local route repair and it has no mechanism to stay packets for a while.

When we compare the two reactive protocols in terms of average end to end delay they perform identically up to 50 nodes in the network but for nodes greater than 50 end to end delay of AODV is lower than DSR value and this implies that for dense networks AODV outperform DSR protocol and also the average end to end delay of DSDV increases significantly when the nodes greater than 80.

➢ Section One Conclusion

From the above evaluation the following conclusions have been made on the performance of on demand routing protocols for different QoS metrics.

- By comparing these protocols on many performance metrics, we have reached to a conclusion that reactive protocols are better than proactive protocols.
- Specifically, for application such as video transmission in which the priority is to have minimum delay in a network with larger connection and greater mobility then AODV protocol is the best choice.
- The choice of protocol that should be used for MANET is totally dependent on the type of application required.

4.3 SECTION – II: Comparison between AODV and QOS-BAODV.

From the results and conclusion obtained from section one, AODV routing protocol was selected due to its best performance for delay sensitive application. Despite the stated advantage this protocol didn't provide the required QoS requirement for our application and modification have been performed as described in chapter three. In this section extensive simulation was performed to evaluate the performance enhancement obtained by the proposed protocol which uses node's available bandwidth for route discovery over the original AODV routing algorithm.

4.3.1 Varying Traffic Loads

In this part experimental evaluation was performed by varying the transmitted data rate from 100kbps up to 2Mbps over a network with 50 numbers of nodes. As a background traffic10 CBR flow with 20kbps randomly generated over the network. The channel capacity is assumed as 11Mbps. In this evaluation both for the main and background traffic we used constant bit rate (CBR) data generated by the network.

Parameter	VALUE
MAC Type	802.11
Simulation Time	200
Channel Type	Wireless channel
Routing Protocol	AODV, QoS-BAODV
Simulation Area	1000 * 1000
Traffic Type	CBR
Data Payload	512 bytes/packet
Network Loads Data rate (in Mbps)	0.1, 0.2, 0.4, 0.6, 0.8, 1.0, 1.2, 1.6, 1.8 and 2.0
Background traffic	5packet/sec (20kbps)
Radio Propagation Model	Two Ray Ground
Interface Queue Length	50
Pause Time	2 sec
Mobility Model	Random Way point Mobility
Channel capacity (Data rate)	11M

Table 4.7: Simulation Scenario-4

Data rate (Mbps)	THROUGHPUT (kbps)	DELAY	PDR	PACKET LOSS
0.1	85.38	0.0119	0.9980	12
0.2	190.39	0.0517	0.9830	82
0.4	389.82	0.0637	0.9700	221
0.6	570.99	0.0785	0.9290	392
0.8	749.67	0.1471	0.7263	1221
1.0	872.67	0.2173	0.6447	5009
1.2	981.35	0.2945	0.6290	6837
1.6	1087.62	0.3246	0.5460	10416
1.8	119.54	0.3572	0.5112	13149
2.0	1237.7	0.3811	0.4230	15071

Table 4.8 AODV metrics result vs. data rate.

Data rate(Mbps)	THROUGHPUT(kbps)	DELAY	PDR	PACKET LOSS
0.1	96.21	0.0724	0.9920	11
0.2	195.55	0.0810	0.9850	79
0.4	238.01	0.0670	0.9820	187
0.6	578.88	0.0947	0.9624	204
0.8	792.45	0.0812	0.9488	511
1.0	995.41	0.0711	0.9044	805
1.2	1183.44	0.0844	0.8520	978
1.6	1572.94	0.1270	0.815	1024
1.8	1705.24	0.2584	0.7410	1141
2.0	1897.54	0.2947	0.7140	1415

Table 4.9 QoS-BAODV metrics result vs. data rate.

Figure 4.5 Average delay comparisons

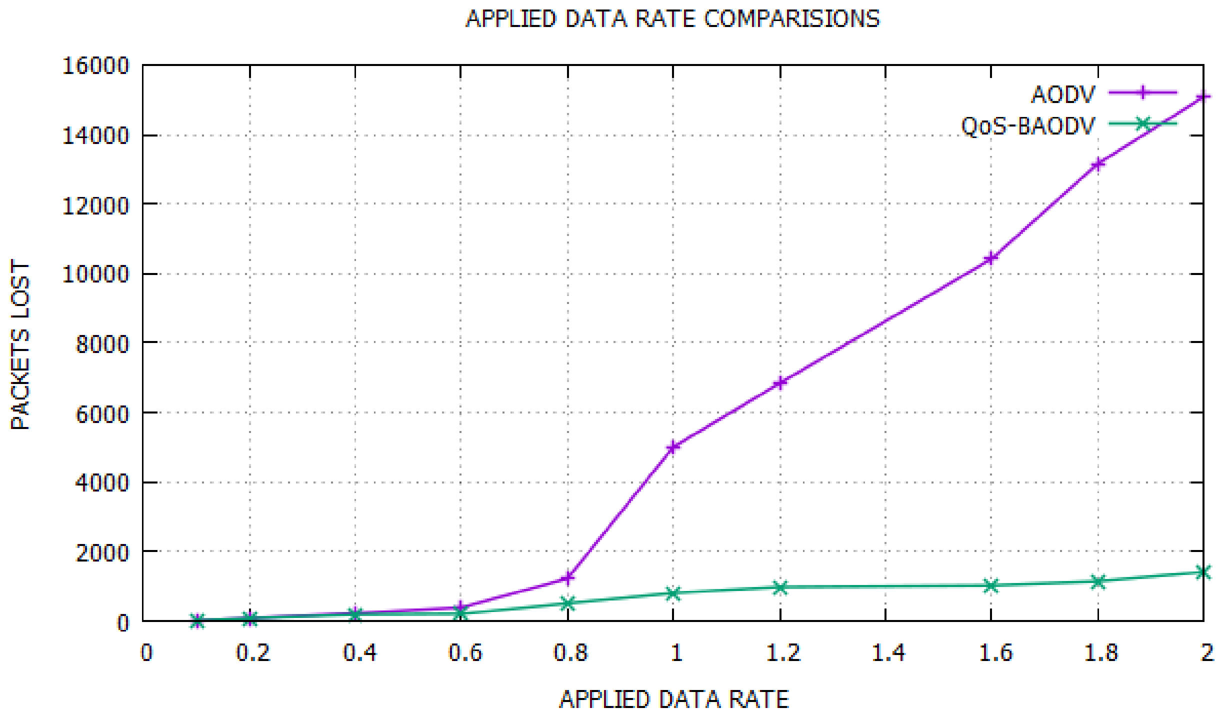

Figure 4.6 Packet loss comparisons

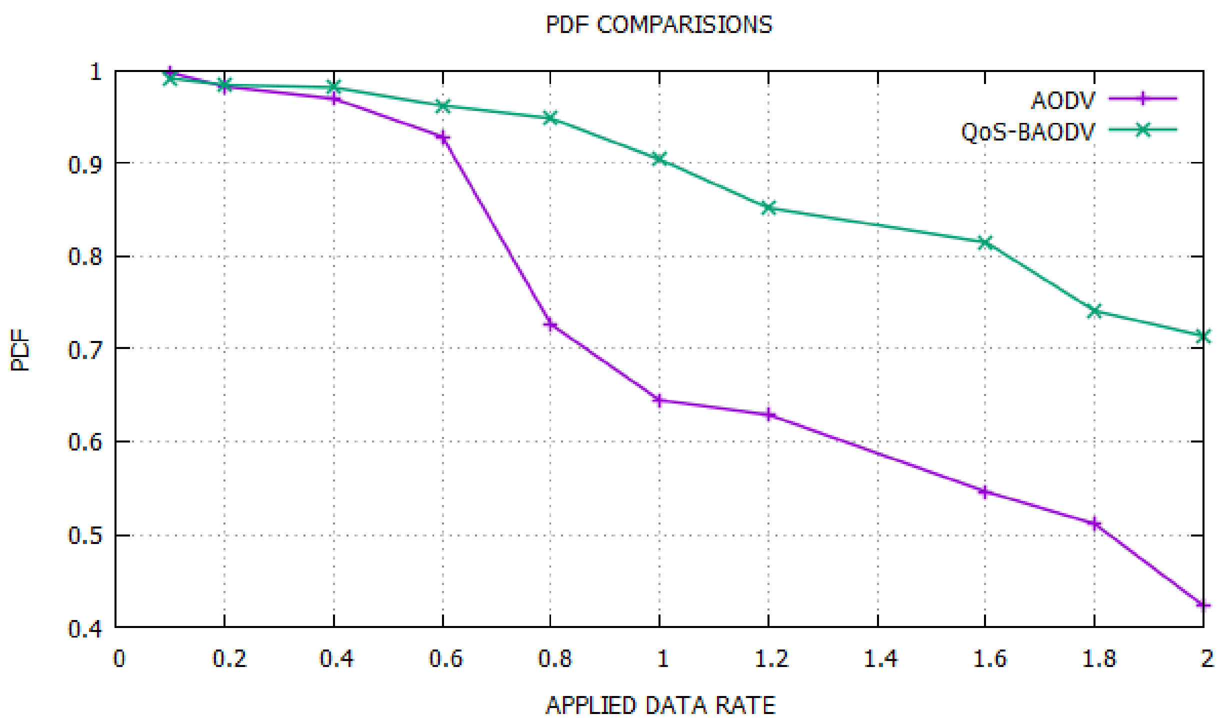

Figure 4.7 Packet Delivery Ratio Comparisons

From the above simulation results, we observed that traffic load has a significant impact on routing protocol performance. For average end to end delay and PDR performance metrics comparison, comparison analysis has been provided by the following two paragraphs.

From figure 4.5 we can see that the average end to end delay encountered by QoS-BAODV is lower than AODV protocol especially when the data rate is higher. The reason is for every data flow the proposed protocol examines the route capacity before injecting the data to the network; hence reduce the packet delay in the route. In some part when the traffic is low in the network (below 0.4Mbps) delay is lower in AODV routing protocol the reason would be, no matter which route the traffic flow chose, the route chosen can provide enough data rate at most of the time.

From figures 4.6 and 4.7 we can realize that both protocols have identical performance when the data rate is below 0.4Mbps, the reason would be when the traffic is light or application bandwidth requirements are low, sufficient bandwidth can be guaranteed for applications in the network to provide a high packet delivery ratio. When the data rate gets higher the network becomes congested, and the buffers becomes full because packets will queue longer till they get the channel access and additional packets of the next flow will be lost completely in case of AODV. Whereas in QoS-BAODV the routes can guarantee the bandwidth requirements of each flow, so the traffic load can be transmitted safely and the possibility of packet loss is less.

From the above evaluation results we have concluded that for higher data flow, the proposed protocol performs very well in all evaluation metrics. But for multimedia application like video transmission this enhancement might not be enough, because such an application needs timing arrival of packets to the destination. To guarantee the timing arrival of multimedia packets, in our network architecture the MAC layer is modified to differentiate the data packets types in the network. This MAC layer enhancement provide the prior channel access for most important packets as described in chapter two.

4.4 Separation of QoS Traffic from Best Effort Traffic Using IEEE 802.11e

In this section a set of experiments was performed to evidence how the IEEE 802.11e technology is able to differentiate QoS traffic from best effort traffic. The evaluation we performed on a network topology that consists of 20 static nodes, containing four source node and four destination nodes to transmit and receive data with different priority. All sources generate 650 kb CBR traffic with different starting time and the channel capacity is assigned to 2Mbps.All nodes have the default configuration for differentiation parameters.

Parameter	**VALUE**
MAC Type	802.11, 802.11e
Simulation Time	100
Channel Type	Wireless channel
Routing Protocol	AODV
Simulation Area	500 * 500
Traffic Type	CBR
Data Payload	512 bytes/packet
Network Loads	650kb
Radio Propagation Model	Two Ray Ground
Interface Queue Length	50
Data Rate	2Mb
Number of nodes	20 static nodes
No of data flow	4 data with different priority

Table 4.10: simulation scenario

The simulation result provided by figure 4.8 below is the practical confirmation for the theory described in chapter two; it shows how DCF provides the best-effort service. Up to 20 seconds station-1 and station-2 transmit their data with the required throughput. When station-3 joins the network to transmit its data the traffic flows become larger than the channel capacity and the throughput decreases equally for all stations.

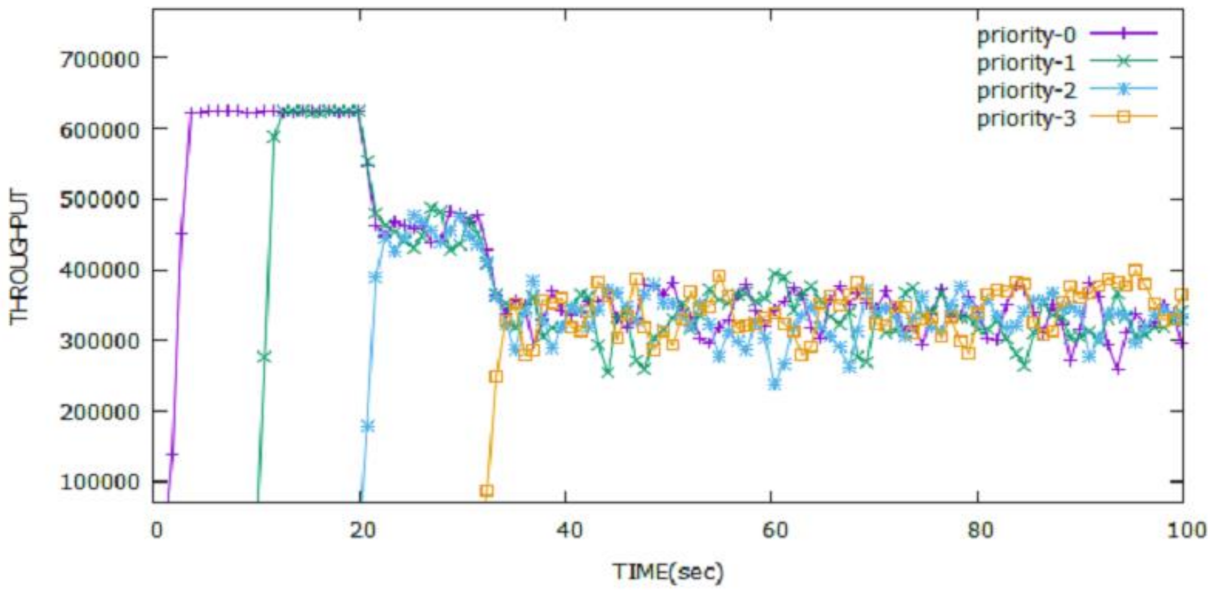

Figure 4.8 Throughput by original network model IEEE 802.11

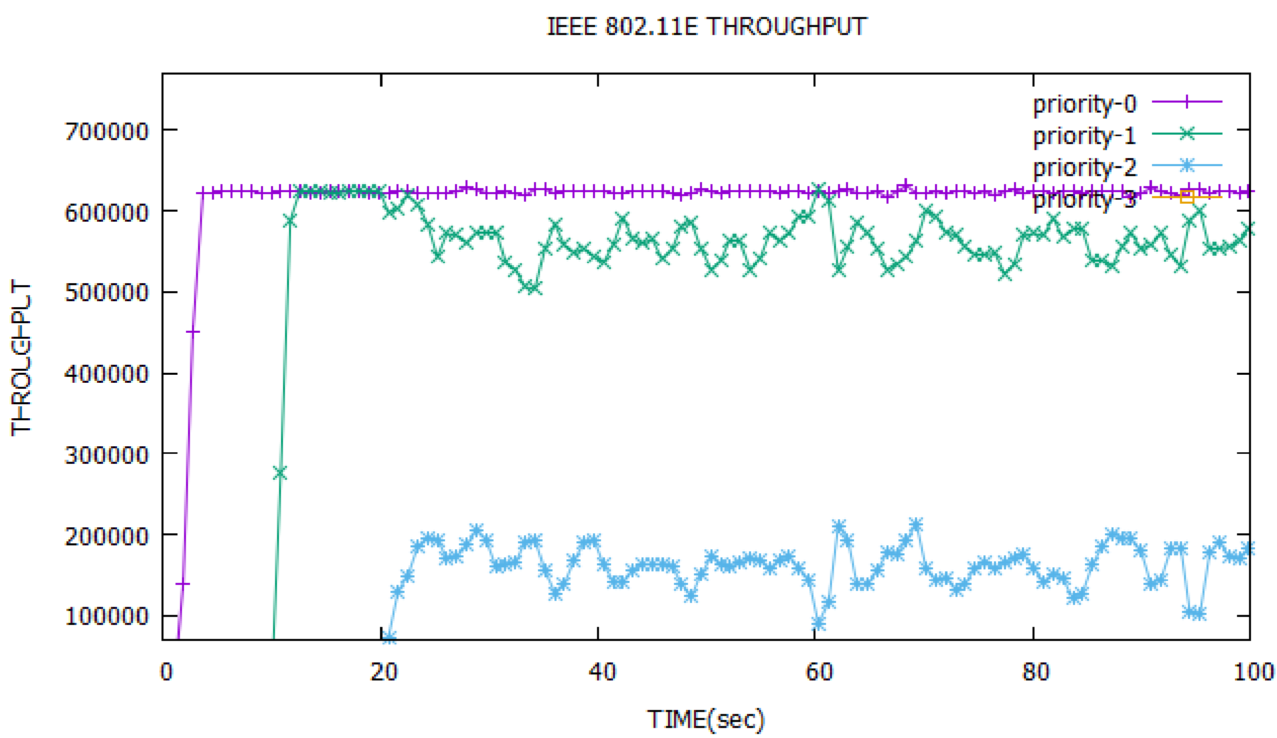

Figure 4.9 Throughputs by IEEE 802.11e

To evaluate performance enhancement achieved by IEEE 802.11e due to its QoS support or data differentiation, the same network topology with different DIFS parameters are applied according to the following table.

Sources	DIFS	CWmin: CWmax
1	20µs	31:1023
2	40µs	31:1023
3	60µs	31:1023
4	80µs	31:1023

Table 4.11 DIFS parameter for DIFS-based Scheme

The above figure shows that even if congestion occurs on the network the required channel bandwidth is provided for high priority data's and the remained bandwidth will be given to the other data flows according to their priorities. In IEEE 802.11e queue structure high priority packets are always put in front, when congestion occur non-real time packets which are located in the tail of the queue are dropped. This clearly puts into evidence that MAC-level QoS support is essential to achieve a global QoS framework in MANETs.

4.5 Final Protocol Evaluation

In the final evaluation test the protocol was evaluated by applying real video data sequence of foreman which contain 400 frames (I=45, P=89 & B=266) and CBR traffic load with 100kb data. The video application starts transmitting after 10 seconds whereas the CBR starts immediately. For preprocessing and post evaluation of video data, EvalVid is integrated with our NS-2 to decode, encode and evaluate the received video data.

In our protocol for the MAC layer enhancement we implemented a method proposed in [16]. In this method three AC assigned for each video frames and one AC for other data. By this mapping algorithm to transmit MPEG-4 video streams as the traffic, since I frame is very important part of GOP it always mapped to AC [0], while the P frame to AC [1], the B frame mapped to AC [2] and other data will be mapped to AC [3] and here prior channel access is given for AC which contains video frames according to their priority.

Here we integrated controlling mechanism to evaluate the video data packets life span not to be larger than 200ms, since a data which arrives longer than this would be useless or it will not be decoded to construct received video as described in chapter three 3.5 section. By doing this the channel bandwidth will increase by dropping the outdated packets from transmitting to the destination.

In this sub-section real video data was transmitted over different network architectures and the overall performance of the proposed system is evaluated. To see the explicit improvement we used the following network architectures.

1. AODV routing with IEEE 802.11 MAC.
2. AODV routing and IEEE 802.11E MAC.
3. QoS-BAODV routing protocol.
4. QoS-BAODV + TTL, for packet loss and PSNR comparison.

Parameter	**Value**
MAC Type	802.11 MAC, 802.11e, proposed
Simulation Time	200
Channel Type	Wireless channel
Routing Protocol	AODV, QoS-BAODV
Simulation Area	500 * 500
Traffic Type	Video + CBR
Radio Propagation Model	Two Ray Ground
Interface Queue Length	50
Data Rate	2Mb
Number of nodes	50 nodes (1video source and 1 receiver)
video source	Foreman , YUV QCIF (176 x 144) (400 frames: I=45, P=89 & B=266)
CBR load	100kbps
Video data rate	260kbps and 520kbps (1 and 2 video flow)

Table 4.12: Simulation Scenario-5

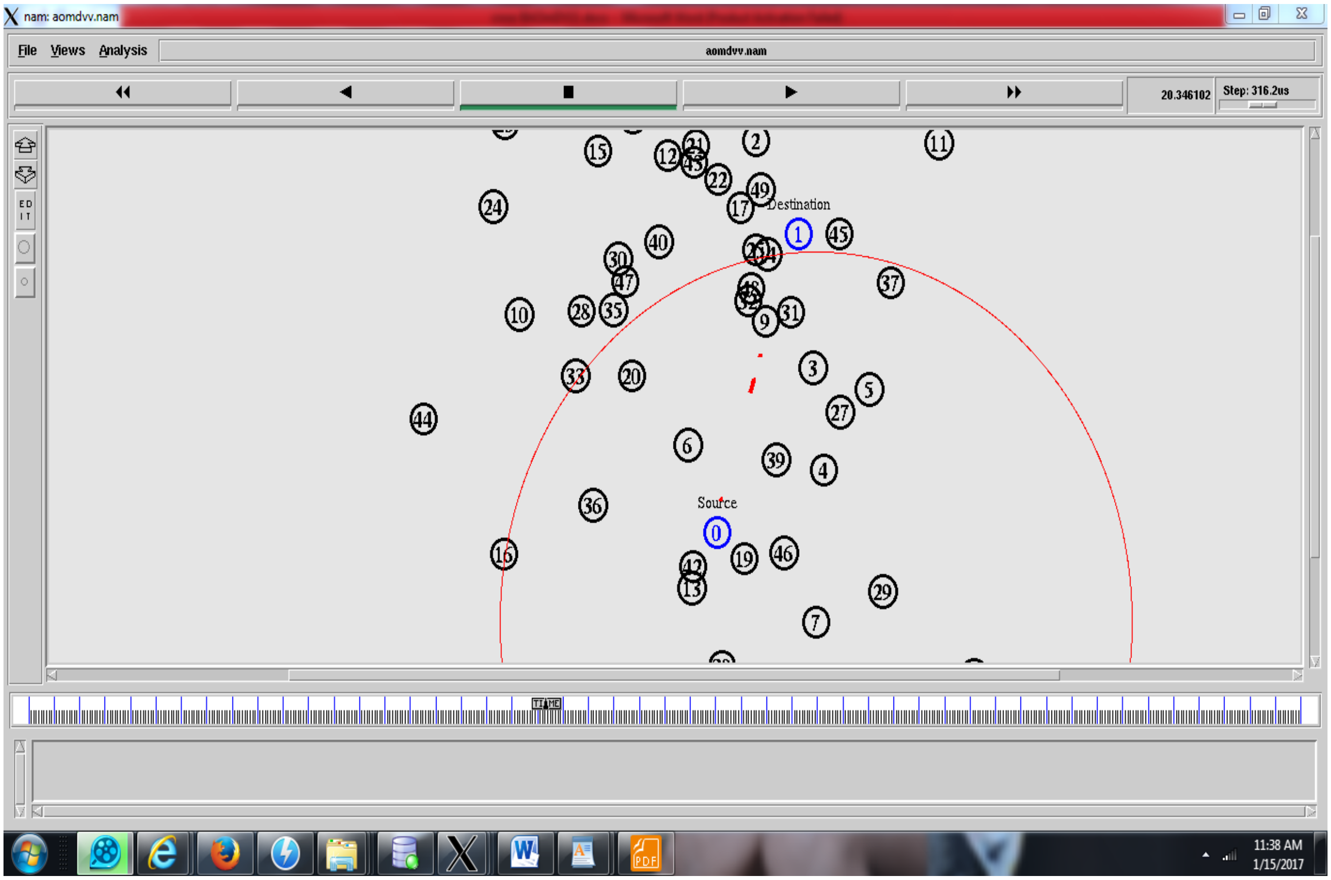

Figure 4.10: 50 nodes with one source-destination network snapshot

To evaluate the above network architectures, we used randomly CBR traffic and encoded video sequence of QCIF format of Foreman (176 * 144). To test the effect of different data rate on the protocol, we applied single and multiple video flows. To see the performance enhancement of our protocol with the above mentioned two other network architectures, a real video data and CBR traffic load are applied to be transmitted on a network with 50 nodes density. The evaluation metrics are number of packet loss, PSNR, visual comparison test on the received video and on two received frame pictures (10^{th} and 200^{th}). Raw video data simulation steps (encoding, transmission and decoding) are described in appendix-B.

4.5.1 Evaluation Scenario I- Single Video Flow With CBR Traffic.

I. AODV routing with IEEE 802.11(FIFO-no packet differentiation)

Figure 4.13 shows the quality of video frame received by this architecture is not good. The reason is FIFO queue mechanism entertains all packets that flow through the network equally, so lately received video packets forced to wait longer until all prior packets in the queue are forwarded. The total data sending rate is 360kbps (100kbps+260kbps) which is affordable by the network.

Figure 4.11 Original frames (10^{th} and 200^{th})

Figure 4.12 AODV-IEEE 802.11received frames (10^{th} and 200^{th})

II. AODV routing with IEEE 802.11E (priority queuing)

In this evaluation test a single video stream or flow and CBR load applied over the original AODV protocol with 802.11E MAC. The quality of received video frame is very good; the reason is even if the queue is full with other previous data whenever a video packet arrives channel access priority will be given irrespective of its arrival time.

Figure 4.13 AODV-IEEE 802.11E received frames (10th and 200th)

4.5.2 Evaluation Scenario II - Two Video Flows and CBR Traffic.

I. AODV routing with IEEE 802.11E

Figure 4.14 AODV-IEEE 802.11E with two simultaneous video flows and CBR load

II. Proposed QoS-BAODV routing protocol.

Figure 4.15 QoS-BAODV-IEEE 802.11E with two simultaneous video flows and CBR load

As shown in Figure 4.14 above, the received video frame quality obtained by this architecture becomes very poor when the video rate is doubled. This shows that applying packet priority mechanism at MAC layer will not provide good result in all cases or performance gets reduced when the sending data rate exceeds the nodes available capacity. In other word, uncontrolled packet injection will result in congestion over the network and some video packets forced to be dropped. This problem can be solved by selecting a capable route (with enough bandwidth) to forward the applied data rate. The total data sending rate is greater than 620kbps (100kbps + 520kbps).

From figure 4.15 we observed that the received video frame quality is improved significantly, the reason is before forwarding the video data to the network in QoS-BAODV protocol an appropriate route that satisfy the application requirement is selected and hence the probability of packet drop will be reduced greatly.

	Average PSNR(dB)	FRAME LOSS NUMBER			Total Lost
		I FRAME	P FRAME	B FRAME	
AODV +802.11	24.55	5	32	45	82
AODV +802.11e	32.28	0	1	2	3
QoS-BAODV	33.11	0	2	6	8
QoS-BAODV + TTL	34.9	0	5	12	16

Table 4.13: The average PSNR and number of frames lost on Single video flow

	Average PSNR(dB)	FRAME LOSS NUMBER			Total Lost
		I FRAME	P FRAME	B FRAME	
AODV +802.11	20.01	12	55	102	169
AODV +802.11e	31.14	8	2	33	43
QoS-BAODV	31.45	0	6	12	18
QoS-BAODV + TTL	33.01	0	16	38	54

Table 4.14: The average PSNR and number of frames lost on multiple video flows

In QoS-BAODV + TTL, data packet's time-to-live value is fixed at 200 ms or 0.2 seconds. The above two tables contains the result of average PSNR and frame loss obtained by the four network architecture when we applied a single and multiple video flows with one CBR traffic along the given network. From the result we observed that the PSNR obtained by the proposed protocol is higher always.

When we see number of frame loss, our protocol always deliver the I frame safely and hence the decoded video quality at the receiving end found to be good. When we compare frame loss of P and B frames between the third and the fourth architectures loss is higher in the last architecture, because when the data flow increases the network is going to be congested and channel priority is given for I frame packets, so the P and B frame packets are forced to stay longer in the queue. Since our architecture is enabled with the feature to delete useless or aged packets, the loss probability of P and B frame packet increased in our architecture. This is due to the fact that the stale packets that exceed the time to live values are dropped.

5. CONCLUSION AND RECOMENDATION

5.1 Conclusion

In this thesis work extensive evaluation test was performed on three routing protocols (AODV, DSDV and DSR) to select a suitable protocol for our delay sensitive multimedia application.

From section - I test evaluation results AODV was selected as a framework protocol due to its stability and capable of performing in high data rate traffic network conditions. Even though AODV is better than other existing protocols in our evaluation metrics, it does not have QoS support to transmit delay sensitive applications like video data over MANETs.

To meet QoS required by the application, major modifications was performed on network and MAC layers of the ordinary AODV protocol. Mainly on the network layer, route selection was changed from number of hops metrics to node's available bandwidth. For estimating node's available bandwidth the MAC layer was modified to sense or listen the channel activities within specified time period.

From the experimental results obtained by the proposed bandwidth aware QoS routing protocol (QoS-BAODV), we can undoubtedly concluded that our protocol is an efficient especially when the network is provided with large data rate and nodes are mobile, since it can better estimate the residual bandwidth in case of frequent route breaks. And also in the proposed architecture channel access priority is given for real time data to deliver to the destination with in the time bound. If some video packets stayed longer than the acceptable delay these data will be dropped to maximize the network resource or bandwidth. These characteristics make the protocol more suitable for real time data.

5.2 Recommendation and Future Work

Even though this thesis provides a broad study and evaluation from different views, there are still some open issues and several research directions that can be tracked to improve the performance of our proposed protocol. The following are some points that can be taken into consideration when deciding to further the work done in this thesis:

As a recommendation for future work this work can be extended to make the system adaptive to the available resources. In other word, enable the application layer to feed a data according to the feedback provided by the lower layers. End to end delay metric can be suggested as an additional metric during the route discovery and maintenance in the routing protocol. In regard to packet priority for channel access in MAC layer, rather than assigning static AC for different data types we further recommended to apply adaptive mapping mechanism, to assign different packet types to different AC depending on network congestion and packet priority level.

REFERENCES:

1. Changling Liu "A Survey of Mobile Ad Hoc network Routing Protocols", published in University of Ulm Tech.Report Series, Nr. 2013-08.
2. Shan Gong 'Quality of Service Aware Routing Protocols for Mobile Ad Hoc Networks'
3. Simulation and Evaluation of Routing Protocols for Mobile Ad Hoc Networks by Georgios Kioumourtzis, MSC thesis, September 2010
4. P.Sakalley, J.Kumar "Review and Analysis of Various Mobile Ad Hoc Network Routing Protocols", ISSN, 2013.
5. S. Corson, J. Macker. Mobile Ad hoc Networking (MANET): Routing Protocol Performance Issues and Evaluation Considerations, January 2009.
6. E. Crawley, and H. Sandick. A Framework for QoS-based Routing in the Internet, 1998.
7. S. Tabatabaei and K. Tabatabaei "Routing and quality of service support for mobile Ad hoc networks"in proceeding of International conference on Computer Engineering and Technology, Chengdu, china, April 2010.
8. C.H.Ke, "How to evaluate MPEG video transmission using the NS2 simulator?", EE Department, NCKU, Taiwan University, Taiwan, 2006.
9. QoE Enhancements in IEEE 802.11e EDCA for Video Transmission through Selective Queueing, Proceedings of the Eleventh International Network Conference (INC 2016)
10. L. Romdhani, and T. Turletti, "Adaptive EDCF: Enhanced service differentiation for IEEE 802.11 wireless ad-hoc networks," in Proc.Wireless Networking 2013.
11. P. Macharla, and M. Misra "A QoS Routing Protocol for Delay-Sensitive Applications in Mobile Ad Hoc Networks" 2nd International ICST Workshop on Intelligent Networks: Adaptation, Communication & Reconfiguration 2008.
12. S. Tabatabaei and K. Tabatabaei "Routing and quality of service support for mobile Ad hoc networks"in proceeding of International conference on Computer Engineering and Technology, Chengdu, china, April 2010
13. L.Chen and W. Heinzelman, "QoS-Aware Routing Based on bandwidth Estimation for Mobile Ad Hoc Networks," IEEE Journal on Selected Areas in Communication, 2005.
14. P. R. J. Jeni, A.V. Julia "A novel QoS frame work for MDSRAODV Routing Protocol in MANET" International Journal of Computer Science and Network Security, October, 2010.

15. Q.Xue and A.Ganz, “Ad hoc QoS on-demand routing (AQOR) in mobile ad hoc networks,”J. Parallel Distrib. Comput., Aug. 2014.
16. Ronan de Renesse, Mona Ghassemian, VasilisFriderikos, A. Hamid Aghvami, "Adaptive Admission Control for Ad Hoc and Sensor Networks Providing Quality of Service," Technical Report, Center for Telecommunications Research, Kingís College London, UK, May 2005.
17. Drs. Baruch Awerbuch & Amitabh Mishra, ‘Medium Access Control (MAC) Protocols for Ad hoc Wireless Networks’, May 14, 2011
18. A Quality Of Service-Aware Routing Based On gossiping Algorithm For Ad Hoc Wireless Networks By Ahyoung Lee B.E., Hansung University, 2015
19. Ronan de Renesse, Mona Ghassemian, "Adaptive Admission Control for Ad Hoc and Sensor Networks Providing Quality of Service," Technical Report, Center for Telecommunications Research, King ís College London, UK, May 2011.
20. Network simulator-2 (NS-2) Notes and Documentation), The VINT Project, April 10, 2001.
21. Thomas Williams & Colin Kelley ‘An Interactive Plotting Program -gnuplot’
22. Jirkaklaue, & Berthold Rathke. Evalvid–A Framework for Video Transmission and Quality Evaluation http://www.tkn.tu-berlin.de/publications/papers/evalvid.pdf
23. K.Chih-Heng, C. Shieh, W.Hwang, and A.Ziviani," An Evaluation Framework for More Realistic Simulations of MPEG Video Transmission ", J. Inf. Sci, Vol. 24, No. 2, 2008.
24. Chin-Heng, Ke. How to evaluate MPEG video transmission using the ns-2 simulator http://140.116.72.80/~smallko/ns2/Evalvid_in_NS2.htm
25. Ksentini, A. and Gueroui, A.; “Toward an improvement of H.264 video transmission over IEEE 802.11e through a cross-layer architecture,” IEEE Communications Magazine, Jan. 2006.
26. C. Perkins, E. Royer, S. Das, and M. Marina, "Performance comparison of two on-demand routing protocols for ad hoc networks," ZEEE Personal Communications Magazine, Feb 2011.

APPENDIX – A

EVALVID TOOLS INTEGRATION WITH NS2

Experimentation of MPEG-4 video transmission is carried out using Evalvid tool. Evalvid tool is an Evaluation Framework for MPEG video transmission in NS2 environment. Following are the steps to integrate evalvid tool with NS2:

Step 1: packet.h needs to be modified to add frametype_ and sendtime_ field in common folder.

Step 2: Modify agent.h to add frame type for MPEG 4 transmission.

Step 3: myevalvid folder is created under ns-2 and the filesmyudp.cc, myudp.h, yudpsink2.cc, myudpsink2.h, and mytraffictrace2.ccare added to the folder.

Step 4: Modifythe file tcl/lib/ns-default.tcl to add default values.

Step5: Makefileneeds to be modified by adding mympeg/myudp.o, mympeg/myudpsink2.o and mympeg/mytraffictrace2.o in the OBJ_CC list.

Step 6: Recompile NS2 by using configure and make commands

APPENDIX – B

Video Preparation and Evaluation Steps

Step 1: Encode the raw yuv video with ffmpeg. fmpeg command is used for compression.

In this step raw QCIF YUV video is converted to m4v.

./ffmpeg .exe -s cif -vcodec mpeg4 -r 30 -b 64 -g 30 -i foreman_qcif.yuv foreman_qcif.m4v

Step 2: Use MP4Box to convert m4v to mp4 MP4Box generates the hint tracks. The server uses these hint tracks to transmit packets over the network.

./MP4Box.exe –hint –mtu 1024 –fps 30 –add foreman_qcif.m4v foreman_qcif.mp4

For some (most) video quality assessment methods you need a reference video. This is either the original YUV file before the encoding or the YUV file created by decoding the coded video.

./ffmpeg -i foreman_qcif.mp4 foreman_qcif_ref.yuv

Step 3: Video Trace File Generation.

The video trace file contains the frame number, type and size and the number of segments in case of frame segmentation. The mp4trace sends an mp4-file per UDP to a specified destination host.

./mp4trace.exe –f –s 192.168.0.2 12346 foreman_qcif.mp4 > foreman_qcif.st

The foreman_st file contains the information about the I, P and B frames. This data file is used in tcl script in NS2 by the source to transmit packets to the destination.

Step 4: Run the tcl script. After simulation, ns2 will create two files, sd_be and rd_be. The file sd_be is to record the sending time of each packet while the file rd_be is used to record the received time of each packet.

nsvideotest.tcl

Step 5: **G**enerate the received video file (err.cmp).

./et.exe sd_foreman_0 rd_foreman_0 foreman_qcif.st 1 1

Step 6: Comparison and reconstruction of the received file.

./etmp4.exe sd_foreman_0 rd_foreman_qcif.st2 foreman_qcif.mp4 Received.mp4

The Evalvid tool etmp4 compares the sender's timestamp file against the receiver's timestamp file. Through this comparison, and the original MP4 video and the video trace file, the tool reconstructs the received MP4 video. During this reconstruction process, the tool also measures delay per frame and frame loss rate.

Step 7: The generation of the received Yuv file: using the ffmpeg codec and the received video.

./ffmpeg.exe -i received.mp4 received.yuv

The YUV video file will be used to calculate the PSNR of video quality in next step.

Step 8: The generation of the PSNR metric: using the psnr Evalvid tool which compares the original and the received Yuv files in order to calculate the PSNR per frame.

./avgpsnr.exe 176 144 420 foreman_qcif.yuvforeman_qcife.yuv

APPENDIX - C

Simulation Script Code for MANETs Routing Protocol Comparison

```
# ==================================================================
# Define options
# ==================================================================
setval(chan)     Channel/WirelessChannel
setval(prop)     Propagation/TwoRayGround
setval(netif)    Phy/WirelessPhy
setval(mac)      Mac/802_11
#set val(ifq)    CMUPriQueue
setval(ifq)      Queue/DropTail/PriQueue  ;# interface queue type
#set val(ifq)    Queue/DTail/PriQ
setval(ll)        LL
setval(ant)      Antenna/OmniAntenna
setval(x)         500 ;# X dimension of the topography
setval(y)         500 ;# Y dimension of the topography
setval(ifqlen)   50 ;# max packet in ifq
setval(seed)     1.0
setval(adhocRouting)  (AODV/DSDV/DSR)
setval(nn)       50 ;# how many nodes are simulated
setval(cp)       "cbr-20packets"
setval(sc)       "scen-p2"
setval(stop)     100 ;# simulation time

Mac/802_11 set dataRate_ 2Mb
Mac/802_11 set basicRate 1Mb
# ==================================================================
# Main Program
# ===================================================================
# Initialize Global Variables
# create simulator instance
set ns_ [new Simulator]
# setup topography object
settopo [new Topography]
# create trace object for ns and nam
settracefd [open 20-AODV.tr w]
setnamtrace [open 20-AODV.nam w]
$ns_ trace-all $tracefd
$ns_ namtrace-all-wireless $namtrace $val(x) $val(y)
# define topology
$topoload_flatgrid $val(x) $val(y)
# Create God
set god_ [create-god $val(nn)]
# define how node should be created
#global node setting
```

```
setchan [new $val(chan)]
$ns_ node-config -adhocRouting $val(adhocRouting) \
        -llType $val(ll) \
        -macType $val(mac) \
        -ifqType $val(ifq) \
        -ifqLen $val(ifqlen) \
        -antType $val(ant) \
        -propType $val(prop) \
        -phyType $val(netif) \
        -channel $chan \
        -topoInstance $topo \
        -agentTrace ON \
        -routerTrace ON \
        -macTrace OFF

# Create the specified number of nodes [$val(nn)] and "attach" #them
# to the channel.
        for {set i 0} {$i < $val(nn) } {incr i} {
        set node_($i) [$ns_ node]
        $node_($i) random-motion 0 ;# disable random motion
        }
# Define node movement model
        puts "Loading connection pattern..."
        source $val(sc)
# Define traffic model
        puts "Loading scenario file..."
        source $val(cp)
# Define node initial position in nam
        for {set i 0} {$i < $val(nn)} {incr i} {
# 20 defines the node size in nam, must adjust it according to your scenario
# The function must be called after mobility model is defined
        $ns_ initial_node_pos $node_($i) 20
}
# Tell nodes when the simulation ends
        for {set i 0} {$i < $val(nn) } {incr i} {
        $ns_ at $val(stop).0 "$node_($i) reset";
}
$ns_ at $val(stop).0002 "puts \"NS EXITING...\" ; $ns_ halt"
puts $tracefd "M 0.0 nn $val(nn) x $val(x) y $val(y) rp $val(adhocRouting)"
puts $tracefd "M 0.0 sc $val(sc) cp $val(cp) seed $val(seed)"
puts $tracefd "M 0.0 prop $val(prop) ant $val(ant)"
puts "Starting Simulation..."
$ns_ run
```

DECLARATION

I hereby declare that this project is my original work and has not been presented for a degree award in this or any other university. I am presenting the thesis for examination for the degree of Masters of Science in Computer Engineering.

Name Ashebir Dereje Bonger

Signature …………………………….

Date ……………………………..