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Performance Comparison of transport protocols, UDP and UDP-Lite for transmission of Different Video Codecs over MANETs

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Abstract- Nowadays, the most widely used technologies for communication are, Mobile Ad-hoc Networks (MANETs) and usage of multimedia. MANET is a self-configurable wireless network in which nodes communicate with each other without any centralized access points or base stations. It prefers to transport the data without any overhead of establishing a connection prior to send data to avoid network congestion. The recent trends in technology have shown that most of the contents (data) send over the Internet are interactive multimedia, which prefer to be delivered erroneous than being discarded, or arriving late. The transport protocol, UDP came as a solution to these problems. It provides no reliability and has low protocol processing overhead. An enhanced version of UDP, called UDP-Lite was also introduced almost a decade ago, which has been specifically designed for transmitting various multimedia applications more efficiently. The aim of this paper is to compare the performances of UDP and UDP-Lite for network delay, buffer overflow and network load for transmitting various video codecs by changing various network parameters.
Keywords: ***MANET, multimedia, OPNET Modeler, UDP, UDP-Lite, video codecs, WLAN.***

1. INTRODUCTION

Wireless Local Area Networks has become one of the most promising and successful technologies in recent years [17]. The usage of Mobile Ad-hoc Networks i.e. WLANs without infrastructure is increasing because they provide the facility to connect anytime at any place. MANETs provide free wireless connectivity to end users, offering an easy and viable access to the network and its services. Another trend is increased use of interactive multimedia applications, like 3D graphics, voice and video etc. over the 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, having varying demands from the underlying network protocol suite (TCP/IP Protocol Suite). High channel capacity and better Internet connectivity has become a basic requirement for all the customers for fast access to the information [6]. 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 [5]. 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, help delivering multimedia applications more efficiently and involves less protocol-processing overhead. In UDP, either whole packet is checksummed, 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 [12]. 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 compared for various network parameters and multimedia applications.

In this paper, OPNET Modeler 14.0 is used to compare 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 nodes, traffic, bandwidth and mobility.

The paper has been organized as follows. A brief discussion about the 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. A description of the OPNET Modeler 14.0 is given in fifth section with assumptions and requirements and simulation results thus obtained. The conclusion is given in section six.

2. LITERATURE REVIEW

UDP, a simple, connectionless, transport layer protocol was proposed which provided minimum protocol mechanism, no delivery acknowlegments 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. [7]. A lightweight version of UDP transport protocol, UDP-Lite was then introduced with increased flexibility in the form of partial checksum. [12]. 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. [10]. Flexible checksumming 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 [17]. 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. [8]. 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) [4].

Xinjie Chang have compared several network simulators like, REAL, INSANE, NetSim, OPNET Modeler, NS-2, VINT, U-Net and Harvard simulator are also discussed. 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 [19]. OPNET (Optimized Network Engineering Tool) was stated as the most powerful software simulation package.

3. OVERVIEW OF UDP AND UDP-LITE

In this section, a brief discussion about the transport protocols, UDP and UDP-Lite is given. The header formats of both the protocols are discussed in detail.

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 [7].

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

```
 0      7 8     15 16    23 24    31
+--------+--------+--------+--------+
|     Source      |   Destination   |
|      Port       |      Port       |
+--------+--------+--------+--------+
|                 |                 |
|     Length      |    Checksum     |
+--------+--------+--------+--------+
|
|          data octets ...
+---------------- ...
```

Figure 1: UDP Header Format [7].

The fields in the header format of UDP are as described below [17]:

- **Source Port** is an optional field, when meaningful, it indicates the port of the sending process, and may be assumed to be the port to which a reply should be addressed in the absence of any other information. If not used, a value of zero is inserted.
- **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. (This means the minimum value of the length is eight.)
- **Checksum** is the 16-bit one's complement of the one's complement sum of a pseudo header of information from the IP header.

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. When the checksum covers the entire packet, which should be the default, UDP-Lite is semantically identical to UDP [12].

UDP and UDP-Lite have similar syntax and semantics. Applications designed for UDP may therefore use UDP-Lite instead. The similarities also ease implementation of UDP-Lite, since only minor modifications are needed to an existing UDP implementation [11].

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

```
 0                 15 16                31
+--------+--------+--------+--------+
|     Source      |   Destination   |
|      Port       |      Port       |
+--------+--------+--------+--------+
|    Checksum     |                 |
|    Coverage     |    Checksum     |
+--------+--------+--------+--------+
|                                   |
:              Payload              :
|                                   |
+-----------------------------------+
```

Figure 2: UDP-Lite Header Format [12].

#

The various fields are as described below [12]:

- **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. Checksum Coverage of zero indicates that the entire UDP-Lite packet is 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).

4. 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 [17].
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" [13].
The whole process of coding and decoding is shown in figure 3.

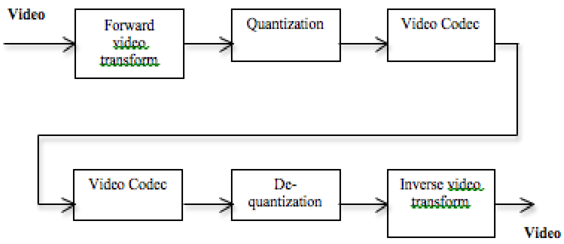

Figure 3: Video Coding and Decoding.

List of lossy video codecs is as given below:
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) [5].
H.263: H.263 was originally designed as a low-bitrate compressed format for videoconferencing,ra 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 [5].
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 [13].
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 [17].

5. SIMULATION AND RESULTS

OPNET Modeler 14.0

The OPNET Modeler software is used to simulate entire networks to analyze and compare the performance of a network. This includes all layers of the OSI reference model, from physical links up to application demands. Its primary function, is the support of network planning groups and application developers.
For the simulations, the workflow of a project could be, create a project followed by a baseline scenario. Then, the network topology we want to use in the scenario is either imported or created. After that, the results and reports to be collected are chosen. The results are gathered and analyzed. Finally, iterations are specified by duplicating the scenario and changing parameters.
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
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 compare 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 1 hour for each scenario and the results obtained from them have been compared in terms of network delay, netwok load and buffer overflow. Like, for Scenario 1, 10 simulations of 1 hour each has been done to obtain the result graphs for UDP. The same techniques are repeated for other scenarios and in case of

UDP-Lite scenarios also. After that, the results are averaged to evaluate and compare their performance for various scenarios. Network delay is calculated as the sum of transmission delay, propagation delay and processing delay within the network. Network Load is the amount of data being carried by the network at a particular time. Buffer overflow is caused by the queuing and access delays at the sending end, all transit node delays, and the receiver buffer delay in the destination node.

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. Network Delay is generally used to evaluate the performance of a network for specific applications or data to be send over that network. Higher is the network delay, lower is the performance evaluated for the network.

The graphs obtained for network load for each scenario are described below.

Figure 4 shows that in the first 10 minutes of simulation, the network delay for UDP and UDP-Lite is increasing at fast pace. The increase in network delay is due to increase in number of nodes trying to access the channel. UDP has lesser network delay than UDP-Lite in the first 25 minutes of simulation. Whereas, UDP-Lite has lesser network delay as compared to UDP in the remaining 35 minutes.

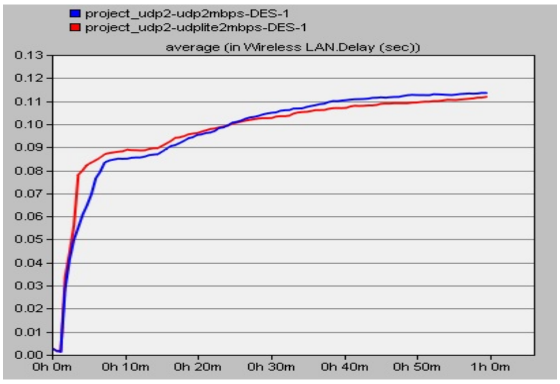

Figure 4 Average Network Delay – Scenario 1.

In Figure 5, the network delay for UDP and UDP-Lite is same throughout the 1 hour simulation. However, a slight increase occurs in case of UDP-Lite.

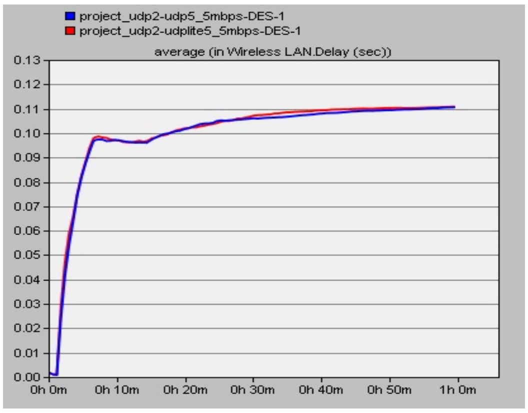

Figure 5 Average Network Delay – Scenario 2.

The network delay for UDP-Lite is quite less than that of UDP for this scenario, as shown in Figure 6. The average difference between the network delays for both protocols is 0.12 seconds.

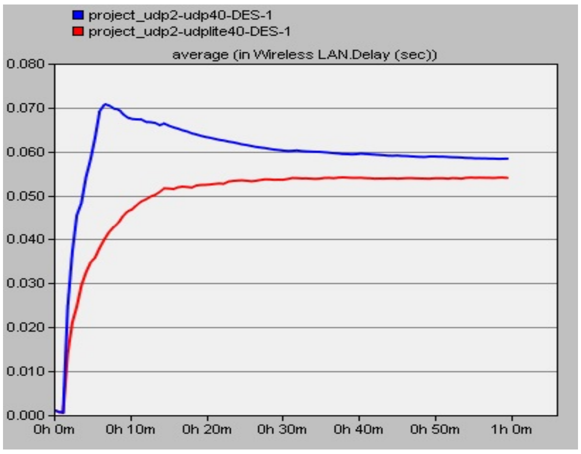

Figure 6 Average Network Delay – Scenario 3.

In Figure 7, the network delay for UDP-Lite is quite less than that of UDP for this scenario with an average difference of 0.12 seconds between the network delays for both protocols.

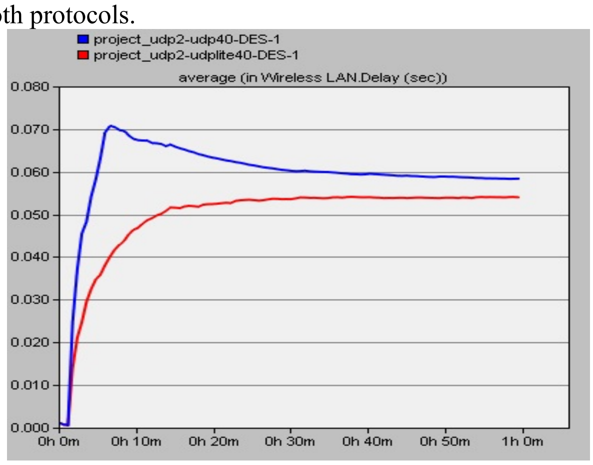

Figure 7 Average Network Delay – Scenario 4.

In Figure 8, the network delay for UDP and UDP-Lite is same in the first 8 minutes of simulation. However, a slight increase occurs in case of UDP afterwards.

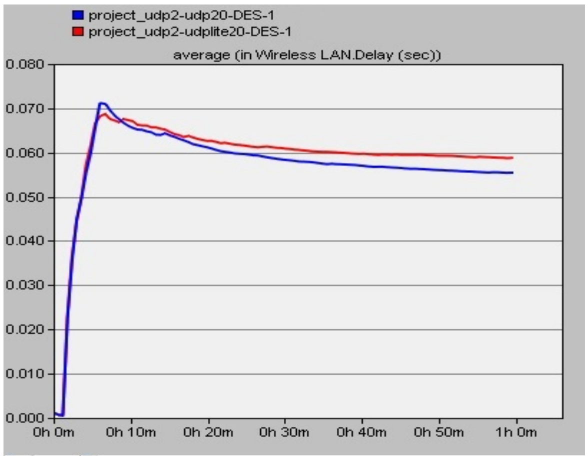

Figure 8 Average Network Delay – Scenario 5.

In Figure 9, the network delay for UDP is quite less than that of UDP-Lite for this scenario with an average difference of 0.15 seconds.

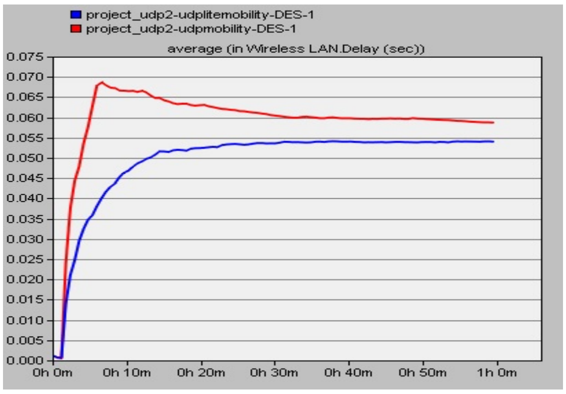

Figure 9 Average Network Delay – Scenario 6.

Network Load

Network Load is the amount of data (traffic) being carried by the network at a particular time. The network load is varies from time to time. Network load is highest at peak hours, when network consumption is high i.e. maximum number of users are connected to the network. Network load tells about how efficiently the network performs under a given condition.

The graphs obtained for network load for each scenario are described below. In Figure 10, the network load for both protocols is almost same throughout the whole simulation. The network load is increasing at much faster pace in the first 15 minutes of simulation, due to increase in the number of sending nodes within the medium.

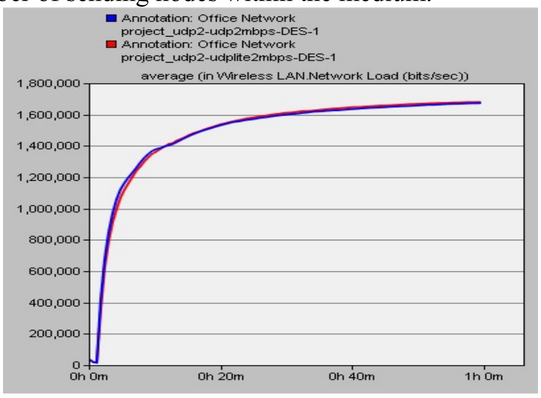

Figure 10: Average Network Load – Scenario 1.

In Figure 11, the network load for protocol UDP is slightly higher than that of protocol UDP-Lite.

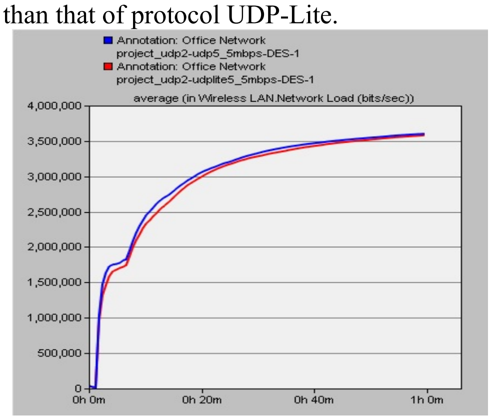

Figure 11: Average Network Load – Scenario 2.

In Figure 12, the network load for the protocols, UDP and UDP-Lite changes at equal pace throughout the simulation.

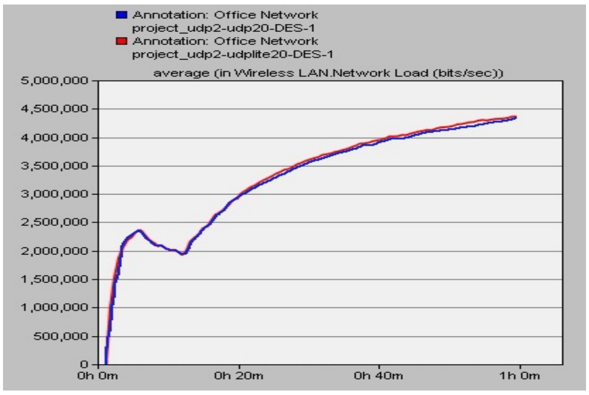

Figure 12: Average Network Load – Scenario 3.

In Figure 13, the network load is increasing for both protocols. The network load curve in case of UDP is visibly higher than that of UDP-Lite.

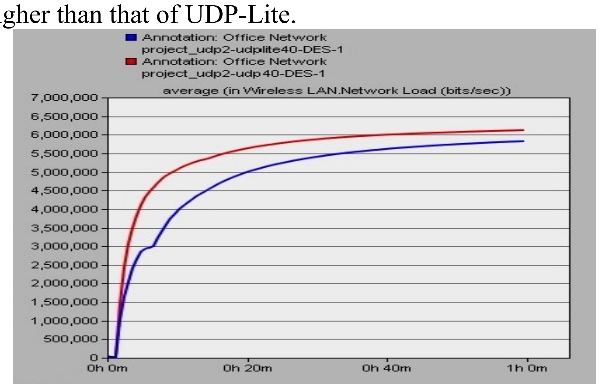

Figure 13: Average Network Load – Scenario 4.

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In Figure 14, the network load curves for protocols, UDP and UDP-Lite are increasing with same pace throughout the 1 hour simulation.

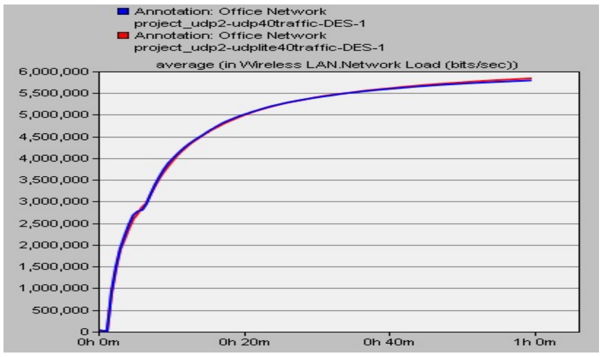

Figure 14: Average Network Load – Scenario 5.

In Figure 15, the network load for UDP is visibly higher than that of UDP-Lite. The results for both protocols are almost same as obtained for Scenario 4.

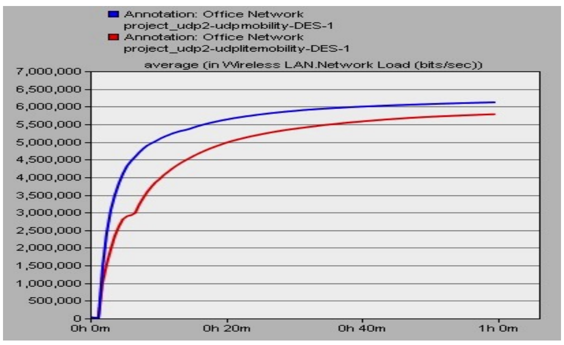

Figure 15: Average Network Load – Scenario 6.

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. Long delays are intolerant in all types of data contents transmitted over the Internet. If buffer overflow occurs at a rapid rate, the performance of the network is affected.

The graphs obtained for buffer overflow for each scenario are described below. In Figure 16, in the 1 hour simulation, the buffer overflow for protocols, UDP and UDP-Lite increases at equal pace for throughout the simulation. However, the rate of increment is quite high in first 20 minutes of the simulations and becomes near to stable in the rest 40 minutes.

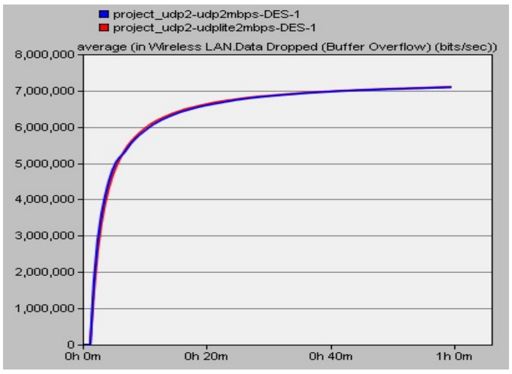

Figure 16 Buffer Overflow – Scenario 1.

In Figure 17, the buffer overflow for protocols, UDP and UDP-Lite increases and decreases at equal pace respectively.

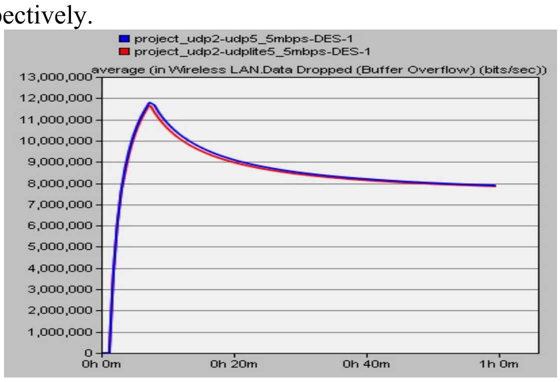

Figure 17 Buffer Overflow – Scenario 2.

In Figure 18, the buffer overflow for protocols, UDP and UDP-Lite again increases and decreases at equal pace respectively.

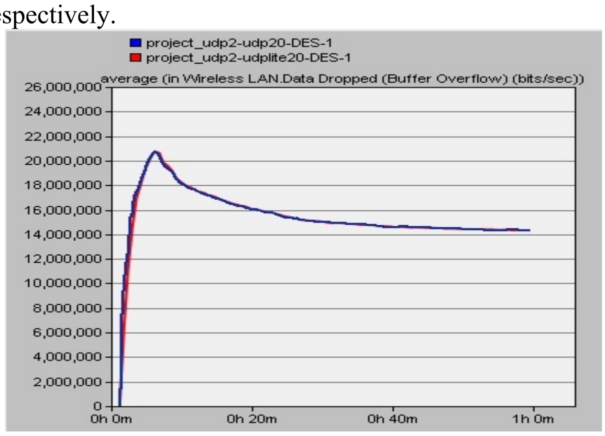

Figure 18 Buffer Overflow – Scenario 3.

In Figure 19, in the first 5 minutes of the simulation, buffer overflow for UDP increases at much faster pace with a difference of 10,000,000 bits/s, but after that remains almost constant for both protocols.

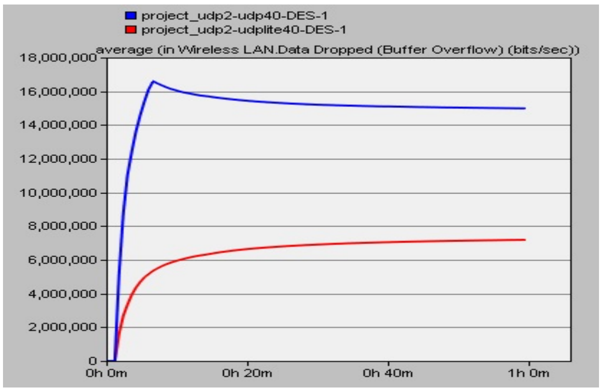

Figure 19 Buffer Overflow – Scenario 4.

In Figure 20, the buffer overflow for protocols, UDP and UDP-Lite again increases and decreases at equal pace respectively.

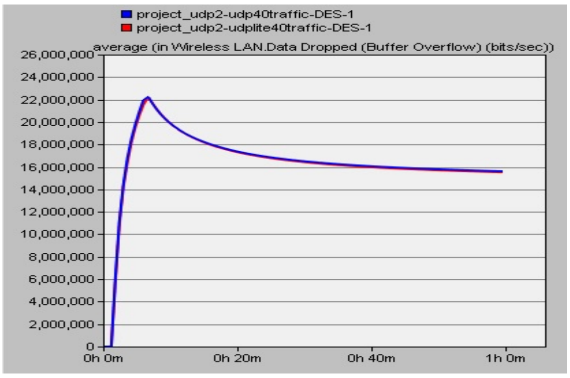

Figure 20 Buffer Overflow – Scenario 5.

In Figure 21, in the first 5 minutes of the simulation, buffer overflow for UDP increases at much faster pace with a difference of 10,000,000 bits/s, but after that remains almost constant for both protocols.

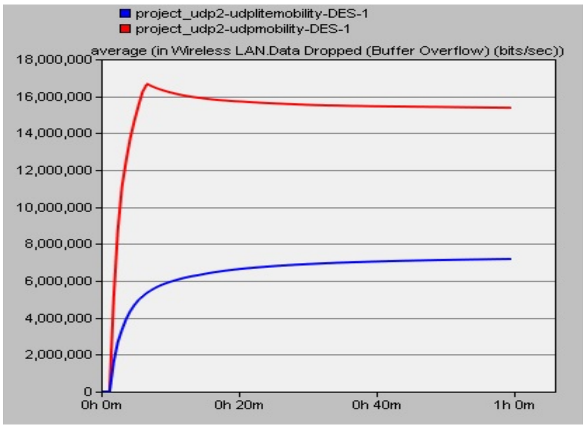

Figure 21 Buffer Overflow – Scenario 6.

The overall results for both protocols, UDP and UDP-Lite for all scenarios in case of network delay, network load and buffer overflow are as given in Table 6.

Table 2 Overall Results

Scenario No.	Network Delay	Network Load	Buffer Overflow
Scenario 1	UDP, UDP-Lite	UDP, UDP-Lite	UDP, UDP-Lite
Scenario 2	UDP	UDP-Lite	UDP, UDP-Lite
Scenario 3	UDP-Lite	UDP, UDP-Lite	UDP, UDP-Lite
Scenario 4	UDP-Lite	UDP-Lite	UDP-Lite
Scenario 5	UDP-Lite	UDP, UDP-Lite	UDP, UDP-Lite
Scenario 6	UDP	UDP-Lite	UDP-Lite

6. CONCLUSION

In recent years, multimedia and WLANs are the most widely used technologies for communication, by the users. The transmission of various video multimedia content is decisive. 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 and compare the performances of both protocols for various video codecs. It has been concluded that the overall performance UDP-Lite is slightly well than UDP (for the used simulation conditions used in this study), in terms of network delay, network load and buffer overflow. For Scenario 1 (2mbps data rate), Scenario 3 (20 nodes), Scenario 4 (base scenario) and Scenario 5 (increased traffic), UDP-Lite has performed better for all three network performance parameters than UDP, i.e. has lesser network delay, network load and buffer overflow. Whereas, for Scenario 2 (5.5mbps data rate) and Scenario 6 (mobility), UDP has lesser network delay as compared to UDP-Lite and UDP-Lite has lesser network load and buffer overflow.

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