Performance Analysis of UDP and UDP-Lite for Transmission of Different Video Codecs over MANETs

Puneet Kaur
Assistant Professor
Sri Guru Granth Sahib World University, Fatehgarh Sahib, Punjab
puneetkaurbajwa@gmail.com

ABSTRACT

Mobile Ad-hoc Network (MANET) is a self-configurable wireless network in which nodes communicate with each other without any centralized access points or base stations. MANETs are widely used for communication among various users because they provide the facility to connect anytime at any place. Major traffic sent over these networks is various forms of multimedia (mostly video), which prefer to be delivered erroneous than being rejected or arriving late. UDP and UDP-Lite are preferred transport layer protocols for transmitting such data. Both protocols provide are unreliable, connectionless and has low protocol processing overhead. The aim of this paper is to analyze the performances of transport protocols, UDP and UDP-Lite for transmitting various video codecs under different network conditions over MANETs.

Keywords: H.261, H.263, H.263+, MANET, MPEG-4, OPNET Modeler, UDP, UDP-Lite.

1. INTRODUCTION

Mobile Ad-hoc Networks (MANETs) have become one of the most promising and successful technologies in recent years [15]. The usage of Mobile Ad-hoc without infrastructure is increasing because they provide flexible connectivity to end users, offering an easy and viable access to the network and its services. Various interactive multimedia applications, like 3D graphics, voice and video etc. constitutes majority of the traffic sent over these wireless networks [3].

The demands of end users are increasing, as the technology is improving. A wide variety of new multimedia applications are being invented daily. High channel capacity and better Internet connectivity has become a basic requirement for all the customers for fast access to the information [5]. In the past few years, YouTube has accounted for 27% of all video traffic sent and received over the Internet. The emerging technologies of video compression are currently a very exciting and challenging time for this area of research. MPEG-4, H.261, H.263, H.236+, H.264 etc. are the various video codecs used widely over the Internet [4]. Various networks are used to send and receive multimedia over the Internet among which MANETs are preferred among others because of ease of installation; decreased headache of physical connections such as wiring and customers can connect anytime and anywhere. Transportation and on-time delivery of these real-time multimedia applications is of major concern. Most popular transport protocols used for these delay sensitive applications are UDP and UDP-Lite. Both protocols provide unreliable services and involve less protocol-processing overhead, help delivering multimedia applications more efficiently. In UDP, either whole packet is check-summed, i.e. the data sent is also checked for errors or none of it. Whereas, UDP-Lite is an extended version of UDP in which partial checksum of packets is possible [11]. In this manner, the corrupted data delivered to the destination is also accepted, making this protocol more favorable to be used in sending and receiving various multimedia applications that require on-time delivery. Performance of UDP and UDP-Lite is evaluated and analyzed for various network performance parameters and video codecs (used for transmitting video multimedia over internet) in this paper.

OPNET Modeler 14.0 is used to evaluate the performance of UDP and UDP-lite in terms of network delay, retry threshold and network load, for various video codecs by altering various network parameters like number of nodes (users), traffic, bandwidth and mobility speed.

The paper has been organized as follows. Literature review is covered in section two. Section three presents the basic overview of transport protocols, UDP and UDP-Lite. Section four includes a detailed explanation of video codecs used for study. In the fifth section, the implementation over Modeler 14.0 and simulation results thus obtained are discussed. The conclusion is given in section six.

2. LITERATURE REVIEW

The study of literature shows that extensive research has been done to evaluate and compare the performance of UDP and UDP-Lite for various multimedia applications.

Beginning from the basic introduction to the protocols, various performance parameters like video quality, delay, error rate etc., have been studied for both protocols. UDP, a simple, connectionless, transport layer protocol was proposed which provided minimum protocol mechanism, no delivery of acknowledgments and duplicate protection to the packets once sent, for on-time transmission of specific time-restricted applications over the Internet like various multimedia contents, text, audio, graphics, video etc. [6]. A lightweight version of UDP transport protocol, UDP-Lite was then introduced with increased flexibility in the form of partial checksum. [11].

In past few years, video-based web traffic continues to grow and dominate the Internet through social networking and catch up TV. YouTube has accounted for 27% of all video traffic sent and received over the Internet. The emerging technologies of video compression are currently a very exciting and challenging time for this area of research [4]. To compare and analyze their performance for an audio coding (24 bytes of data) and a PCM audio (8 kHz sampling frequency) for various transmission methods i.e. UDP, UDP + CRTP, UDP-lite and UDP-lite + CRTP, various simulations of transport protocols, UDP and UDP-lite have been done by Lars-Åke Larzon et al. [9]. Flexible check-summing schemes for wireless network architecture, which support bit-error resilient codecs, were proposed by Amoolya Singh et al. [1]. They modified the transport layer protocols by implementing UDP-lite and PPP-lite to the transport and link layer protocols respectively. As a result, UDP-lite gave better results and significantly better video quality than UDP. An approach was suggested to the use of MPEG-4 and UDP-Lite for the next generation transport for IP multimedia. The authors concluded that UDP-Lite provides more flexibility by enabling delivery of partially corrupted packets and also could provide better video quality especially over an error prone environment [15]. A comparison and transmitting of multimedia streams over three different WLANs scenarios by using OPNET simulator was presented by Mohamed M. Abo Ghazala, et al. [7]. The scenarios were implemented with different number of hosts per Access Point (AP). Performances were evaluated using end-to-end delay, traffic received (bps), data dropped (bps), delay (sec), load (bps), media access delay (sec) and throughput (bps). By using UDP and UDP-lite as transport layer protocols respectively, the effects of wireless channel on the quality of the transmitted real-time Ultrasound Video were studied, and the efficiency of using both is evaluated on the basis of Bit Error Rate (BER) and Peak Signal to Noise Ratio (PSNR) [3].

Xinjie Chang has compared several network simulators like, REAL, INSANE, NetSim, OPNET Modeler, NS-2, VINT, U-Net and Harvard. A network simulation scenario containing several Ethernet subnets connected by an ATM network backbone has been modeled to compare end-to-end delay and packet loss ratio [17]. OPNET (Optimized Network Engineering Tool) was stated as the most powerful software simulation package.

3. OVERVIEW OF UDP AND UDP-Lite

User Datagram Protocol (UDP)

UDP is a connectionless transport layer protocol. It involves a procedure to send messages with a minimum of protocol mechanism. The protocol is simple, transaction oriented, but the delivery and duplicate protection are not guaranteed. If so, arrive in order, appear duplicated, or go missing without notice. UDP has protocol identification number, called protocol identifier, 17 (21 octal) when used in the Internet Protocol [6].

The UDP Header contains four fields of 2 bytes each. It is as shown in figure 1.

Source Port (2 bytes)	**Destination Port** (2 bytes)
Length (2 bytes)	**Checksum** (2 bytes)
Data (if any)…	

Figure 1: UDP Header Format [6].

The field Source Port in the header indicates the port of the sending process, and may be assumed to be the port to which a reply should be addressed. Destination Port has a meaning within the context of a particular Internet destination address. Length is the length in octets of this user datagram including this header and the data. Checksum is the 16-bit one's complement of the one's complement sum of a pseudo header of information from the IP header, the UDP header, and the data, padded with zero octets at the end (if necessary) to make a multiple of two octets [16].

UDP-Lite

UDP-lite (Lightweight User Datagram Protocol) is also a transport layer protocol, similar to the User Datagram Protocol. UDP-Lite provides a checksum with an optional partial coverage. When using this option, a packet is divided into a sensitive part (covered by the checksum) and an insensitive part (not covered by the checksum). Errors in the insensitive part will not cause the packet to be discarded by the transport layer at the receiving end host. UDP and UDP-Lite have similar syntax and semantics. The similarities also ease implementation of UDP-Lite, since only minor modifications are needed to an existing UDP implementation [10].

The UDP-lite header format also contains 4 fields of 2 bytes each. It is as shown in the figure 2.

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Source Port (2 bytes)	**Destination Port** (2 bytes)
Checksum Coverage (2 bytes)	**Checksum** (2 bytes)
Data (if any)…	

Figure 2: UDP-Lite Header Format [11].

Source Port is an optional field, when meaningful, it indicates the port of the sending process. Destination Port has a meaning within the context of a particular Internet destination address. Checksum Coverage is the number of octets, counting from the first octet of the UDP-Lite header, which is covered by the checksum. The UDP-Lite header must always be covered by the checksum. This means that the value of the Checksum Coverage field must be either 0 or at least 8. The receiver must discard an UDP-Lite packet with a Checksum Coverage value of 1 to 7. The Checksum field is the 16-bit one's complement of the one's complement sum of a pseudo-header of information collected from the IP header, the number of octets specified by the Checksum Coverage (starting at the first octet in the UDP-Lite header) [11].

3. VIDEO CODECS

Video codecs are used to transmit various video formats efficiently over the Internet. A video codec is a device or software that performs video compression or decompression for digital video. The compression techniques being used usually employ lossy data compression [15].

Video codecs attempt to represent a fundamentally analog data set in a digital format. The encoding process the video is compressed to send over the Internet more efficiently. The decoding process is an inversion of each stage of the encoding process. The one stage that cannot be exactly inverted is the quantization stage. There, a best-effort approximation of inversion is performed. This part of the process is often called "inverse-quantization" [12]. The whole process of coding and decoding is shown in figure 3.

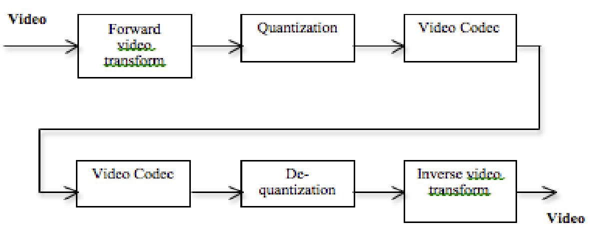

Figure 3: Video Coding and Decoding.

Various video codecs are available for transmitting video multimedia over the internet. Such as MPEG-4, MPEG-1, MPEG-2, H.261, H.263, H.263+, H.264, Divx, Cinepak etc. The lossy video codecs used for this study are:

H.263+: H.263+ is the second edition of the ITU-T H.263 international video-coding standard. It retained the entire technical content of the original version of the standard, but enhanced H.263 capabilities by adding several annexes, which can substantially improve encoding efficiency and provide other capabilities (such as enhanced robustness against data loss in the transmission channel) [4].

H.263: H.263 was originally designed as a low-bitrate compressed format for videoconferencing, developed by the ITU-T Video Coding Experts Group (VCEG) in 1995/1996 as one member of the H.26x family of video coding standards in the domain of the ITU-T. H.263 has since found many applications on the Internet (on sites such as YouTube, Google Video, MySpace, etc.). The original version of the RealVideo codec was based on H.263 [4].

H.261: H.261 is a ITU-T video coding standard, agreed in November 1988. It was the first video codec that was useful in practical terms over the Internet. H.261 was originally designed for transmission over ISDN lines on which data rates are multiples of 64 kbit/s [12].

MPEG-4: MPEG-4 is a video compression technology developed by MPEG. It belongs to the MPEG-4 ISO/IEC standards. It is a discrete cosine transform compression standard, similar to previous standards such as MPEG-1 and MPEG-2. Several popular codecs including DivX, Xvid and Nero Digital implement this standard [15].

4. SIMULATION AND RESULTS

OPNET Modeler 14.0

OPNET Modeler 14.o has been used to simulate various network scenarios over MANET. It is a GUI- based software tool used for simulating and modeling networks. It also provides a GUI integrated debugging and analysis.

For the base network simulation, data rate (bandwidth) of 11 Mbps is chosen. The various physical and media access control layer parameter values used in our experiment are according to IEEE 802.11b default values. The various simulation parameters are as per following Table 1.

Table 1: Simulation Parameters and their Value(s)

S. No.	Simulation Parameter	Value
1	Number of nodes	40
2	Simulation time	1 hr
3	Area covered	4000x4000m
4	Traffic Source	CBR
5	Mobility Model	None

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6	Operational mode	802.11g
7	Data rate	11 mbps
8	Command Mix (Get/Total) for ftp	50%
9	Videoconferencing	30 fps
10	Audio	G.711 silence

To analyze the performance of UDP and UDP-Lite, six scenarios have been created by changing the number of nodes, bandwidth, traffic and mobility in the base network scenario for MANET.

Various scenarios implemented are as under:

- **Scenario 1:** A base scenario consisting 40 nodes with data rate of 11 mbps.
- **Scenario 2:** A scenario has been implemented by changing the number of nodes from 40 nodes to 20 nodes with data rate 11 mbps.
- **Scenario 3:** A scenario consisting 40 nodes in which data rate has been reduced to 2 mbps.
- **Scenario 4:** A scenario consisting 40 nodes in which data rate has been reduced to 5.5mbps.
- **Scenario 5:** A scenario consisting 40 nodes in which extra Constant Bit Rate (CBR) traffic has been introduced with data rate 11 mbps.
- **Scenario 6:** A scenario consisting 40 nodes in which mobility (Random Way Point) has been implemented with data rate 11mbps.

The simulations have been run for one hour for each scenario and the results obtained from them have been analyzed. Like, for Scenario 1, ten simulations of one hour each has been done to obtain the result graphs for UDP. The same simulation criterion is repeated for other scenarios and in case of UDP-Lite scenarios also.

The performance of each protocol is evaluated on the basis of seven network performance parameters, i.e. media access delay, retransmission attempts, buffer overflow, retry threshold, network load, throughput and network delay.

Media Access Delay

Media access delay is the time taken for the data to reach the MAC layer until it is successfully transmitted on the wireless medium. It is measured in seconds.

The results obtained for media access delay for each scenario for UDP are shown in Figure 4. It is increasing at much faster pace in the first 10 minutes of simulation due to increase in the number of nodes competing to gain access of medium for each scenario. In the rest 50 minutes, the media access delay differs at an average of 0.03 seconds for each scenario.

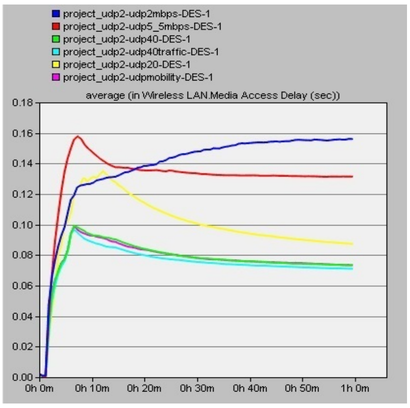

Figure 4 Average Media Access Delay – UDP.

Media access delay for each scenario for UDP-Lite are shown in Figure 5. In the same manner, as seen for UDP, the media access delay for various scenarios of UDP-Lite is also increasing at a fast pace in the first 10 minutes of simulation. In the rest 50 minutes of simulation, for Scenario 4 (base scenario), the media access delay remains constant i.e. 0.06 seconds, increasing at much slower rate for Scenario 1 (2 mbps data rate) and decreasing for all other scenarios.

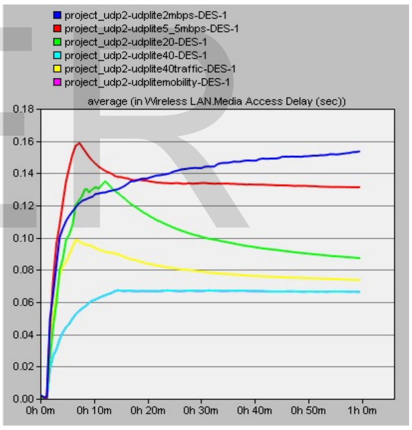

Figure 5 Average Media Access Delay – UDP-Lite.

Retransmission Attempts

It is the total number of attempts by all wireless LAN MACs in the network until either packet is successfully transmitted or it is discarded as a result of reaching short or long retry limit.

Retransmission attempts for each scenario for UDP are shown in Figure 6. In the first 10 minutes, the retransmission attempts are high for Scenario 1 (2 mbps data rate), Scenario 2 (5.5 mbps data rate), Scenario 4 (base scenario) and Scenario 5 (increased traffic). After 10 minutes of simulation, the retransmission attempts are decreasing for each scenario. It is least for Scenario 3 (20 nodes) throughout the simulation time.

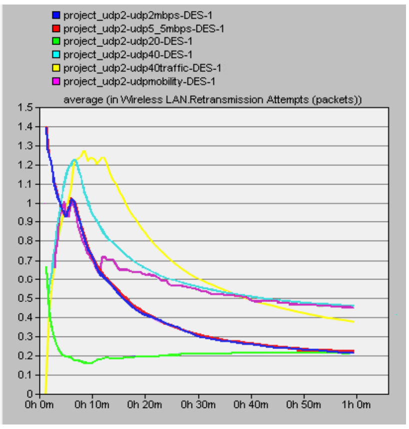

Figure 6 Average Retransmission Attempts – UDP.

For each scenario for UDP-Lite, the retransmission attempts are described in Figure 7. Scenario 6 (mobility) has the highest retransmission attempts, whereas they are lowest for Scenario 3 (20 nodes).

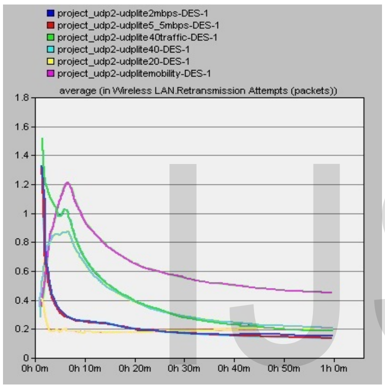

Figure 7 Average Retransmission Attempts – UDP-Lite.

Buffer Overflow

A buffer overflow is caused by the queuing and access delays in the source node, all transit node delays, and the receiver buffer delay in the destination node.

The buffer overflow obtained for each scenario for UDP are presented in Figure 8. It tends to be highest if the traffic is increased and lowest when the bandwidth (network data rate) is decreased.

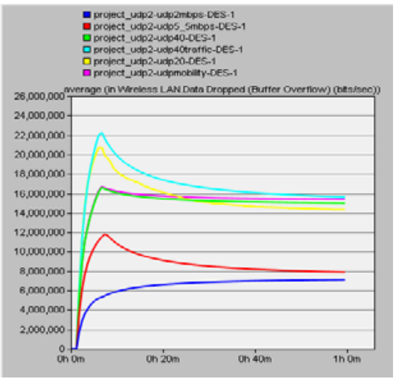

Figure 8 Average Buffer Overflow – UDP.

Buffer overflow for each scenario for UDP-Lite are presented in Figure 9. The buffer overflow is increasing for all scenarios in the first 10 minutes of simulation. And after decreasing to some extent, it becomes constant in all the scenarios.

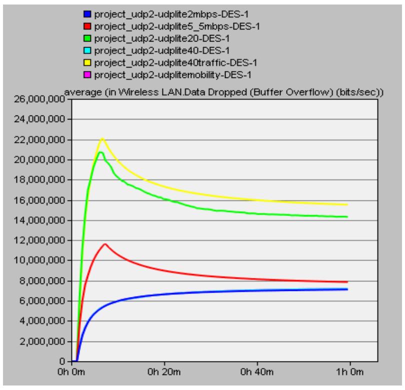

Figure 9 Average Buffer Overflow – UDP-Lite.

Retry Threshold

The retry threshold value is the number of times to try the message flow transaction again. The value of retry threshold is always kept greater than 1.

The results obtained for retry threshold for each scenario for UDP are shown in Figure 10. In the first 10 minutes of the simulation, the retry attempts are increasing for all scenarios, after that the retry attempts are decreasing continuously for all scenarios.

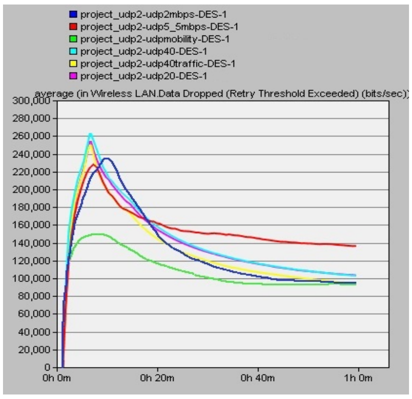

Figure 10 Average Retry Threshold – UDP.

The retry attempts are least for Scenario 4 (base scenario) and highest for Scenario 2 (20 nodes). The results for retry threshold for each scenario for UDP-Lite are presented in Figure 11.

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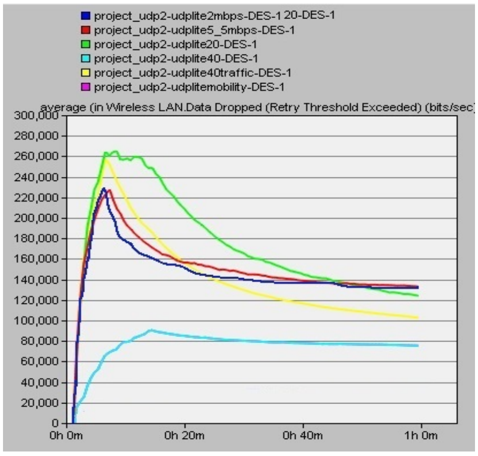

Figure 11 Average Retry Threshold – UDP-Lite.

The retry threshold is maximum for Scenario 3 (20 nodes) and Scenario 6 (mobility), and minimum in case of Scenario 4 (base scenario).

Network Load

Network Load is the amount of data (traffic) being carried by the network at a particular time. Network load tells about how efficiently the network performs under a given condition

The network load for each scenario for UDP are shown in Figure 12. The network load is increasing at much faster rate in the first 10 minutes, and stabilizes to some extent for rest of the simulation time.

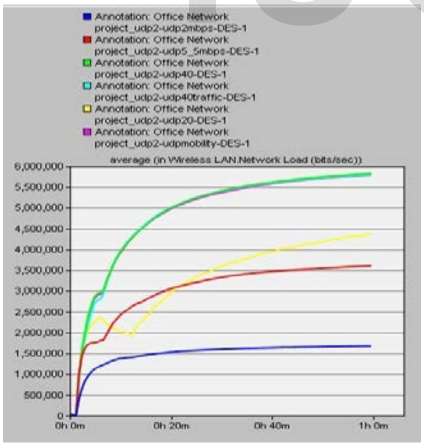

Figure 12 Average Network Load – UDP.

The results obtained for network load for each scenario for UDP-Lite are presented in Figure 13 .

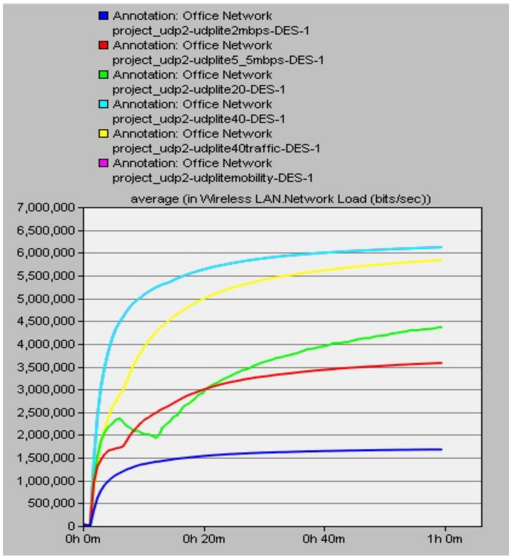

Figure 13 Average Network Load – UDP-Lite.

Throughput

Throughput or network throughput is the average rate of successful message delivery over a communication channel. It measured in bits per second (bit/s or bps).

Average throughput obtained for each scenario for UDP are presented in Figure 14. It is increasing for all the scenarios throughout the simulation time. Except for Scenario 1 (2 mbps data rate), the throughput becomes constant after first 20 minutes of simulation.

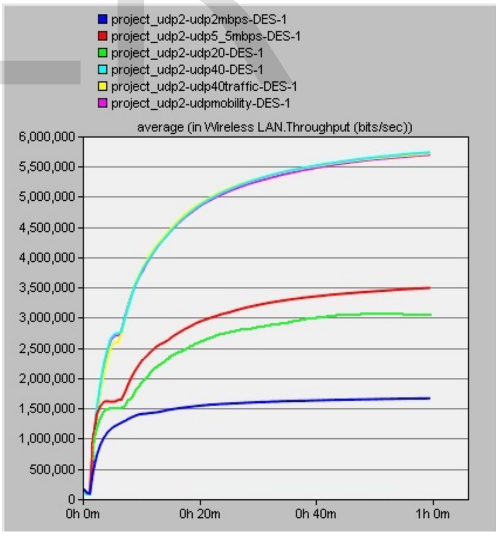

Figure 14 Average Throughput – UDP.

Scenario 1 (2 mbps data rate) has the least throughput, whereas Scenario 4 (base scenario), Scenario 5 (increased traffic) and Scenario 6 (mobility) have similar as well as the highest throughput.

The results obtained for throughput for each scenario for UDP-Lite are shown in Figure 15. It is increasing for all the scenarios throughout the simulation time.

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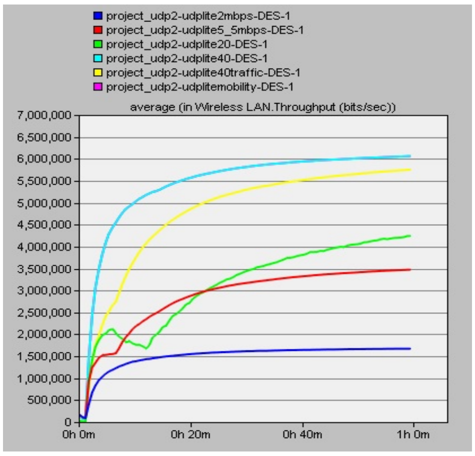

Figure 15 Average Throughput – UDP-Lite.

Network Delay

Network delay refers to the time taken for a packet to be transmitted across a network from source to destination. It is calculated as the sum of transmission delay, propagation delay and processing delay within the network.

The results obtained for network delay for each scenario for UDP are presented in Figure 16. The network delay is much higher in the first 7-8 minutes of the simulation. For Scenario 1 (2 mbps data rate) and Scenario 2 (5.5 mbps data rate) it id increasing at a much slower pace afterwards. However, for rest of the scenarios, the network load decreases at slower pace.

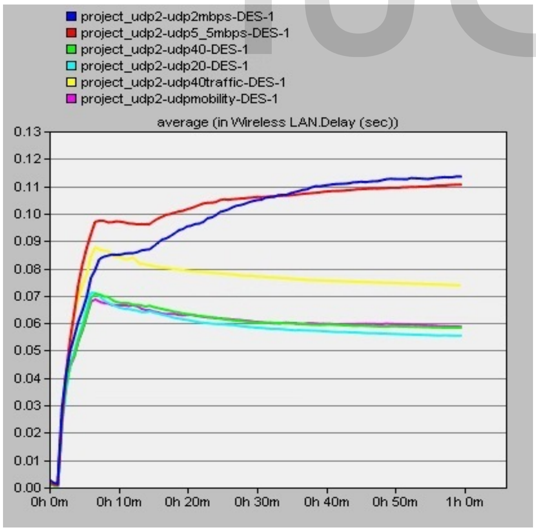

Figure 16 Average Network Delay – UDP

It is maximum in Scenario 2 (2 mbps data rate) and minimum for Scenario 3 (20 nodes). The network delay for each scenario for UDP-Lite are shown in Figure 17 . The network delay is much higher in the first 10 minutes of the simulation.

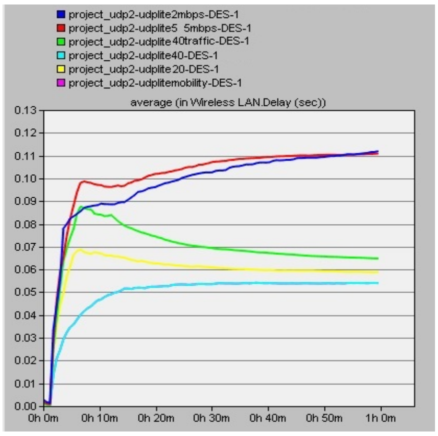

Figure 17 Average Network Delay – UDP-Lite.

It is maximum in Scenario 2 (5.5 mbps data rate) and minimum for Scenario 4 (base scenario). For various network performance parameters, the best and the worst performance of UDP are as listed in table 2.

Table 2 Performance Evaluations - UDP

S. No.	Network Parameter	Best Performance	Worst Performance
1.	Media Access Delay	Scenario 5	Scenario 1
2.	Retransmission Attempts	Scenario 3	Scenario 5
3.	Buffer Overflow	Scenario 1	Scenario 5
4.	Retry Threshold	Scenario 6	Scenario 2
5.	Network Load	Scenario 1	Scenario 4, Scenario 5, Scenario 6
6.	Throughput	Scenario 4, Scenario 5 Scenario 6	Scenario 1
7.	Network Delay	Scenario 3	Scenario 2

The best and worst performance of UDP-Lite for various network parameters, is shown in table 3.

Table 3 Performance Evaluations – UDP-Lite

S. No.	Network Parameter	Best Performance	Worst Performance
1.	Media Access Delay	Scenario 4	Scenario 1
2.	Retransmission Attempts	Scenario 3	Scenario 6
3.	Buffer Overflow	Scenario 1	Scenario 3, Scenario 4, Scenario 5
4.	Retry Threshold	Scenario 4	Scenario 3, Scenario 6
5.	Network Load	Scenario 1	Scenario 4, Scenario 6

6.	Throughput	Scenario 4, Scenario 6	Scenario 1
7.	Network Delay	Scenario 4, Scenario 6	Scenario 2

With decrease in network bandwidth performance of both protocols degrades for each parameter. The performance of UDP is enhanced when the mobility factors are changed, but UDP-Lite showed degraded performance to some extent. However, when the number of nodes is reduced to half and traffic is increased, both protocols showed stable performances.

5. CONCLUSION

In recent years, transmission of interactive multimedia over MANETs is the most widely used technologies for communication by the users. The transmission of various video multimedia content is crucial. UDP and UDP-Lite (transport protocols) are well known for transmitting multimedia over the Internet. By changing various network parameters, various network simulations have been performed to analyze the performances of both protocols for various video codecs. On the basis of performance of each protocol for various scenarios, UDP shows best overall performance with increase in mobility speed and worst for decrease in network bandwidth. UDP-Lite has the best performance under basic network conditions (base scenario). And, the performance of UDP-Lite degrades with increase in mobility speed.

Future work may include various other network parameters, such as Quality of Service (QOS), Bit Error Rate (BER), Terrain Modeling Module (TTM, specified in OPNET Modeler), etc. Enhancements can be suggested and implemented to improve the efficiency of the protocols, UDP and UDP-Lite.

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