

**PERFORMANCE EVALUATION OF TCP, UDP AND DCCP FOR  
VIDEO TRAFFICS OVER 4G NETWORK**



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2015**

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## Abstrak

Sistem mudahalihat Generasi Keempat (4G) telah digunakan secara lebih meluas berbanding generasi terdahulu seperti 3G dan 2G. Antara sebabnya termasuklah kadar penghantaran data 4G yang lebih tinggi, dan ia menyokong keseluruhan fungsi multimedia. Selain itu, sokongannya terhadap lokasi geografi yang luas membolehkan teknologi tanpa wayar menjadi semakin canggih. Matlamat utama 4G ialah bagi membolehkan komunikasi berasaskan suara dilaksanakan oleh pengguna tanpa batasan. Bagi memenuhi matlamat tersebut, kajian ini berusaha menjawab persoalan-persoalan berikut: (1) adakah protokol lama sesuai dengan teknologi baru ini; (2) protokol manakah mempunyai prestasi terbaik; Selain itu, kajian ini juga mempersoalkan dan (3) protokol manakah yang mempunyai kesan terbesar terhadap truput, lengah, dan kehilangan paket; Persoalan-persoalan tersebut amat penting, ditimbulkan bagi menilai kesan 4G terhadap protokol-protokol utama (khususnya *User Datagram Protocol* (UDP), *Transmission Control Protocol* (TCP), dan *Datagram Congestion Control Protocol* (DCCP)). Menggunakan *Network Simulator-3* (NS-3), prestasi penghantaran MPEG-4 video merangkumi aspek truput, lengah, kehilangan paket, dan nisbah kadar penghantaran paket serta kesesakan pada stesyen utama menggunakan UDP, TCP, dan DCCP telah dinilai melalui teknologi *Long Term Evolution* (LTE) 4G. Hasil ujian menunjukkan DCCP mempunyai truput dan lengah yang lebih baik. Namun, jumlah kehilangan paket adalah lebih tinggi berbanding UDP dan TCP. Berdasarkan dapatan tersebut, DCCP adalah disarankan sebagai protokol penghantaran bagi video waktu nyata membuat penghantaran video.

**Keywords:** 4G, LTE, TCP, UDP, DCCP, Kawalan Kesesakan, protokol penghantaran

## Abstract

Fourth Generation (4G) system has been used more widely than the older generations 3G and 2G. Among the reasons are that the 4G's transfer rate is higher and it supports all multimedia functions. Besides, its' supports for wide geographical locus makes wireless technology gets more advanced. The essential goal of 4G is to enable voice-based communication being implemented endlessly. To achieve the goal, this study tries to answer the following research questions: (1), are the old protocols suit with this new technology; (2), which one has the best performance and, (3) which one has the greatest effect on throughput, delay, packet delivery ratio and packet loss. The aforementioned questions are crucial in the performance evaluation of the most famous protocols (particularly User Datagram Protocol (UDP), Transmission Control Protocol (TCP), and Datagram Congestion Control Protocol (DCCP)) within the 4G environment. Through the Network Simulator-3 (NS-3), the performance of transporting MPEG-4 video stream including throughput, delay, packet loss, and packet delivery ratio are analyzed at the base station through UDP, TCP, and DCCP protocols over 4G's Long Term Evolution (LTE) technology. The results show that DCCP has better throughput, and lesser delay, but at the same time it has more packet loss than UDP and TCP. Based on the results, DCCP is recommended as a transport protocol for real time video.

**Keywords:** 4G, LTE, TCP, UDP, DCCP, Congestion Control, Transport Protocol

## **Acknowledgement**

First of all, I would like to take this opportunity to express my sincere appreciation to my supervisor Dr. Shahrudin Awang Nor for his invaluable advice, coaching, support, giving of practical exposure and fruitful discussion throughout this thesis without which I would not have succeeded in carrying out this research. I am also grateful to many individuals who have contributed to the development of the ideas and the completion of my thesis. Last but not least I would like to thank all my course mates provided morale support and guidance to me for completing this thesis.

Wisam Abduladheem Kamil

2015



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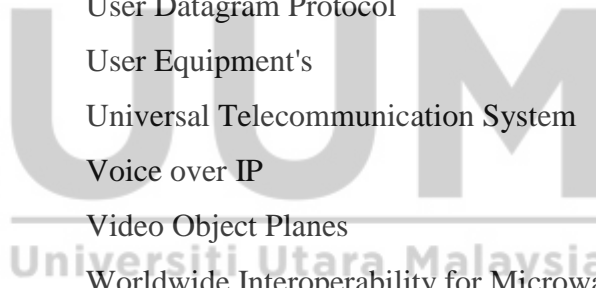
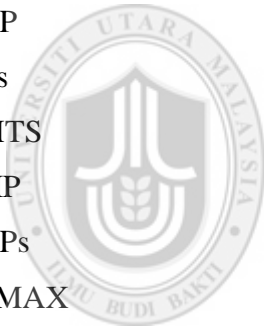


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## List of Abbreviations

1G	First Generation
2G	Second Generation
3G	Third Generation
3GPP	Third Generation Partnership Project
4G	Fourth Generation
AVC	Advanced Video Coding
CLEP	College Level Examination Program
CPU	Central Processing Unit
DCCP	Datagram Congestion Control Protocol
DCE	Direct Code Execution
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name System
ECN	Explicit Congestion Notification
EDGE	Enhanced Data rates for GSM evolution
ENBs	Evolved Node B
GPLv2	General Public License, version 2
GSM	Global System for Mobile Communication
GSoC	Google Summer of Code
GUI	Graphical User Interface
HSPA	High Speed Packet Access
HVXC	Harmonic Vector eXcitation Coding
IMT-Advanced	International Mobile Telecommunications-Advanced
IPS	Internet Protocol Suite
IR	Infrared Wireless
ITU	International Telecommunication Union
ITU-R	International Telecommunication Union-R
LTE	Long Term Evolution
LTE BS	LTE-Base Station
MAC	Media Access Control
MIMO	Multiple-Input Multiple-Output

MPEG4	Moving Picture Experts Group
NACK	Negative Acknowledgement
NS-3	Network Simulator
PHY	Physical Layer
QoS	Quality of Service
RIP	Routing Information Protocol
RTP	Real-time Transport Protocol
RTT	Round Trip Time
SC-FDMA	Single-Carrier Frequency Division Multiple Access
SCTP	Stream Control Transmission Protocol
SS	Subscriber Station
SVC	Scalable Video Coding
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
UEs	User Equipment's
UMTS	Universal Telecommunication System
VoIP	Voice over IP
VOPs	Video Object Planes
WIMAX	Worldwide Interoperability for Microwave Access



# **CHAPTER ONE**

## **INTRODUCTION**

Wireless communications become an everyday commodity. It has evolved from being an expensive technology for a few selected individuals to today's ubiquitous systems used by a majority of the world's population. Wireless communication technologies are often divided into generations. First Generation (1G) was the analog radio systems of the 1980s. Second Generation (2G) was the first digital wireless systems. Third Generation (3G) was the first wireless systems handling broadband data.

The Long-Term Evolution (LTE) is often called the Fourth Generation (4G) [1]. Wireless communication network under Information and Communication Technologies (ICT) is perhaps the most vital element reshaping the economic growth of the world. With the evolution from 1G to 2G and from 3G to 4G, the technology shifts from telecommunication to multimedia communication. Nowadays mobile technology has changed the perspective of user towards the real-time world by enabling people to live in both business and social environment. These trends induced the invention of 5G which, comparing with 4G, will have 1000 times the system capacity, 10 times the spectral efficiency, 25 times power efficient and data rate up to 10Gbps for low speed and 2Gbps for high speed moving mobiles [2]. As a result of the advancements in wireless communication the network traffic has been increased.

The LTE delivered higher data rates and met the burgeoning data demand [3]. In LTE deployment, three transport layer protocols are the most recommended and

widely used, they are User Datagram Protocol (UDP) [4], Transmission Control Protocol (TCP) [5] and Datagram Congestion Control Protocol (DCCP) [6]. The data delivering rate is significantly influenced by the performance of the transport protocol that is used in the scenario of wireless networking [7].

Recent network traffic analysis and predictions indicate that video accounts for a growing proportion of the network traffic. Cisco, the worldwide leader company in IT, predicted that video transmitted to and from wireless devices will be accounted more than 80% of all consumer internet traffic in 2019, up from 66% in 2014. These kinds of predictions have been motivated by the developments of high efficient video coding standards that significantly improved the video comparison efficiency. The term “H.264” refers to the most important family of these standards including MPEG-2 Moving Picture Experts Group (MPEG-2), MPEG-4 Part 2 Moving Picture Experts Group (MPEG-4), H.264/MPEG-4 Part 10 Advanced Video Coding standard (H.264/AVC) and Scalable Video Coding (SVC) [8]. Overall, due to the vast improvements in wireless communication technologies and video comparison efficiency, it is highly important to consider evaluating the performance of video network transport in networking studies.

### **1.1 Problem Statement**

In today’s telecommunications, the video traffic has burgeon under the development of LTE, the truly underlying access technology of 4G networks [1][8]. In LTE deployment, three transport layer protocols are the most recommended and widely studied, they are UDP, TCP and DCCP [9]. Although the LTE deployment is rapidly pace, there is a lack of performance evaluation of its protocols. Therefore, an extensive analysis is needed to evaluate the performance of various protocols for

high end applications like multimedia applications. The problematic behavior of the three protocols in multimedia applications entails highlighting the pros and cons of their performance [10]. The best performing protocol in video transferring even does not characterized because of the conflicted conclusions. This drawback due to several reasons. Firstly, the impression [11]–[19] of existing comparison studies does not consider the performance of the TCP, UDP and DCCP protocols in transferring video under LTE access environment. Secondly, although the studies have considered the protocols their conclusions about the effect of number of nodes on the performance are still diversified. The emphasis is to determine which protocol can better meet the quality of service for MPEG-4 video over wireless local area network [18]. Therefore, providing an integral analyzing study about the performance of the three protocols in LTE environment will help developers and researchers in choosing the proper protocol to be used in video traffic.

### **1.2 Research Questions**

1. How to design simulation scenario for UDP, TCP and DCCP transport layer protocols, which are using to transmission video over LTE in 4G environment by NS-3?
2. What is the impact of UDP, TCP and DCCP transport layer protocols on video transmission in 4G environment?

### **1.3 Research Objectives**

1. To design verity simulation scenario to performance UDP, TCP and DCCP transport layer protocols in terms of video transmission in 4G environment by using NS-3 simulator.

2. To analyze the performance of UDP, TCP and DCCP transport layer protocols in term of delay, throughput, packet delivery ratio and packet loss as performance metrics to highlight the variety impacts of transport layer protocols behaviors for video transmission in 4G.

#### **1.4 Scope of Study**

The demand on video traffic has been increased. Transport layer of OSI model is one layer that plays the main role in this functionality. This study focuses on the widely used protocols for video data transmission, i.e. UDP, TCP and DCCP. Several video coding standards have been developed for the purpose of video transmission efficiency. Most of the new efficient video comparison standards are derived from MPEG-4. Therefore, this dissertation highlighted MPEG-4 as main video coding standard and scoped out other low efficient video coding standards that is not designed for video traffic. Faster data transfer, i.e. 4G, is the prominent environment in wireless communication. Three basic techniques are included in this service, they are WiMax, HSPA+ and LTE.

The LTE technology is designed to be the only real 4G network environment. Therefore, this research sticks with LTE because of its coverage is stable comparing with the former two techniques. Apart from that, the simulation tool is crucial for modeling any real phenomenon. Without such realistic modeling, several characteristics and important insights in real environment will be lost. In contrast to old simulators, NS-3 simulator fully supports LTE technology in terms of visualization, modeling and adaptable scripting which allow to be integrated with



real networks performance. Hence, it has been selected for conducting experiments and to explore the impact of each of the selected protocols.

### **1.5 Significance of Study**

This research is significant since this study is effectiveness of the 4G through video traffic stream, especially the delay and video transmission time. The performance of TCP, UDP and DCCP protocols is studied, supporting to determine which one is useful in 4G environment. In fact, the vast majority of consumers nowadays have used smart phone, and they are definitely looking for 4G supportive devices. Hence, they are searched the performance of the video payload. Therefore, the present study evaluates the video traffic and reveals which protocol is more useful for the 4G smart phones and their applications.

### **1.6 Thesis Organization**

**Chapter One** shows the Introduction to the area of research. It briefly describes the background, current scenario, and scope of the study.

**Chapter Two** will study and discuss the literature review of the UDP, TCP and DCCP and working of these protocols. This chapter covers all the functional differences between these transport layer protocols.

**Chapter Three** describes baseline of the NS-3 simulation tools and problem of statement.

**Chapter Four** consists of the techniques of LTE network and its gateways and performance of network. This chapter also consists of pseudo code of TCP, UDP and DCCP protocol.

**Chapter Five** shows the results of various parameters like average delay, throughput, packet delivery ratio and loss packets with comparison scenario. In last, chapter 6 consists of conclusion and descriptive future work.



## **CHAPTER TWO**

### **BACKGROUND AND RELATED WORK**

#### **2.1 Introduction**

Long Term Evolution (LTE), normally promoted as 4G LTE, is a rivet for remote correspondence of high velocity information for versatile telephones and information terminals. It depends upon the Global System for Mobile Communication (GSM)/ Enhanced Data rates for GSM evolution (EDGE) and Universal Telecommunication System (UMTS)/ High Speed Packet Access (HSPA) system innovations, expanding the limit and pace utilizing an alternate radio interface together with centre system enhancements.

In spite of the fact that it is advertised as a 4G remote administration, LTE (as specified in the Third Generation Partnership Project (3gpp) Release 8 and 9 report arrangement) does not fulfil the specialized necessities that the 3gpp consortium has received for its new standard era, which were initially set by the International Telecommunication Union-R (ITU-R) association in its International Mobile Telecommunications-Advanced (IMT-Advanced) determination. Notwithstanding, because of showcasing weights and the critical progressions that Worldwide Interoperability for Microwave Access (WIMAX), HSPA, and LTE bring to the first 3G innovations, International Telecommunication Union (ITU) later chose that LTE, together with the previously stated advances, could be called 4G. However, TCP throughput and the transmission postponement of UDP throughout the handover process by utilizing our outside LTE advances. Therefore, the important protocols that transmit the Moving Picture Experts Group (MPEG4) through them and over

LTE are introduced in this chapter. Section 2.2 discusses various transport layer protocols. Section 2.3 focuses on the user datagram protocol while Sections 2.4 and 2.5 present the transmission control protocol and datagram congestion control protocol, respectively. Section 2.6 presents the 4G technology/ LTE environment. Section 2.7 focuses on MPEG-4 video code standard. Section 2.8 concerns simulation and its circumstances. Section 2.9 discusses the close works in literature before the chapter is summarized in Section 2.10.

## 2.2 Transport Layer Protocols

The services and features of some transport layer protocols, i.e., UDP, TCP, and DCCP are shown in table 2.1. Each has their own features and relevance for particular application under specific environments.

*Table 2.1: Services and features provided by TCP, UDP and DCCP.*

Features and Services	TCP	UDP	DCCP
Reliable	Yes	No	No
Connection - Oriented	Yes	No	Yes
Congestion Control	Yes	No	Yes
Sequence Number	Yes	No	Yes

## 2.3 User Datagram Protocol

The UDP has been structured by David P. Reed [20] and is considered the backbone of the Internet Protocol Suite (IPS) [21]. However, the protocol does not have the ability for the handshaking mechanism to guarantee packet reliability, data integrity and packet ordering.

UDP is a connection-less protocol working on transport layer [22] . The header size of UDP protocol is 8 bytes including the fields source port address, destination port address, length and checksum. All fields are of 16 bits i.e., 2 bytes each. It is unreliable due to the lack of acknowledgement in the data transfer. Thus, an application program running over UDP should deal precisely with the issues of End-to-End communication that a connection-oriented protocol would have managed. These issues may be any of the retransmission for consistent delivery, flow control, packetization and reassembly, and congestion control etc. It is fast due to no connection establishment and tear down phase[4]. Therefore, it is more suited to small applications which do not need reliable connection. The most common use of UDP is in Domain Name System (DNS) services. To get the IP address for a requested URL from DNS, UDP is used as a transport layer protocol. Other application layer protocols which use UDP as a carrier protocol on transport layer are Dynamic Host Configuration Protocol (DHCP) [23], Routing Information Protocol (RIP) [24] and Voice over IP (VoIP) [25].

Nevertheless, Time-sensitive and Real-time applications, for example, video traffic and voice, are using UDP due to the dropping packets, which are preferable to delayed ones. Owing to the stateless nature of UDP, network applications, such as Trivial File Transfer Protocol, and online games, also use it as a transport protocol [26] [24]. At transport layer, the UDP is located in Figure 2.1.

#### **2.4 Transmission Control Protocol**

TCP is another Internet Protocol Suite (IPS) core protocol that functions well when two end-systems at a higher level interact. However, the stream of bytes provides

packet reliability through TCP [27] whereas this protocol also performs some management tasks, such as controlling rate and message during the regulating of traffic congestion and communication.

TCP acts as a transport layer that hides the underlying systems administration points of interest from correspondence provisions[5]. One of the best cases of TCP applications is the web browser [28]. So, other common main applications include web server, e-mail, and file transfer. In Figure 2.1, the TCP functioning layer has been shown.

## **2.5 Datagram Congestion Control Protocol**

The DCCP is a convention of the transport layer with dependable association setup, blockage control, and characteristic transaction competence [29]. However, the primary configuration goal and broadening over the conventional UDP is the affirmation of blockage control for datagram streams. At that point, DCCP has a scheduled outline that divides the focal part purpose of the convention from the usage of the blockage control instrument. DCCP is envisioned for multimedia functions, such as streaming media, which can be assisted from manipulation over the adjustments between delay and reliable in-order delivery. TCP may not be suitable for these applications because congestion control and reliability in-order delivery can result in arbitrarily long delays. UDP can avoid long delays, but for congestion control the governing application will have to deal on its own. DCCP provides built-in congestion control, including Explicit Congestion Notification (ECN) support, for unreliable datagram flows, avoiding the arbitrary delays related with TCP. A DCCP feature is a connection quality on whose value the two endpoints

make an agreement. Several advantages of a DCCP association are coordinated by characteristics. For example, congestion control mechanism used in the two half-connections. The endpoints attain the arrangement in the course of option of exchange negotiations in DCCP headers. The primary uses of DCCP protocol are round-trip time occasionally, such as in the initial values for the certain times. DCCP round-trip time measurements are performed by congestion control mechanisms[6]. According to RFC793, DCCP implementations follow TCP's general principle of robustness, i.e. "Be conservative in what you do, while be liberal in what you accept from others." DCCP is a transport layer protocol that deploys unicast, bidirectional connections of congestion-controlled and unreliable datagrams. Figure 2.1 locates the DCCP in OSI.

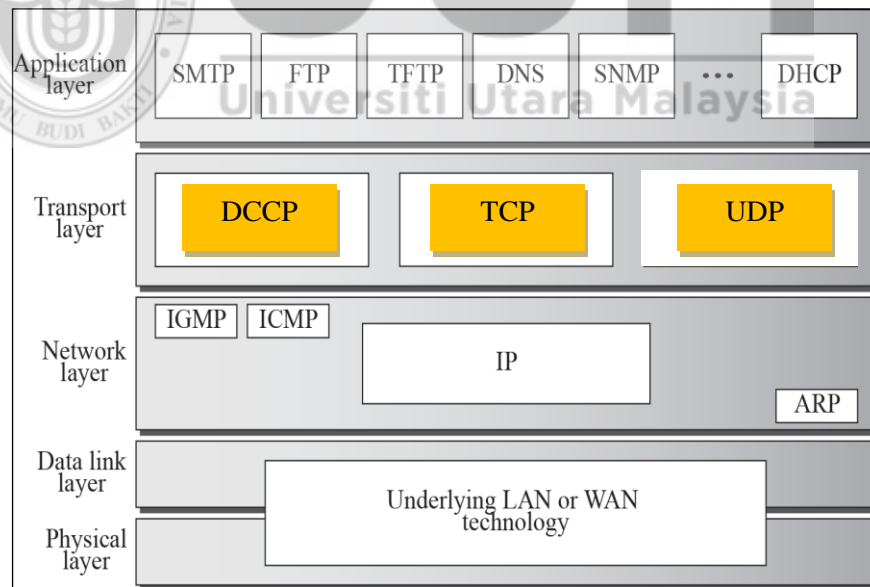


Figure 2.1: The DCCP location in OSI model.

## 2.6 Fourth Generation / Long Term Evolution

Fourth Generation blankets over billions of supporters; more than 80% of the worldwide versatile business sector [1]. However, the number of worldwide subscribers, in 2008, utilizing High-Speed Packet Access (HSPA) networks surpassed 70 million [30]. So, HSPA is a 3G evolution of GSM that supports high-speed data transmission by means of WCDMA technology.

The global use of HSPA technologies among clients and businesses have accelerated, representing continuous traffic growth for high-speed wireless networks worldwide, whereas extensive efforts are proceeding in the 3G Partnership Project (3GPP) to create a novel criterion for the development of GSM/HSPA technology towards a packet-optimized method known as LTE, with the intention of meeting the continuous demands in Internet traffic [31].

The main purpose of the LTE standard is to design plans for a new radio-access technology that can suitably handle higher data rates and is beneficial for low latency, and better spectral efficacy [32].

However, the spectral efficacy target for the LTE scheme is three to four times more than the existing HSPA scheme [33]. These uncompromising spectral efficacy targets need to push the technology envelope by using advanced air-interface mechanisms. For example, low-PAPR orthogonal uplink multiple access based on the Multiple-Input Multiple-Output (MIMO), Single-Carrier Frequency Division Multiple Access (SC-FDMA), inter-cell interference mitigation methods, multi-antenna technologies, low latency channel structure, and Single-Frequency Network (SFN) broadcast [30]. For the wireless, broadband data speed transaction, Figure 2.2



explains how the wireless data transfer. LTE technology will also continue to develop, with operators already making a considerable amount of progress in increasing the data speeds of their existing networks by adopting multiple-carrier LTE-A technologies. Therefore, while there remain monetisation and interconnect issues around LTE, these advancements will enable operators to offer many of the services that have been put forward in the context of 5G long before 5G becomes a commercial reality [2]. Therefore, this project intends to use LTE for the testing, explaining the performances over previous surviving data.

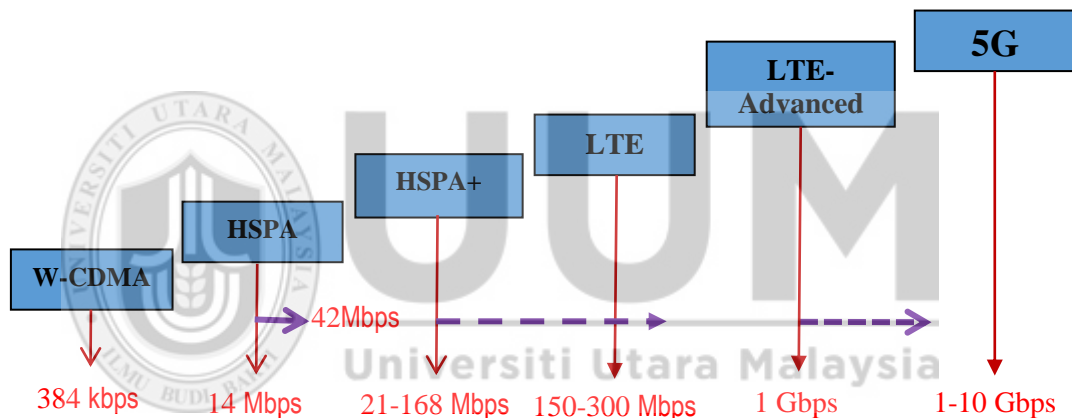


Figure 2.2: The grown of the telecommunication world.

## 2.7 Moving Picture Expert Group

The MPEG4 video coding standard is the most appropriate format for video communication over the Internet [31]. Utilized for low touch rates, MPEG 4 empowers true pictures to exist together with PC created partners. MPEG4 makes conceivable the detachment of genuine from workstation created pictures for distinctive medications emerging from the interface with clients [1], [28], [29], [34]–[37].

The primary role of the system is the competence for constant versatile encoding, which upgrades system use and empowers MPEG4 senders to be more receptive to changes in system conditions. MPEG4 creates feature in three separate edges, for example, (I, P, and B) that assist to encode unique parcels of the feature information in distinctive levels of value. The architecture of MPEG4 is shown in Figure 2.3.

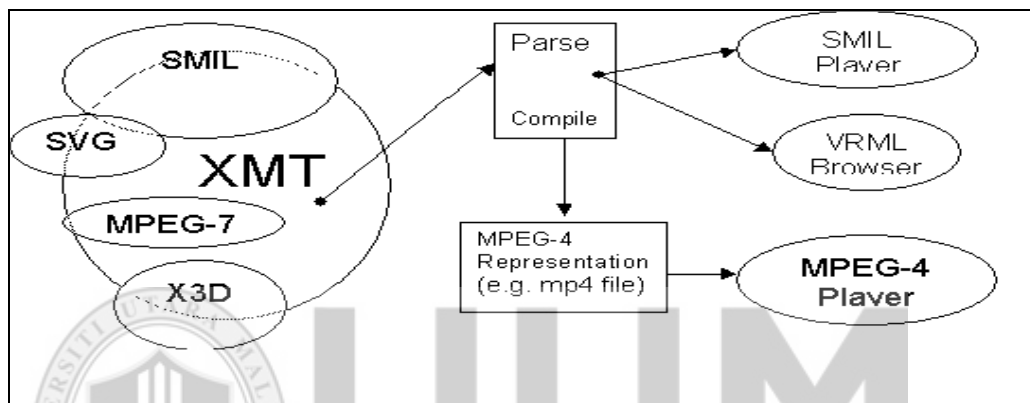


Figure 2.3: MPEG4 architecture [38].

Three separate sorts of edges are accessible in the MPEG-4 arrangement. Interceded casings, or I-outlines, are encoded freely and might be acknowledged as reference casing [28]. However, predicted frames or P-frames depend on preceding I or P-frames and consist of predicted motion data and error information. Thirdly, bi-directionally predicted frames or B-frames depend on both previous and next frames. These frames are shown in Figure 2.4.

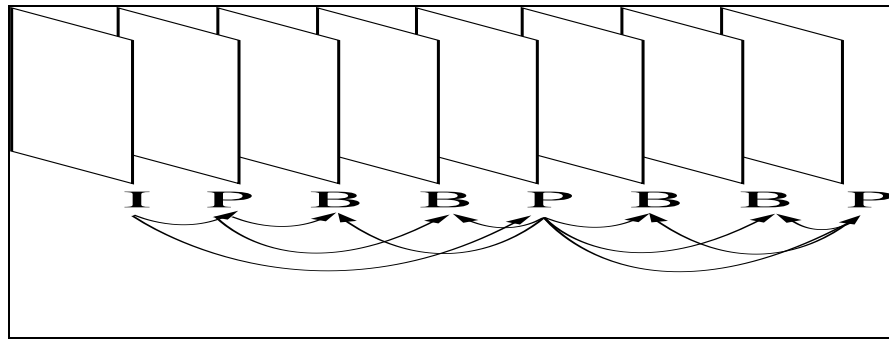


Figure 2.4 The MPEG4 frames[35].

With chunk bundling, the transmission of both I-frame data and B-frame data, for example in the same packets, is possible. Such setup is advantageous if there is a minute motion in consecutive frames that result in small (possibly 100-byte) frames.

The DCCP sender accordingly responds to each sack received. If duplicate sacks are received for a stream with enabled reliability, these sacks must be retransmitted as needed.

The bottleneck capacity is 512 Kbps and the average video stream payload - rate is 270 Kbps [36]. However, the receiver that has the initial payload start time is six - seconds after establishing the connection, thereby providing the receiver a chance to establish almost 300 kilobytes of buffered data before payload. The next figure shows the buffer size with respect to time for three distinct values of Round Trip Time (RTT) [37].

The audio and video formats standardized as MPEG-4 are particularly well suited for streaming media because of the media quality achieved at lower bitrates and have become very popular in streaming media applications [39].

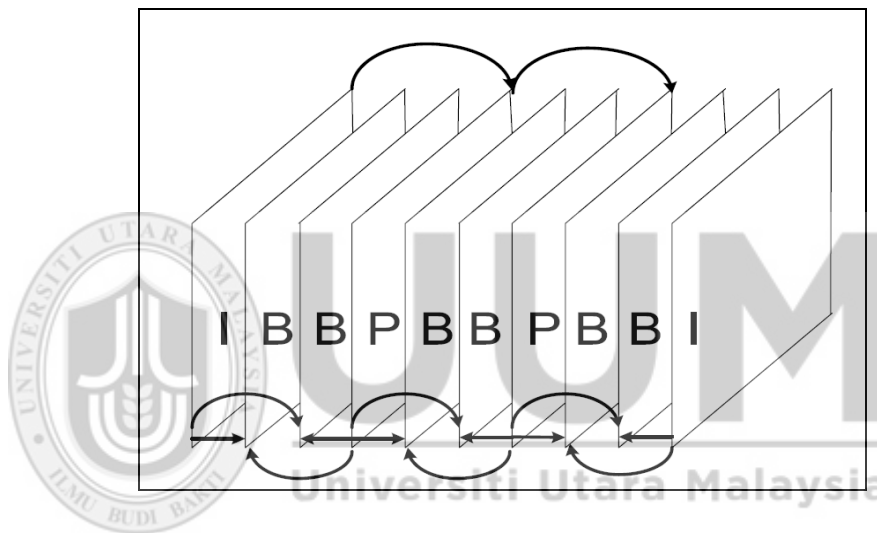
There are some parameters used to send video files under testbed. For example, Video sequence akiyo\_cif.yuv, Frame rate/type 30fps/IPP, Video codec MPEG4, and

Video bit rate 559.35 Kbps for GOP. The source file can be downloaded from this link <http://www2.tkn.tu-berlin.de/research/evalvid/cif.html>. These videos considered as standard for testing and evaluating.

The MPEG-4 video contains three different types of frames. Intra-coded frames (I-frames) contain the full image texture content while predictive-coded frames (P-frames) contain motion vectors that enable reconstruction of the frame using the previous I or P frames. The third type of frame is a bidirectional predicted frame (B-frame). These B-frames are reconstructed similarly to P-frames using motion vectors and both of the immediately adjacent I or P frames [40]. Sequences of frames, or Video Object Planes (VOPs), are divided into sets of dependent frames called GOVs (Group of VOPs). Each of these GOVs starts with an I-frame and encompasses all frames dependent on this I-frame. One such sequence would be: IPBBPBBPBBPBBPBB, as shown in the figure 2.5.

The bit rate provided by MPEG-1 and MPEG-2 is up to 1.5 Mbps and 15 Mbps, respectively, while the bit rate provided by MPEG-4 ranges between 4.8 kbps to 64 kbps. The MPEG-4 standard refers to frames as VOPs because they are not constrained to be rectangular, unlike traditional video frames. Video Object Plane (VOP) is the basic object in MPEG-4 [41]. The VOP represents a video frame with a rectangular shape. A VOP is a combination of I, P and B frames while a combination of VOP is called Group of VOP (GOV). A GOV always starts with an I-frame [42]. In P and B frames, the motion vectors for each block are simply encoded as variable length codes. Because MPEG-4 was designed to operate over loss transport layers, several error recovery and error concealment techniques were included in the standard. One of these is the insertion of a resynchronization header at periodic

intervals in the bit stream. This header includes enough information to restart the block decoding process. Optionally, additional information can be included in these headers to recover even from a corrupt voice over IP (VoIP) header. Another key error recovery feature included in MPEG-4 is reversible variable length codes. All the variable length coded information in MPEG-4 can be decoded in both the forward and reverse directions. This enables a decoder to work backwards from a resynchronization header to recover as much of the bit stream as possible.



*Figure 2.5: The dependency of I, P and B frame [43].*

The performance of MPEG-4 video has been analysed using Stream Control Transmission Protocol (SCTP) as transport layer protocol over 802.11 wireless access medium in [44]. SCTP retransmission overhead delay has been evaluated using computer simulations. The performance of WiMAX network is analysed using traffic measurements in [45]. In addition, MPEG-4 introduces two new compression methods: Harmonic Vector eXcitation Coding (HVXC) and College Level Examination Program (CLEP), for extremely low bitrate speech applications. These new methods work electively down to 2kbits/sec. As a further complication, MPEG-

4 part 10 introduces an improved video encoding method called Advanced Video Coding (AVC) [46]. This encoding scheme is identical to ITU-T Rec H.264 [47] but distinctly different from that in MPEG-4 part 2. AVC allows the decision about whether to encode data in an Intra or Predictive manner (I or P/B) to be made on a sub-frame level (the slice). In addition, AVC offers significantly improved motion compensation and adds a different blocking filter to the decoding path. Since Internet streaming was one of the planned uses of MPEG-4, part of the standard discusses the transportation of MPEG-4 over IP networks. The Real-time Transport Protocol (RTP) is the recommended solution.

Transmission Control Protocol (TCP) provides high reliability data transfer which ensures that each packet is received successfully and sequentially lead to generate significant delay, which is not suitable for real-time applications.

The most widely used protocols over transport layer are TCP and UDP. Both of these have some drawbacks during use in real time applications. TCP increases delay in transfer of data but achieves reliable transfer of data while UDP does not provide any acknowledgment, hence it overcomes the problem of delay but lacking a congestion control mechanism. In order to improve video streaming performance, UDP should be enhanced to reduce packet loss rate. The Media Access Control (MAC) support on transport layer protocols has been explicitly used in some existing reliable protocols, which employ congestion control. To provide congestion control over transport layer using UDP, DCCP has been designed to provide timely delivery of data and congestion a control mechanism.

## 2.8 Discussion on Related Work

Currently, most of the multimedia applications use UDP, TCP and DCCP as main transport layer protocols. In literature, several analysing studies deal with these protocols under various environments. However, existing studies do not encircle their performance in LTE environment, in spite of its fundamental rule in accelerating video traffic in today's telecommunications. The performance of protocols is an important part to be measured in the evaluation of any network environment. The majority of works focus on UDP, TCP and DCCP protocols under different technologies (e.g. LAN, WiMAX and Wi-Fi) and topologies. The variety of existing work also depends on the performance metrics such as throughput, delay, packet loss and packet delivery ratio.

In general, there is no integral vision in terms of protocols, metrics, technology, type of video data, number of nodes and environment that exist in literature. Nosheen et al., [18] present a comparison study to evaluate the Quality of Service (QoS) when sending MPEG-4 video considering transport protocols DCCP, UDP and SCTP over wireless local area network. They used three (3) nodes. Throughput is used as an evaluation metric. If the transfer files video is less than 4Mbps then the throughput of both SCTP and DCCP is almost 100% without packet loss, but if it becomes more than 5Mbps then the DCCP maintains its throughput and it shows minimum packet loss, and SCTP losses its performance. Hence, the DCCP protocol is better than SCTP protocol in the case of delay behaviour for the range of video traffic rates. It can be concluded from this study that both SCTP and DCCP achieved better throughput than UDP and the QoS for DCCP better satisfies than SCTP for transport

video traffic. However, other efficient protocols in transferring video such as TCP need to be included in this study. All possible metrics need to be considered and the modern environments in wireless local area network need to be intensified.

In contrast to Nosheen et al., Azad et al. [12] addresses the shortcoming of not considering all metrics by using delay, throughput, jitter and packet loss in assessing the performance of TCP, UDP and DCCP protocols. Also, they used six (6) nodes instead of three. The protocol behaviour in video applications is evaluated using MPEG-4 video coding with clear topology (see Figure 2.6) for wire network.

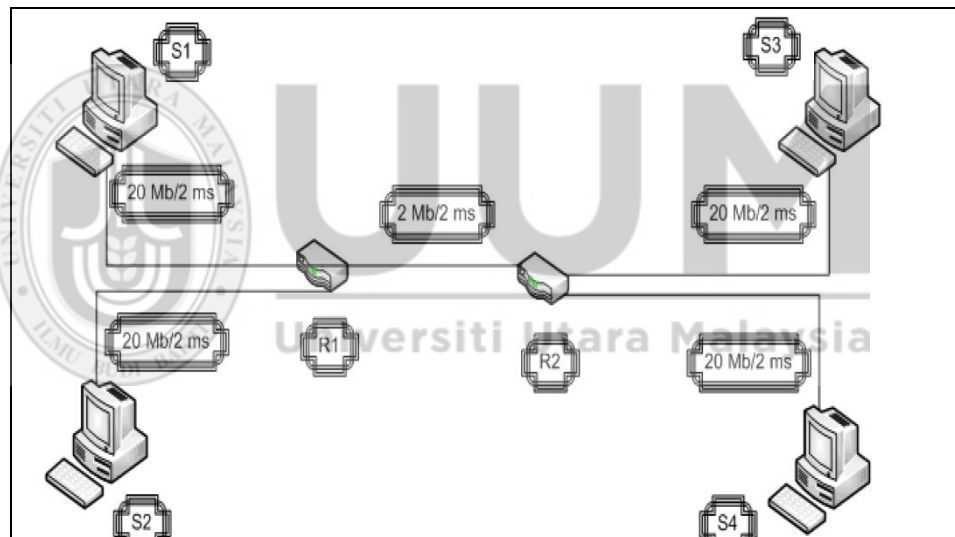


Figure 2.6: Competing traffic topology of azad et al. study [12].

The result shows that DCCP can be used as a transport layer protocol for video applications and assures superior QoS than others for transmitting video under congestion. The same environment shortcoming in the study of Nosheen et al. [18] can be concluded where wireless network is not studied.



Based on these comparison studies, the network developers are enabled to enhance protocols. In Suherman et al. [48] the Negative Acknowledgement (NACK) is used to improve UDP and medium access (MAC) layer for video streaming applications in WiMAX networks. In their suggested topology, they used four (4) nodes. The mobile node or Subscriber Station (SS) is the source of video traffics. The negative acknowledgement NACK-based protocols are not completely designed for video transmission. Therefore, they used Infrared Wireless (IR) protocol and adapt the MAC-assisted transport layer protocol through an early bandwidth request mechanism to accommodate the retransmitted packet, as in Figure 2.7.

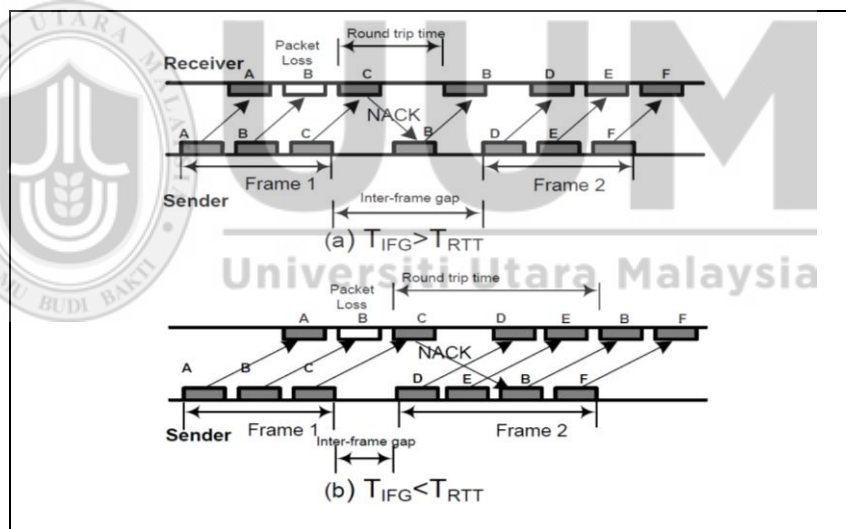


Figure 2.7: Inter-frame Retransmission (IR) Protocol [48].

In [49], the authors demonstrated Multiple TCP connections to yield significantly improved quality of video streaming over wireless channels while enhancing the reliability of video delivery compared to UDP. To reduce the delay and jitter of video streaming, the parallel TCP scheme has been used for reliable video transmission over some type of wireless channels. The authors used two nodes (2)

only. Comparing the existing scheme for video streaming over single and multiple TCP connections have been achieved. For N TCP connections, the initial window size starts with N instead of 1 window as in the case of a single TCP connection (shown in Figure 2.8).

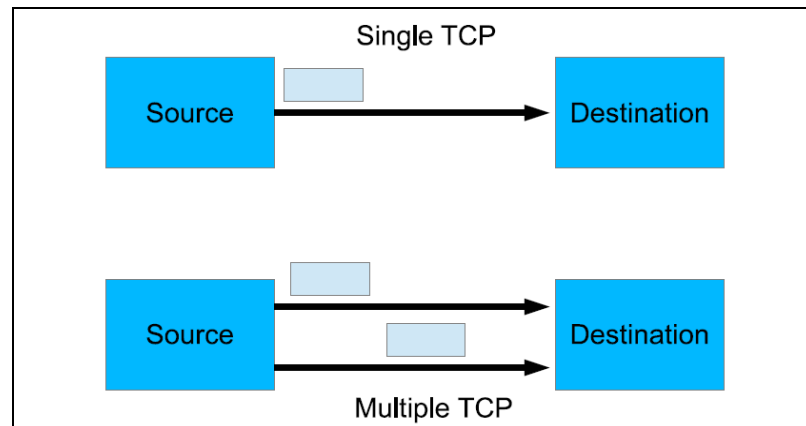


Figure 2.8: Data transfer over parallel TCP [49].

Al-Akaidi et al. [45] make a cross-layer between transport and MAC layers to increase the video quality and reduce delay time. They used WiMAX-based surveillance system. Furthermore, they used five (5) nodes in the experiments. Outdoor video surveillance applications can take advantage of a WiMAX network. They avoid two-way handshaking in the transport layer to prevent processing delays. Instead, by enabling the MAC layer to read the transport layer header in order to provide service to the transport layer. They make small change in the MAC layer and the WiMAX device is still compatible with other implementations as in Figure 2.9. Other studies focused on specific aspects while forgetting the generic aspects like protocols, metric, environment, technology and type of video data considered. Hassan et al. [50] investigated the usage of Concurrent Multipath Transfer (CMT) on SCTP protocol using Fi-Wi Networks.

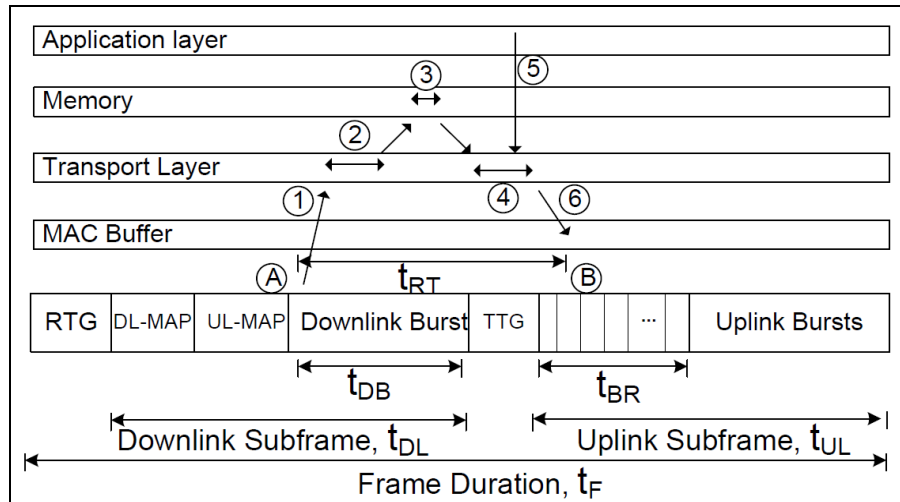
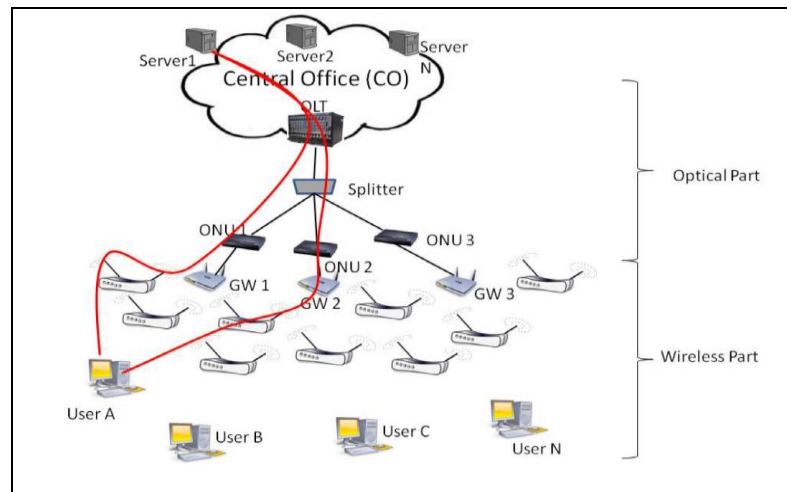


Figure 2.9: NACK and retransmitted packet flows [45].

The SCTP under Fi-Wi network provides bandwidth redundancy, and aggregation. The simulation results show a significant performance gain in terms of throughput when compared with TCP. The topology used to measure the protocol performance is mesh network, as shown in the Figure 2.10.

In [11] through simulations, performance of UDP, SCTP and DCCP protocols the transport of MPEG-4 video traffic over WiMAX as underlying access technology is analysed. More advanced analysis can be found in [12] where the behaviour of video applications over TCP, UDP and DCCP is considered. They analysed the performance of the three protocols in transferring video over transport layer under competing and non-competing topology. The MPEG-4 video codec is used in the analysis. In [19] various experiments were conducted to analyse the performance of TCP and UDP based applications in a IEEE 802.16 deployed network is focused.



*Figure 2.10: CMT setup between user node A and Server [50].*

They have used three (3) nodes in the experiments. The performance of two new transport protocols DCCP and SCTP is focused in [18]. In their study, there is an emphasis to determine which protocol can better meet the Quality of Service (QoS) for MPEG4 video over wireless local area network. Similar analysis in [13]–[15] where experimental results for DCCP compared with TCP and UDP, are presented whereas [15] uses two (2) nodes. In [16] the performance of UDP, TCP and voice in a static wireless multi-hop network experimentally and by simulation is investigated. Measurements are carried out in a network consisting of eight stations with IEEE 802.11b interfaces in a ring eight (8) nodes topology. They analysed the impact of hop count, packet size and collision avoidance mechanism on the performance metrics throughput, jitter and delay. In [17] the performance of paced and standard TCP when coexisting with DCCP over short and long delay link networks is investigated. The topology used six (6) nodes. On the other hand, in [51] one can find LTE system is investigated without highlighting transform protocols. In [52] the performance of DCCP is not analysed in LTE handover. Unfortunately, the LTE

environment does not fully consider at all in these comparisons. Recently, Singh and Hans [53] used 10 – 50 nodes as a topology for evaluating the performance of TCP, UDP and DCCP. They did not consider any multimedia application. Table 2.2 summarizes the difference between the related studies.

*Table 2.2: Comparative of evaluation studies*

Author	Protocol	Number of Nodes	Technology	Type of traffic
Ahsan Kazmi and Hassan Zaidi [50]	SCTP, TCP	6	Fi-Wi	Test Concurrent Multipath Transfer
Nosheen and Malik [18]	DCCP ,SCTP	3	Wi-Fi	sending MPEG4 video over wireless local area network
Azad and Mahmood [12]	DCCP, TCP and UDP	6	LAN	sending MPEG4 video over local area network
Chaurasia and Jagannatham [49]	TCP, UDP	2	LTE	sending video transmission over MIMO wireless channels
Al-Akaidi and Hamzaoui [45]	TCP and MAC	5	WiMAX	WiMAX-based surveillance system. Sending video over WiMAX network for security monitoring
Chowdhury et al., [15]	DCCP,TCP,CCI D2 and CCID3	2	Not mention	Sending fixed size packet because the audio/video streaming applications sent fixed size packets over deferent environment.
Hofmann et al., [16]	UDP, TCP	8	WLAN	The performance of a 3-hop testbed in a static string topology with WaveLAN adaptors.
Nor et al., [17]	TCP, DCCP	6	LAN	Investigate the performance of paced and standard (unpaced) TCP when coexist with DCCP over short and long delay link networks.
Rath et al., [19]	TCP, UDP	3	Wireless	Experiments conducted to analyze the performance of (TCP) and (UDP) based applications in a IEEE 802.16 deployed network.
Suherman, et al., [48]	UDP	4	WiMAX	send video streaming application
Singh [53]	TCP, UDP, AODV, DSR and DSDV	10-50	AdHoc Networks (MANETs)	The goal of our experiment is to examine and analyze the effect of different traffic conditions with various factors and parameters on the performance of adhoc networks.
The proposed study	TCP, UDP and DCCP	10, 20, 30	LTE	Sending MPEG-4 video over 4G LTE

The general conclusion from previous studies depicted in Table 2.3, in addition to their conflicted insights, their related parameters did not reflect the real options for qualified video delivery neither satisfied the performance evaluation in LTE large scale environment. On the other hand, the main parameters of the present study which are using MPEG-4 standard coding, unifying TCP, UDP and DCCP, considering LTE are not considered in [16] [17] [19] [53]. On the other hand, the said parameters are diversified in other studies [18] [12] [49] [45] [15] [48]. However, it is difficult to characterize the performance TCP, UDP and DCCP protocols for MPEG-4 video transmission under challenging environment, i.e. LTE. In [18] [12], the video delivery aspect has been analyzed using MPEG-4, the video standard coding, in different environments they are WiFi and LAN in addition to the difference in the selected protocols.

Although these studies built their final insights on the similar scale of number of nodes (3 for [18] and 6 for [12]) their setting is not enough to draw good indications about video transmission in challenging environments such as LTE or WiMAX. The superiority of DCCP in [18] over SCTP not UDP or TCP. This limitation has been covered by [12] because of the existence of TCP, UDP and DCCP. However, the implication of [12] cannot be projected on the quality of video delivery services of LTE; it still applicable only for LAN environments.

In terms of video transmission, the studies [49] [45] [15] [48] have solved the limitations of [18] [12] by considering LTE and WiMAX environments except the study [15] which does not mention the environment under which the simulation has been conducted. Again, it is difficult to draw a unified conclusion among these studies because in three of them one of the protocols is missing. For instance, the

DCCP in [49], UDP and DCCP in [45], TCP and DCCP in [48] and UDP in [15] are not considered. Another shortcoming is that the said studies did not determine which video standard coding they have used. Their conclusions will not be able to be generalized for MPEG-4 because of its characteristics.

## **2.9 Summary**

The video transferring over internet is growing exponentially. This chapter ends with the emphasis about the need of evaluating its performance over each protocol. Existing studies have dealt extensively with the problem in WiMAX, WiFi, WLAN, MANET and HSPA+ environments while it has concisely highlighted LTE environment in video streaming. In this regard, the literature does not converge toward the only real environment for faster data transferring, i.e. LTE. Moreover, the performance of perspective protocols in transformation layer is analysed separately without clear results or sharp conclusions. Although MPEG-4 is able to reduce the traffic it still relies on which protocol is utilized in 4G LTE channel.

## **CHAPTER THREE**

### **METHODOLOGY**

#### **3.1 Introduction**

This chapter will discuss the research methodology for the performance evaluation for transport layer protocols UDP, TCP and DCCP for MPEG4 video transformation over LTE 4G network. Also, it will explain the important steps that are commonly used by researchers in mapping the research components to each other. Several methodologies and approaches can be used to conduct research based on the performance evaluation aims. Descriptive approach is a common theme for research design in information technology [54]. It aims to seek the knowledge of nature of reality and improve the research performance [55]. The rest of the chapter is organized as follows. The general design of research is provided in Section 3.2, while Section 3.3 discusses the main steps for identifying the research problems. Section 3.4 presents the suggestion step, which is highlighted the research objectives for this study. Section 3.5 presents the first method is the development of network performance simulation while Section 3.6 presents the second method, which is analysing the performance of UDP, TCP and DCCP in transferring MPEG-4 video in LTE environment. Section 3.7 summarizes the main components of this chapter and links them to each other.

#### **3.2 Research Design Methodology**

The research framework is depicted in Figure 3.1. After the awareness of the problem is set up, the suitable methods to answer each the research questions are suggested. The main output of this research is the evaluation of the performance of



UDP, TCP and DCCP protocols in transmission video traffic in LTE environment. Many researchers used research design methodology developed by Vaishnavi & Kuechler [56] as it is accepted by many researchers in the information and communications processing system.

Evaluating the network performance in such cases is complicated. For such systems, simulation technique is appropriate. The model that simulate this complex system and the experiments that are conducted with the model are designed. Through simulated models, the two objectives are achieved: conducting scenario simulations experiments and analysing the performance of the compared protocols over wireless local area network.

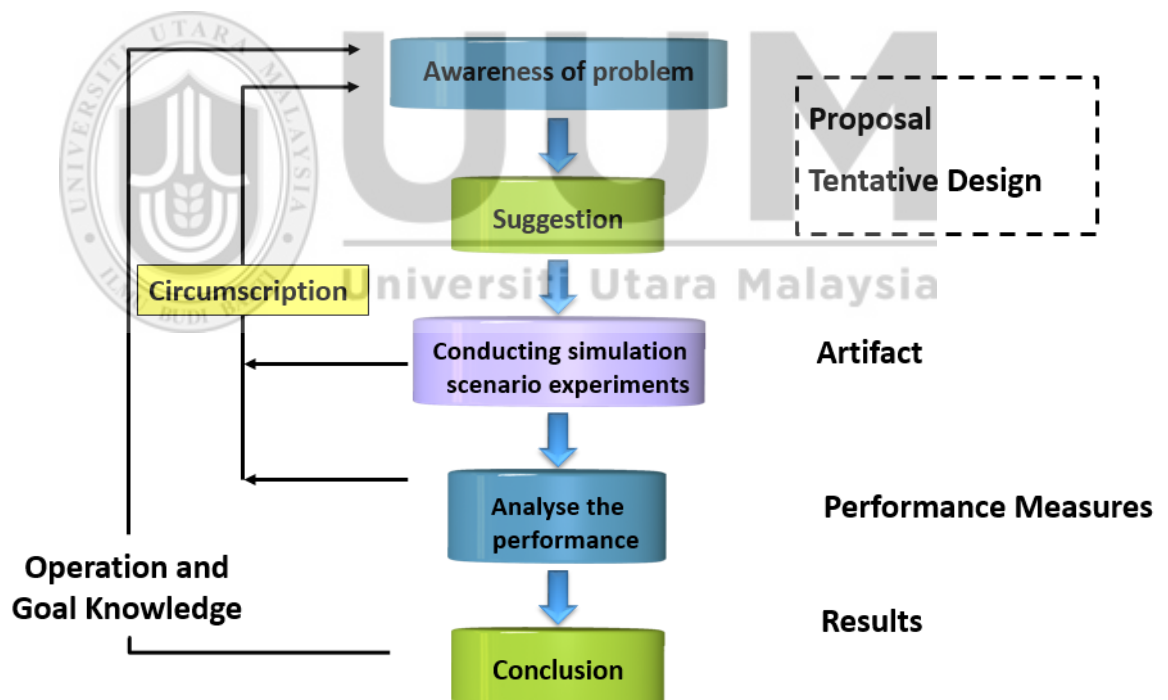


Figure 3.1: Research framework [56].

### 3.3 Awareness of the Problem

After the increase in user demand for multimedia transmission, efficiency and reliability in the LTE network is needed, so the internet developers can use any transport layer protocols (TCP, UDP or DCCP) to be sure it will reach to the user satisfactions. From this point, this research starts deep understanding of these protocols under video transmission in different environments to bonder all the characteristics that can affect the performance of video streaming in 4G network. Furthermore this step can doing by the following sub steps:

- 1- Studying the behaviour characteristics for each protocol.
- 2- Comprehensively analysing the previous performance evaluation studies and the area that were covered during the literature.
- 3- Identify the problem that lack of performance evaluation of TCP, UDP and DCCP protocols to transmission video streaming in LTE 4G networks.

### 3.4 Suggestion

After this research, addressing the problems associated with lack of performance evaluation for important transport layer protocols over 4G networks, the suggestion is to re-investigate the performance of these protocols. Conducting scenario simulation experiments is important in making fair comparison and identifying the performance metrics that can measure the different behaviours of these protocols in the same conditions and simulation platform. This suggestion (performance evaluation) can obtain by the following steps:

- 1- Conducting verity simulation scenario experiments with different number of nodes (10, 20 and 30) for UDP, TCP and DCCP transport layer protocols by using NS-3 simulator and to measure the performance of these protocols under this study via send video data streaming from LTE nodes to the one Base Station.
- 2- Analysing the performance of UDP, TCP and DCCP transport layer protocols to measurable metrics (i.e., throughput, packet delivery ration, packet loss, and delay) that are be used in the simulation scenario to impact of transport layer protocols behaviors for video transmission in LTE 4G and recommend which one is more useful for it.

#### **3.4.1 Design the Simulation Scenario**

This research suggests design simulation scenario with different number of nodes to make sure that different network behaviour is covered and considered. The all specification details for the simulation scenario design will be discuss in Chapter Four, with its pseudo codes containing more comprehensive steps. Figure 3.2 show simulation scenario design (more details in Chapter Four).

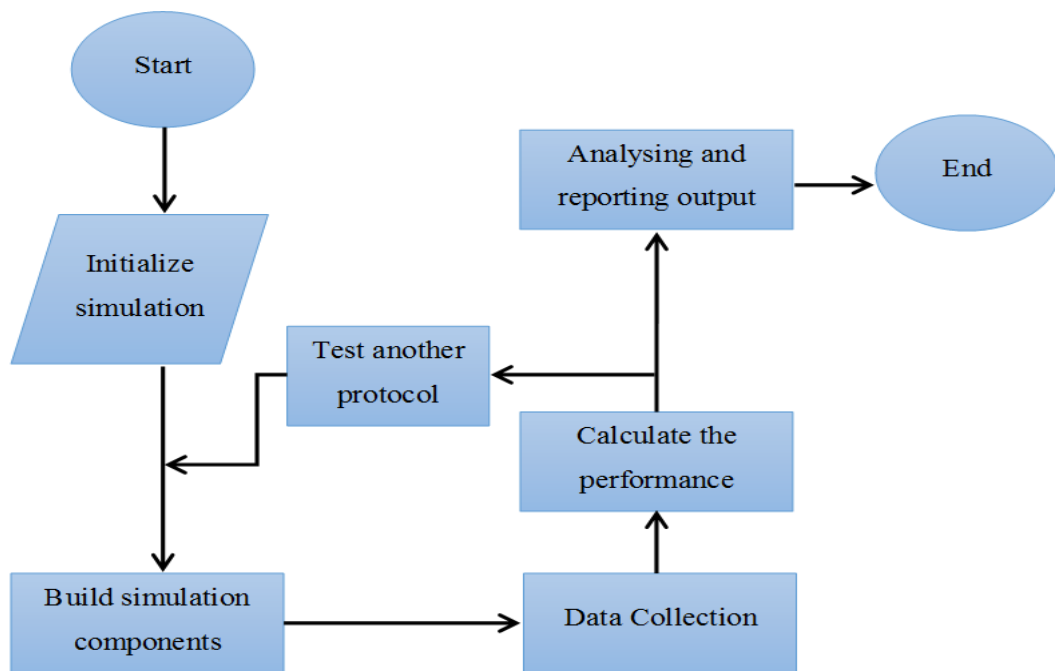


Figure 3.2: Simulation scenario design.

### 3.4.2 Implement the scenario

There are many network simulators that are used by the network researcher to implement and evaluate the simulation scenario like OPNET [57], OMNET++ [58], NS-2 [59] or NS-3 [60]. In this project, NS-3.22, which was released in February 2015, is used to implement the scenario and evaluate the transport layer protocols (TCP, UDP and DCCP) for sending video streaming in LTE 4G environments.

### 3.5 Performance Evaluation Tools

NS-3, for Internet systems, is a discrete-event network simulator targeted mainly for research and educational purposes. It is fully free software, which is licensed under the GNU General Public License, version 2 (GPLv2) license, and is easily available for usage in research and development. There is also an already existing network simulation tool NS-2 [59]. However, there are some differences, as well as better

points, than already existing NS-2. The reasons to adopt the NS-3 with differences from NS-2 are shown in Table 3.1.

*Table 3.1: Shown the strength of NS-3.*

	<b>Existing core NS-2 capability</b>	<b>NS-2 contributed code</b>	<b>Existing NS-3</b>
<b>Applications</b>	Ping vat, telnet, FTP, multicast FTP, HTTP, probabilistic and trace-driven traffic generators, web cache	NSWEB, Video traffic generator, MPEG, generator, Bonn Traffic, Portola, Agent, SIP, NSIS, NS2volp, Agent/Plant	On Off Application, asynchronous sockets API, packet sockets
<b>Transport Layer</b>	TCP (many variants), UDP, SCTP, XCP, TFRC, RAP, RTP multicast: PGM, SRM, RLM, PLM	TCP PEP, SCPS-TP SNACK, TCP Pacing, DCCP, Simulation Cradle, TCP Westwood, SIMD, TCP-RH, MFTP, OTTERS, TCP Eifel	UDP, TCP
<b>Network Layer</b>	Unicast: IP, mobile, generic dist., vector and link state, IPingIP, source routing, Nix vector	AODV+, AODV-UU, AOMDV, ns-click, ZRP, IS-IS, CDS, Dynamic, Link state, DYMO, OLSR, ATM, Ant Net, Mobile IP, GPRS, RSVP, PGM, PLM, SSM, PUMA, ActiveNetworks	Unicast: IPV4, global static routing Multicast: static routing MANET: OLSR
<b>Link Layer</b>	ARP, HDLC, GAF, MPLS, LDP, Diffserv, Queueing: Drop Tail, RED, RIO, WFQ, SRR, Sementic packet Queue, REM, priority, VQ, MACs: CSMA, 802.11b, 802.15A(WPAN), satellit Aloha	802.16, 802.11e HCCA, 802.11e EDCA, 802.11a multirate, UWB DCC-WAC, TDMA DAMA, EURANE, UMTS, GPRS, Blue Tooth, 802.11 PCF, 802.11 PSM, MPLS, WFQ schedullers, Bandwidth Broker, CSFQ, BLUE	Point To Point, CSMA, 802.11 MAC low and high and rate control algorithms
<b>Physical Layer</b>	TWO Way, Shadowing, Omni Antennas, Energy model, Satellite Repeater	ET/SNRT/BER-based phy, IR-UWB	802.11a Friis propagation loss model, log distance propagation loss model, basic wire (loss, delay)
<b>Support</b>	Unlcase: IP, MobileIP, generic dist, vector link state, IPingIP, source routing, Nixvector	Emulation, CANU mobility, BonnMotion mobility, SGB Topology Generators, NSG2, simd, NS2measure, ns-2u/akaros-2, yavista, tracegraph, huginn, multistate error model, RPI graphing pachage, jTrana, GEA	Random number generators, tracing , unit tests, logging, call backs, mobility visualizer, error models

NS-3 [61] is executed with the help of C++ with current hardware proficiencies, computational complexity is not a problem like NS-2 [61]. The main reason for this is because NS-3 can be developed with C++ completely. A simulation script can be

written as a C++ program, which is impractical in NS-2. There is partial support for Python in visualization and scripting. Because NS-3 is implemented in C++, all normal C++ memory management functionalities such as new, delete, memory allocation, and free are still available.

Automatic delete-allocation of objects is sustained with the indication of track number of points to an object. This technique is helpful if packet objects are the main concern [62]. NS-3 plays a better role than NS-2 in terms of memory organization. The accumulation method stops unnecessary parameters from being cached, and packets don't hold idle reserved header space. NS-3 retains a package, which is called PyViz (Visualizer module) and is a python-based real time package for visualization.

### **3.5.1 Network Simulator-3**

The NS-3 simulator is a discrete event simulator, which is used for research and education list. The project NS-3 was started in 2006 [61], [63]–[65] as an open source, and is not an extension of NS-2, because NS-3 is a new simulator that does not support the NS-2 API's. The NS-3 is built to provide the open-source extensive network simulation platform for research and education community. Some simulation platforms provide users with a single, integrated Graphical User Interface (GUI) environment in which all jobs are carried out. In NS-3, there are numerous external animators and tools for data analysis and visualization are also available.

Since summer of 2010, NS-3 became a very famous simulator in the methodology of network and development of the LTE [61]. However, it would provide the standard simulation of LTE devices, for example, MAC layer, Physical Layer (PHY).

Because of the complexity of the LTE for simulating the transport layer, it is better to simulate each protocol of them in different conditions. The development of the LTE module for NS-3 has innovated out during the Google Summer of Code 2010. This module offers a primary implementation of LTE units, which consists of the proliferation models, PHY and MAC layers. Because of the natural intricacy of the LTE standard and the constrained time of the Google Summer of Code (GSoC) framework [64].

On the other hand, the proposed scenario permits the recreation of a few significant parts of LTE frameworks such as downlink RRM and MAC planning. Additionally, it gives a great foundation to further augmentations. Moreover, the advancement of a complete device, i.e., the most dominant characteristic of NS-3 is given by this study strategic module, which is given below:

1. A state-of-the-symbolization Adaptive Modulation and Coding (AMC) plan for the downlink through reproducing TCP, UDP and DCCP over LTE, and sending video traffic.
2. Traffic bearers (with their QoS parameters).
3. LTE – NS-3 simulation should bear the video load balancing, by scheduling upper link, and download link for the buffering MPEG4 [61].
4. The outside 4G LTE devise model is built from C++. However, the module is manufactured totally in C++ and, at the time of this written work, includes 89 classes and gives or takes 9000 lines of code. It is critical to comment that the graph just reports the most vital information parts and capacities. A few insights about the relationship around classes have been excluded because of space constraints. Unmistakably, with a specific end goal to legitimately assess, in the

meaning of a specific remote model dependent upon the Spectrum schema, the first thing to do is to characterize a set of frequencies/channels to use at the TCP, UDP and DCCP.

### 3.5.2 NS-3 Tools

- **WAF:** The built-in system Waf is used on the NS-3 project as a compiler since the scripting of NS-3 is done in C++ or Python.
- **Node:** Node is any network device (end or intermediate device) in terms of NS-3.
- **Application:** This is represented in C++ by the predefined class Application. This class is responsible for providing a method which is used to manage the representation of our version of user-level application in simulation.
- **Channel:** The media over which data is flown in this network is called channel.
- **NetDevice:** NetDevice is installed in a node for providing the facility to a node to communicate with another node in the simulation by channel.
- **Tracing:** All network events can trace time-by-time, using this tool and big file called trace file will be created.
- **NetAnim:** There are two ways to provide animation in NS-3: the PyViz method or the NetAnim method [66]. Figure 3.3 shows the base station eNB transmit to server node.



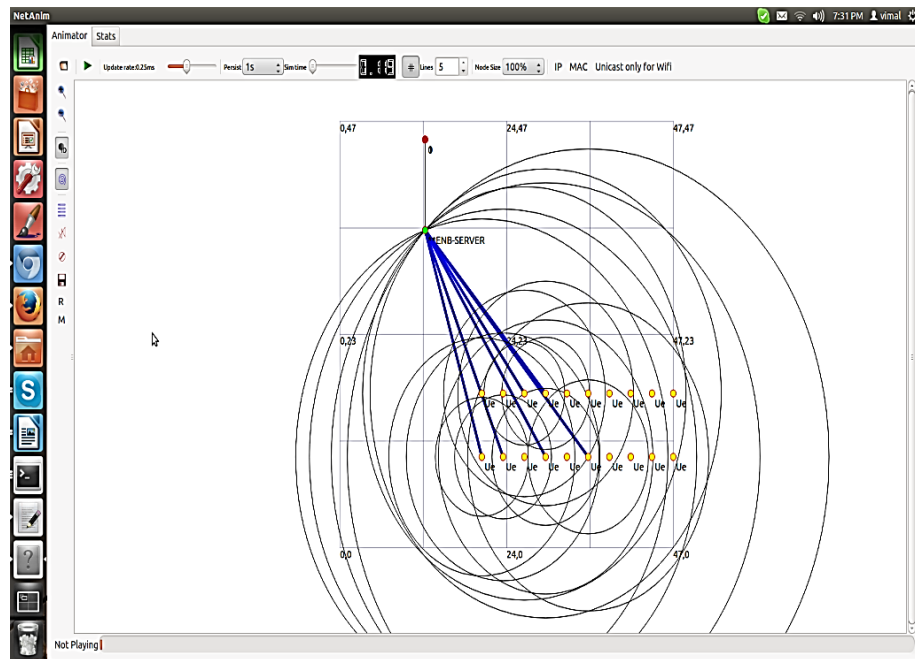


Figure 3.3: Transmission between eNB Server and UEs Node.

- **Flow Monitor:** Flow monitor is a network monitoring framework for the NS-3. It is also detected to passing flows automatically [67]. By this balanced amount of data to capture, it also minimizes the output file result size coming out from simulation like trace file, as well as Central Processing Unit (CPU) performance. In order to manage huge output file result, we use flow monitor to reduce time/memory overhead.

### 3.5.3 LTE Model in NS-3

With a specific end goal to model LTE frameworks to a level of detail that is sufficient to permit a right assessment of the aforementioned viewpoints, the accompanying prerequisites have been recognized. 4G LTE – NS-3 should scale up to several Evolved Node B (eNBs) and many User Equipment's (UEs). These guidelines out the utilization of a connection level test system. For example, a test

system whose radio interface is demonstrated with a granularity up to the image level. This is in light of the fact that to have an image level model it is important to actualize the whole PHY layer indicator transforming, whose gigantic computational unpredictability extremely constrains renovation [68]. Truth be told, connection level test systems are typically restricted to a solitary eNBs and one or a couple of UEs.

The S1-U interface is modeled in a realistic way by encapsulating data packets over GTP/UDP/IP, as done in real LTE systems. The corresponding protocol stack has been shown in Figure 3.4. As shown in the figure, there are two different layers of IP networking. The first one is the end-to-end layer, which provide end-to-end connectivity to the users; this layer involves the UEs, the P-GW and the remote host (including eventual Internet routers and hosts in between), but does not involve the eNodeB. By default, UEs are assigned a public IPv4 address in the same subnet network as the P-GW address. The P-GW address is used by all UEs as the gateway to reach the Internet.

The second layer of IP networking is the EPC local area network. This involves all eNodeB nodes and the S-GW/P-GW node. This network is implemented as a set of point-to-point links which connect each eNodeB with the S-GW/P-GW node; thus, the S-GW/P-GW has a set of point-to-point devices, each providing connectivity to a different eNodeB.

The readers are advised to refer to the Design Documentation of LENA [6] for more details about the design criteria and implementation of the LTE and EPC models in LENA.

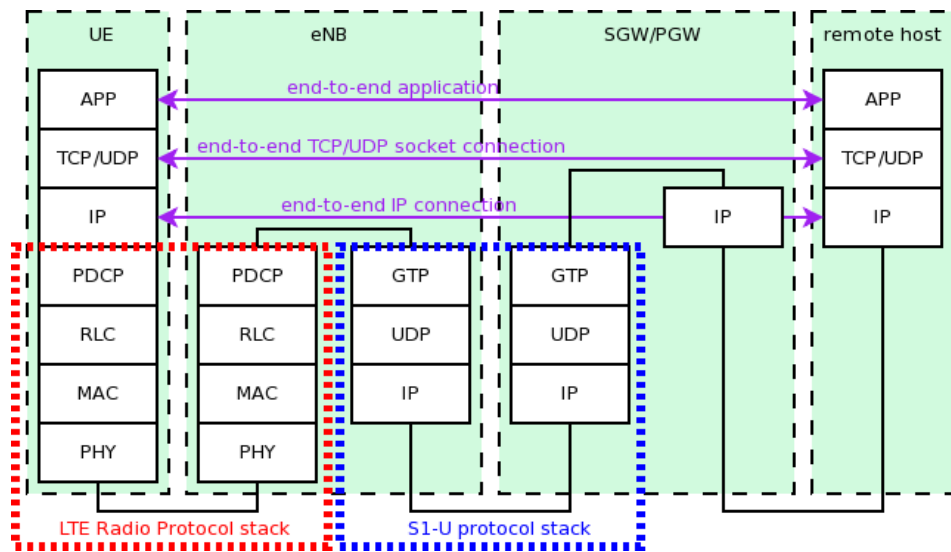


Figure 3.4: LTE data plan protocol stack [69].

In Table 3.2 all simulation setting and scenario parameters are presented to illustrate the require configuration for performance evaluation scenario. In this study the performance of TCP, UDP and DCCP transport layer protocols is carried out under LTE technology for video streaming transmission. According to these settings the number of nodes should be large in order to provide a realistic performance analyse as LTE is a large number of nodes environment. There is a great emphasis on using large number of nodes evaluations to satisfy the requirements of video delivery in LTE networks. Oyman et al.,[70] stated that “LTE networks were developed in order to deliver mobile video services to a large number of users”. Hence, this study uses 10-30 as a number of nodes. The performance analysis is carried out by varying type of protocol and number of nodes for each scenario with 125000 number of packets, 1024 byte packet size as fixed values. The protocols are ready now to be installed at the nodes upon the installation of Internet stack. The subsequence step is channel installation between nodes. A point-to-point network channel is used to ensure that

single-hope connection is existed. The NetDevice can be used to allow a host system or virtual machines to interact with a simulation.

*Table 3.2: The parameters of simulation scenario*

Parameters	First scenario	Second scenario	Third scenario
Number of Nodes	10 nodes	20 nodes	30 nodes
Number of packets	125000 Packets	125000 Packets	125000 Packets
Packet Size	1024 byte	1024 byte	1024 byte
Transport Layer Protocol	TCP, UDP and DCCP	TCP, UDP and DCCP	TCP, UDP and DCCP
Connection Channel	Point-to-Point	Point-to-Point	Point-to-Point
Net Device Type	LTE	LTE	LTE
Interval	100ms	100ms	100ms
Data rate (for channel)	100 Mb/s	100 Mb/s	100 Mb/s
Mobility Model	Constant Position Model	Constant Position Model	Constant Position Model
<b>MPEG-4 Characteristics</b>			
Frame High	252	252	252
Frame Width	288	288	288
Data Rate	65 kbps	65 kbps	65 kbps
Total bitrate	65 kbps	65 kbps	65 kbps
Frame Rate	30 frame/second	30 frame/second	30 frame/second
Video Name and Type	Highway_cif.mp4	Highway_cif.mp4	Highway_cif.mp4

The time interval duration can be controlled using the attribute during an NS-3 simulation to display the topology and animate the packet flow between nodes. The DataRate attribute specifies the number of bits per second that the device will simulate sending over the

channel. This scenario focuses on the verify connection number to make a congestion link. Therefore, the nodes movement are kept fixed because the movement will make a difficulty to illustrate the bottleneck in the base station.

### 3.6 Analyze the Performance

The evaluation step is very important to evaluate the transport layer protocols for video transmission in LTE 4G networks by using a simulation scenario described in the previous sections. The main aspects of designing the said scenario are detailed in Chapter Four, while the implementation and the performance analyse are detailed in Chapter Five. This performance evaluation will be done by the same metrics that are used by other researchers in literature [11][16][18][52][53].

1. **Throughput:** Defines the rate that something can be processed. It means in the network, the amount of effective message delivery over a communication channel, perhaps the delivery over a physical or logical link. Throughput is usually measured either bits per second (bit/s or bps), or data packets per second (p/s or pps). The performance is good when the throughput is high. The following formula is often used to calculate Throughput value, as shown in the equation (1).

$$\text{Throughput} = \frac{\text{Number of Received Packets}}{\text{Last Packet Sent Time} - \text{First Packet sent Time}} \quad (1)$$

2. **Packet loss:** For one reason or another, the packets are dropped from node. This causes unreliable delivery in the network. If a user has something which is less than the complete success in transmitting and receiving packets, then packet loss occurs. It can require much slower download and upload speeds, reduced quality VoIP audio, pauses with streaming media. Packet loss is a

metric where anything greater than 0% should cause concern. Moreover, packet loss happens in the wireless network more than the wired network because of sharing media among nodes. The performance is good when the packet loss is low. The following formula is often used to calculate packet loss value. See equation (2).

$$\text{Packet Loss} = \Sigma \text{Packets Send} - \Sigma \text{Packets Received} \quad (2)$$

3. **Packet Delivery Ratio:** It is referred to the number of packets effectively delivered to an endpoint as compared to the amount of packets that has been sent out by the sender. It means that the total number of arrived packets is divided by the total number of sender packets. See equation (3).

$$\text{PDR} = \frac{\Sigma \text{Totale Number of Received Packets}}{\Sigma \text{Totale Number of Send Packets}} \quad (3)$$

4. **Delay:** This matric is also important to check network performance. To explain how by instance, with a live audio stream, it is far more imperative to send recent packets quickly than to assure that stale packets are finally sent. Some of the protocols give high priority for packet delivery guaranty and does not care about the real time delivery. Such a network might use control protocol for congestion management, adding even more complexity, and as a consequence have more delay. Delay is the time faced by a packet to move or travel across the network from one node to another. The performance is good when the Delay is low. See the equation (4).

$$\text{Delay} = T_r - T_s \quad (4)$$

Where 'Ts' is the sending time of a particular packet and 'Tr' is receiving time of that packet. Mean delay is the average delay computed using the relation shown in equation (5).

$$\text{Mean Delay} = \frac{\text{Totale Delay}}{N} \quad (5)$$

Where 'N' is the total number of packets received during simulation time.

### 3.7 Conclusion

This step describes the final findings. Here the insights gained from analysing the three protocols, TCP, UDP, and DCCP are summarized. The performance metrics like delay, throughput, and packet delivery ratio and packet loss are the compasses that will be translated into observations. The observations are justified so that the results answer the research questions, as per our simulation results on network simulator.

### 3.8 Summary

This chapter presents all research methodology steps that were used in this research. Design Research methodology is used to complete all the requirements. The steps start with problem awareness to show the right way for research problem identification. The suggestion step presents the research objectives, then conducting experiments step presents both design and implementations sub-steps for this research and prepares the results for the next important step, which is evaluation step, by using some common performance metrics delay, throughput, packet delivery ratio and Packet loss. Finally, conclusion step is used to conclude the finding of this research.

## **CHAPTER FOUR**

### **DESIGNING SIMULATION EXPERIMENTS AND IMPLEMENTATION**

#### **4.1 Introduction**

Evaluating the performance of a network is a complicated task. Simulation is the most suitable technique for this kind of system, in which a model that simulates the system needs to be designed and implemented. This chapter presents the design for variety simulation scenario experiments to send video data streaming from LTE nodes to the one Base Station. The chapter explained how different number of nodes (10, 20 and 30) using NS-3 simulator can affect the performances of UDP, TCP and DCCP transport layer protocols. Section 4.2 presents the design of the simulation scenario, together with the logical flow of essential steps for performance evaluation scenario. The implementation of this design is explained in details in Section 4.3 before the chapter is summarized in Section 4.3.

#### **4.2 Designing Simulation Scenario**

Designing a simulation scenario is the most challenging step in network evaluation processing. This section outlines the main processes in the proposed design. These are i) initializing simulation parameters; ii) creating LTE User Equipment's nodes (UEs), Base Station node (LTE BS) and Evolved Node B (eNB); iii) setting transport layer protocol (DCCP, UDP or TCP); iv) setting nodes positions; v) setting point to point channel; vi) transmission video streaming from UEs to eNB; vii) checking if all video streaming transfer; and viii) collecting the performance data via flow monitoring tool. Figure 4.1 provides a high level view of the evaluation process.



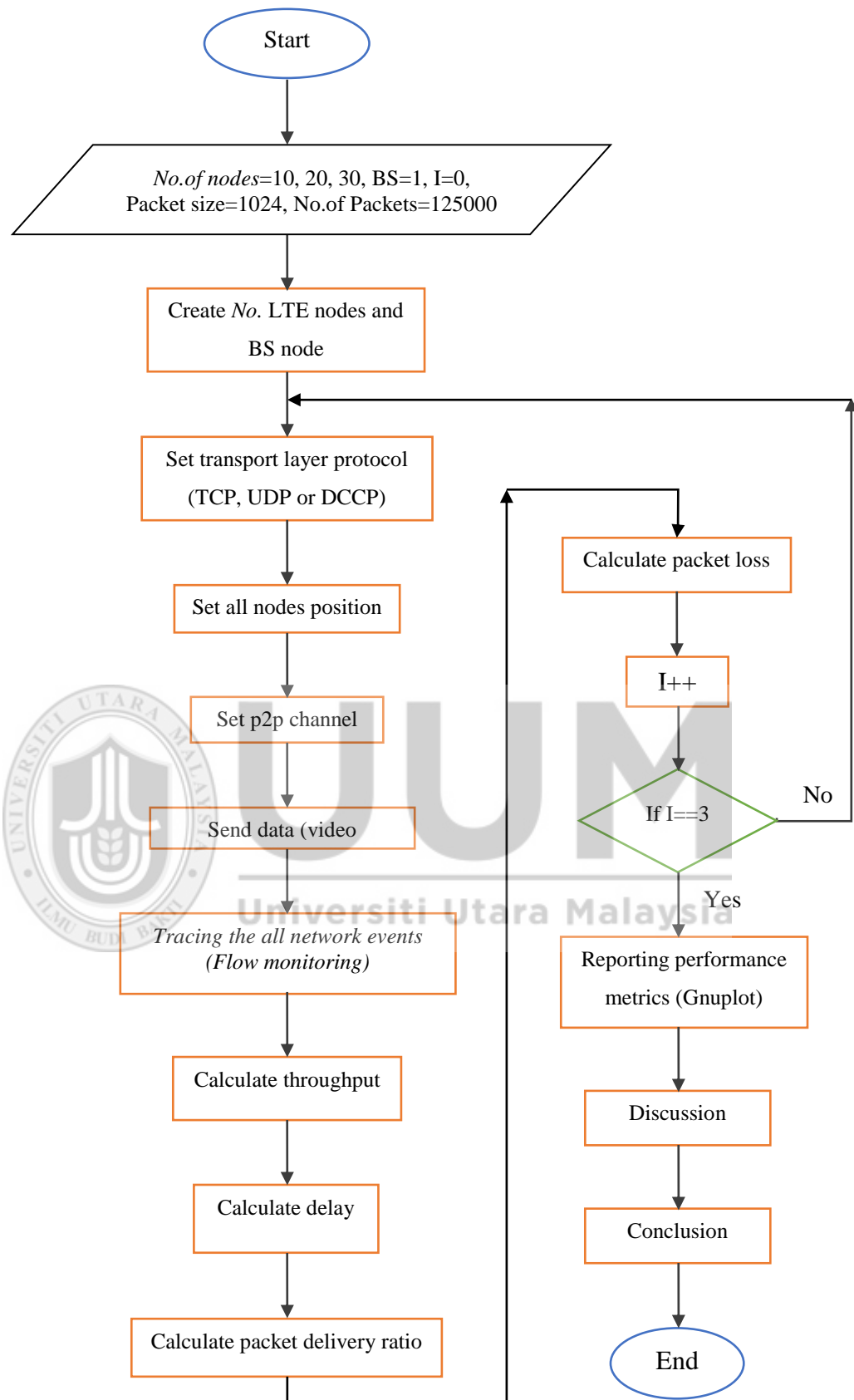


Figure 4.1: Simulation scenario flow.

Initializing simulation parameters includes setting a number of nodes, assigning the trace file of video sample, specifying packet size, choosing the transport layer protocol, selecting the suitable connection channel (point-to-point) appropriate data rate, configuring the net device type as LTE for the all nodes types, setting the channel interval and selecting mobility model. After set up types of nodes, it is important to consider their position in the network area. In the next sections, the impact of distance has been investigated. Different distances are considered in the experiments. The point-to-point channel is the most appropriate choice to connect the user nodes with the base station directly. The IPv4 is selected in the present design to assign a logical address for all nodes.

The MPEG-4 coding standard is used in application part to send video stream from user nodes to eNB node. The transfer stop condition helps in checking whether transmission is complete or not. It is highly recommended to use the time or video size as stop condition. This design uses video size as stop condition. This iterated process will produce a tracing for all network events. The network events are recorded and handled by the flow monitoring tool that is included in NS-3 simulator. It keeps record for the number of sent and received packets, the amount of sent and received bytes and the end-to-end delay, together with the simulation times. Records are accumulated iteratively to produce the total number of sent and received packets, the total amount of sent and received bytes and the overall end-to-end delay.

The number of received packets by the difference between last packet sent time and first packets sent time produces the Throughput, which has been mentioned in Chapter Three. The difference between the summations of packet sent and packet received produces the packet loss metrics. The packet delivery ratio metrics results

from the total number of packets received divided by the total of number of packets sent. Finally, the difference between the sending time and receiving time is calculated.

The mean of delay can be produced by dividing the total delay to the number of packets. After that, the performance is reported by graphical tool *Gnuplot* to show the behaviour shapes of each protocol (TCP, UDP and DCCP) in term of delay, throughput, and packet delivery ratio and packet loss with different number of nodes for MPEG-4 video stream in 4G LTE network.

### **4.3 Implementing Scenario Simulation**

In this research, two NS-3 simulator versions are adopted in performance evaluation: NS-3.17 and NS-3.10 with Direct Code Execution model (DCE). The former simulator is used to implement the performance evaluation scenario for MPEG-4 video streaming with TCP and UDP, while the latter simulator is used for DCCP performance evaluation scenario in the same 4G LTE environment. Figure 4.2 is an informal high-level description of the C++ code for all scenario components included in NS-3 simulator. The main components of this scenario are `initialize_parameters`, `create_nodes`, `create_channel`, `install_mobility`, `install_internet_stack`, `install_LTE_netDevice`, `assign_IPv4address`, `send_MPEG4videoStreaming`, `define_flow_monitoring`, `display_results` where their sequence is important for the completeness of the scenario. The following subsections provide greater details about each component.

Firstly, every scenario in the network needs to initialize its required parameters that are used in the whole scenario. Figure 4.3 describe the parameter initialization

procedure for thirty (30) UEs nodes. The number of nodes is changeable to twenty (20) or ten (10) nodes as mentioned in Chapter Three.

```

performance_evaluation ()
  initialize_parameters ()
  create_nodes ()
  create_channel ()
  install_mobility ()
  install_internet_stack ()
  install_LTE_netDevice ()
  assign_IPv4address ()
  send_MPEG4videoStreaming ()
  define_flow_monitoring ()
  display_results ()
end-performance_evaluation ()

```

Figure 4.2: The snapshot for all performance evaluation components.

```

initialize_parameters ()
  enbDist ← 250
  numberOfUEs ← 30
  numberOfEnBs ← 1
  numberofBS ← 1
  MTU ← 1500
  dataRate ← 100Mb/s
  delay ← 0.01
end-initialize_parameters

```

Figure 4.3: The snapshot for initialization parameters procedure.

The LTE nodes need to be created. The objects for each of the UEs (30), eNBs (1) and BS (1) nodes are created from NodeContainer class. Yet, the created nodes are blank, i.e. there is no network device attached. Figure 4.4 shows the process of creation.

```

create_nodes ()
  UEs ← create_UEs (numberOfUEs)
  EnBs ← create_EnBs (numberOfEnBs)
  BS ← create_BS (numberOfBS)
end-create_nodes

```

Figure 4.4: The snapshot for nodes creation procedure.

The subsequent step is channel installation between nodes. A point-to-point network channel is used to ensure that single-hop connection exists. After that,

some important settings in the channel are assigned, including MTU, dataRate and delay, as in Figure 4.5.

```

create_channel ()
  create_point_to_point (numberOfUEs)
  create_point_to_point (numberOfEnBs)
  set_point_to_point_attribute_MTU(MTU)
  set_point_to_point_attribute_dataRate (dataRate)
  set_point_to_point_attribute_delay (delay)
end- create_channel

```

Figure 4.5: The snapshot for nodes creation procedure.

This research uses ConstantPositionMobilityModel class to identify the mobility model for all nodes with fix position for each node. Therefore, the position (10, 40, 0) is allocated for the eNBs node while UEs position is started from x= 250 and it is increased by three (3) for each node with the same eNBs mobility model (see Figure 4.6).

```

install_mobility ()
  y←17
  create_mobility1 (EnBs)
  create_mobility1.set_position_allocated (10, 40, 0)
  create_mobility2 (UEs)
  for (i=1 to numberUEs)
    create_mobility2.set_position_allocated (enbDist, y, 0)
    enbDist← enbDist+3
    if (i = 10)
      enbDist←250
      y←10
    end_if
  end_for
  create_mobility1.set_mobility_model (constantPositionMobilityModel)
  create_mobility2.set_mobility_model (constantPositionMobilityModel)
end- install_mobility

```

Figure 4.6: The snapshot for mobility installation procedure.

The protocols are ready now to be installed at the nodes upon the installation of internet stack. Therefore, the procedure showed in Figure 4.7, i.e. `install_internet_stack`, is responsible for completing this task for all nodes.

```
install_internet_stack ()
    create_internet_stack (UEs)
    create_internet_stack (EnBs)
    create_internet_stack (BS)
end- install_internet_stack
```

Figure 4.7: The snapshot for internet stack installation procedure.

All the nodes are prepared to be LTE nodes. This can be done by Installing LTE netDevice procedure using `LteHelper` class that is provided by NS-3 simulator. Hence, the LTE nodes become ready to receive the IPs. Figure 4.8 describes this implementation.

```
install_LTE_netDevice ()
    LTE_netDevice_UEs ← create_LTE_netDevice_UEs (UEs)
    LTE_netDevice_EnBs ← create_LTE_netDevice_EnBs (EnBs)
    LTE_netDevice_BS ← create_LTE_netDevice_BS (BS)
end- install_LTE_netDevice
```

Figure 4.8: The snapshot for LTE netdevice installation procedure.

The logical IP address for each node needs to be assigned. The `IPv4AddressHelper` class is used to create IPv4 object then `IPv4InterfaceContainer` class is required to assign the object to each in the network. The IP assignment process is shown in Figure 4.9.

The transport layer protocols will be set as TCP, UDP or DCCP before the video streaming start. After this step, the video streaming sample file is ready to be transferred from each UEs node to eNBs node. However, it cannot transfer real video

data from node to another in NS-3 unless the evalvid model is installed and configured within the simulator.

```

assign_IPv4address ()
    interface_UEs ← create_interface_UEs (LTE_netDevice_UEs)
    assign_IPv4 (interface_UEs)
    interface_EnBs ← create_interface_EnBs (LTE_netDevice_EnBs)
    assign_IPv4 (interface_EnBs)
    interface_BS ← create_interface_BS (LTE_netDevice_BS)
    assign_IPv4 (interface_BS)
end- IPv4address

```

Figure 4.9: The snapshot for IPv4 address assignment procedure.

A loop for sending packets from source to destination is repeated until all video data is transferred. This operation occurs from each node to send the same video file size in the same time. Figure 4.10 illustrates the procedure of sending MPEG-4 video streaming.

```

send_MPEG4videoStreaming ()
    set_transport_layer_protocol () /*the TCP, UDP or DCCP are set in this step*/
    install_evalvid() /*this model to transfer real MPEG-4 video (trace file) between
nodes*/
    for(i = 1 to numberOfUEs)
        send_from_UEs_to_eNBs(UEs[i], videoTrfaceFileName)
        /* videoTrfaceFileName =
        "highway.st" where
        highway.st is the trace file
        highway.mpeg4 video
        sample*/
    for
        send.start_second (9)
        send.stop_second (90)
    end_for
end- send_MPEG4videoStreaming

```

Figure 4.10: The snapshot for MPEG-4 video streaming procedure.

Flow monitoring is considered a main tool in NS-3 to monitor all network events that are obtained by defining flow object from FlowMonitorHelper class to be installed in

all nodes in the network. The flow monitoring keeps record for seven events, including txPacketSum, rxPacketSum, txBytesSum, rxBytesSum, lostPacketSum, dropPacketSum and delaySum. These records will be used to calculate the performance metrics while the Gnuplot object handles the dataset for each performance metric, to be displayed later using display\_results procedure. In display\_results procedure, all dataset will be assigned to files with the extension “.plt”. After that, Gnuplot program draws these files and creates pictures with the extension “.png”. Finally, Gnuplot program will display these pictures for all metrics. Figures 4.11 and 4.12 show greater details regarding flow monitoring and results’ display, respectively.

```

define_flow_monitoring ()
  create_flow_monitoring.installAll ()
  /* start define throughput graph properties */
  fileName ← define_fileName(throughput) /* delay, packet delivery ratio or packet
loss*/
  graphName ← define_graphName(fileName.png)
  plotFile ← define_plotFile(fileName.plt)
  Gnuplot ← create_Gnuplot(graphName)
  Gnuplot.setTitle(plotFile)
  Gnuplot.setLegend ("In kbps", "throughput")
  /*Gnuplot.setLegend ("node communication", "delay"),
  Gnuplot.setLegend ("PDR value%", "nodes") or
  Gnuplot.setLegend ("packet loss", "simulation
time") */
  Gnuplot.appendExtra(set_x_range[0:+25])
  Gnuplot.appendExtra(set_y_range[100:+8000])
  /* [0.0: +0.2], [90: +110] or [0: +200] */
  throughputDataSet ← create_throughputDataSet ()
  delayDataSet ← create_delayDataSet()
  PDRDataSet ← create_PDRDataSet ()
  packetLossDataSet ← create_packetLossDataSet ()
  for (i = firstEvent to lastEvent )
    txPacketSumt+1 ← txPacketSumt + txPacket
    rxPacketSumt+1 ← rxPacketSumt + rxPacket
    txBytesSumt+1 ← txBytesSumt + txByte
    rxBytesSumt+1 ← rxBytesSumt + rxByte
    lostPacketSumt+1 ← lostPacketSumt + lostPacket
    dropPacketSumt+1 ← dropPacketSumt + dropPacket
    delaySumt+1 ← delaySumt + delay
    throughputDataSet.add (i, (txBytesSum * 8)/1024)
    delayDataSet.add (i, (delaySum / txPacketSum))
    PDRDataSet.add (i, (rxPacketSum * 100 / rxPacketSum))

```



```

                                lossPacketDataSet.add (i, (lostPacketSum / 10))
                                end_for
                                end- define_flow_monitoring

```

Figure 4.11: The snapshot for flow monitoring procedure.

```

display_results ()
    throughput.plt.addDataSet (throughputDataSet)
    delay.plt.addDataSet (delayDataSet)
    packetDeliveryRatio.plt.addDataSet (packetDeliveryRatioDataSet)
    packetLoss.plt.addDataSet (packetLossDataSet)
    draw (throughput.plt)
    draw (delay.plt)
    draw (packetDeliveryRatio.plt)
    draw (packetLoss.plt)
    display (throughput.png)
    display (delay.png)
    display (packetDeliveryRatio.png)
    display (packetLoss.png)
end- display_results

```

Figure 4.12: The snapshot for results display procedure.

In addition, NetAnim is a virtual tool to show the mobility of nodes and packets transmission in NS-3. Figure 4.13 depicts the functionality of NetAnim for transmission video packets from all nodes to the eNBs node.

#### 4.4 Summary

The experimental design for the simulation scenario for performance evaluation and its implementation has been done. Each part includes fundamental operations that are governed by a logical flow, which is the main contribution to this chapter. This chapter walks-through initializing parameters, creating nodes, setting point- to-point channel, installing LTE net device for each node and internet stack, assigning IPs for all nodes, choosing transport layer protocols (TCP, UDP or DCCP), sending real MPEG-4 video streaming using evalvid model, monitoring the network events by

flow monitoring tool and getting the results through its iterations to be drawn and displayed using Gnuplot program. Overall, this chapter proposes a detailed guideline for the network developers who are interested in studying the video traffic performance in LTE environment, focusing on TCP, UDP and DCCP protocol.



## **CHAPTER FIVE**

### **PERFORMANCE ANALYSIS**

#### **5.1 Introduction**

After describing the performance evaluation tools (NS-3 simulator, NS-3 tools and models and LTE models in NS-3) in Chapter Three and their design in Chapter Four, this chapter presents the results of evaluating TCP, UDP, and DCCP protocols. The performance analyze process builds on the base of initializing simulation, building simulation components, data collection and calculating performance metrics. The results then reported using the graphical tool Gnuplot in NS-3. The graphical analysis of the protocol performance metrics like delay, throughput, Packet Delivery Ratio and packet loss are given in Section 5.2 and summarized in Section 5.3.

#### **5.2 Result Analysis of UDP/TCP/DCCP**

This section shows the network performance by measuring the throughput, delay, packet loss, and packet delivery ratio for three different scenarios. We have shown previously that the network topology consists of three parts. The mobile unit call (UEs) which is communicating directly with base station, the base transceiver station (BTS) also calls Evolved Node B, (abbreviated as eNodeB or eNB), and the end terminal which is server in our scenario. This server receives the packets from mobile units. In order to measure network performance we have created three different scenarios 10 UEs, 20 UEs, and 30UEs connect directly to one eNB and the eNB connected to server node.

The topology setting is another important aspect in performance evaluation. Several researchers used 10, 20 and 30 nodes as a setting for the number of nodes to study

the effect of different network traffic with transport layer protocols [53]. This research used the same rationale. The rest of this section is organized based on performance evaluation metrics. The following subsections are discussing the results of throughput, end-to-end delay, packet delivery ratio, and packet loss. The results of network performance for UDP, TCP and DCCP protocols with different scenarios are presented. It is worth mentioning that the x-axis, i.e. the simulation time, relies on number of nodes to be transmitted and the size of video transmitted. For example, in throughput experiments for 10 nodes the x-axis scale ends at the 12<sup>th</sup> second while for 20 nodes it ends at 25<sup>th</sup> second.

### **5.2.1 Comparison Analysis for Throughput**

This subsection investigates the comparative performance of the three protocols over LTE systems by using the throughput metric. The throughput in the network refers to the rate of successful message delivery over a communication channel. It perhaps refers to the delivery over a physical or logical link. The measurement unit of throughput is usually either bits per second (denoted by bit/s or bps), or data packets per second (denoted by p/s or pps). In general, this metric play essential role in evaluating the performance of protocols in the new wireless communication systems.

Figure 5.1 showed that the DCCP protocol has the best throughput in the environment of the LTE network of 10 nodes. The scenario here supposes all the ten nodes are sending MPEG-4 video file at a same time to the eNB node (as explained in the Chapter 4).

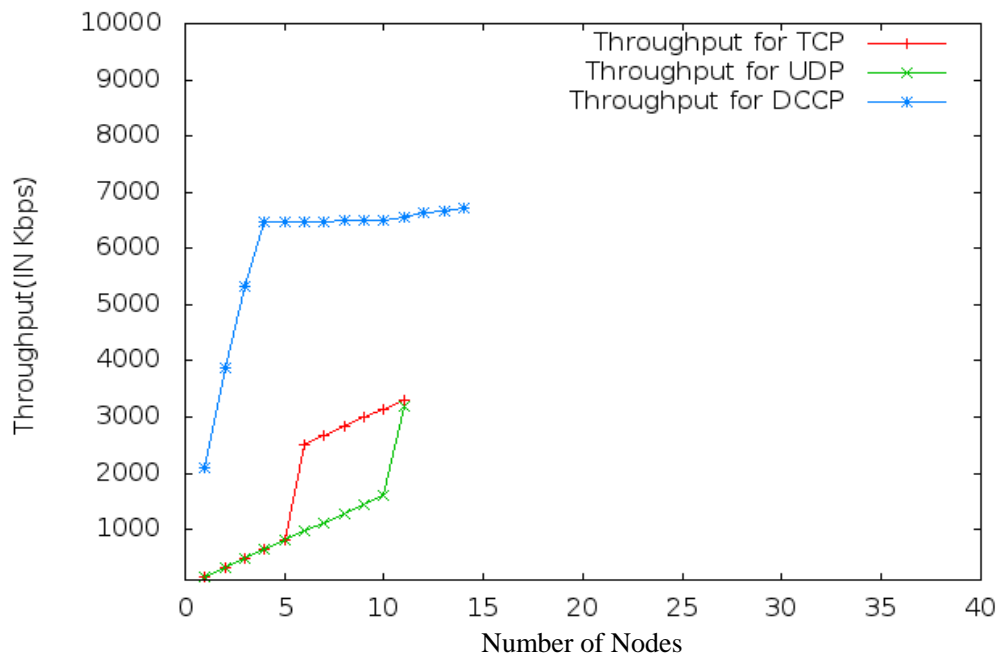


Figure 5.1: The results of comparing the throughput of TCP/UDP/DCCP protocols for 10 nodes.

Figure 5.2 shows that the DCCP protocol still has the best throughput in the environment of the LTE network if the number of nodes is increased to 20 nodes. All these results are taken from NS-3 simulator, which is already valid.

Figure 5.3 shows the continuity of DCCP protocol in its best throughput in the environment of the LTE network if the number of nodes is increased to 30 nodes. Results confirm the stability of this protocol against the increasing in the number of video file sending by the 30 nodes with unexpected behavior of other protocols.

Furthermore, as the number of nodes increases, the throughput of complete network will be improved. The consistent growth of graph shows that the network is capable to handle all these nodes number. To get the peak performance, there is no bottleneck up to this limit of node numbers. The value of throughput is given in kbps. As the nodes increase, the throughput grows higher. From the total throughput

result, as the number of nodes increases, the throughput doubles, which is approximately from 2805.15 Kbps to 5593.25 Kbps.

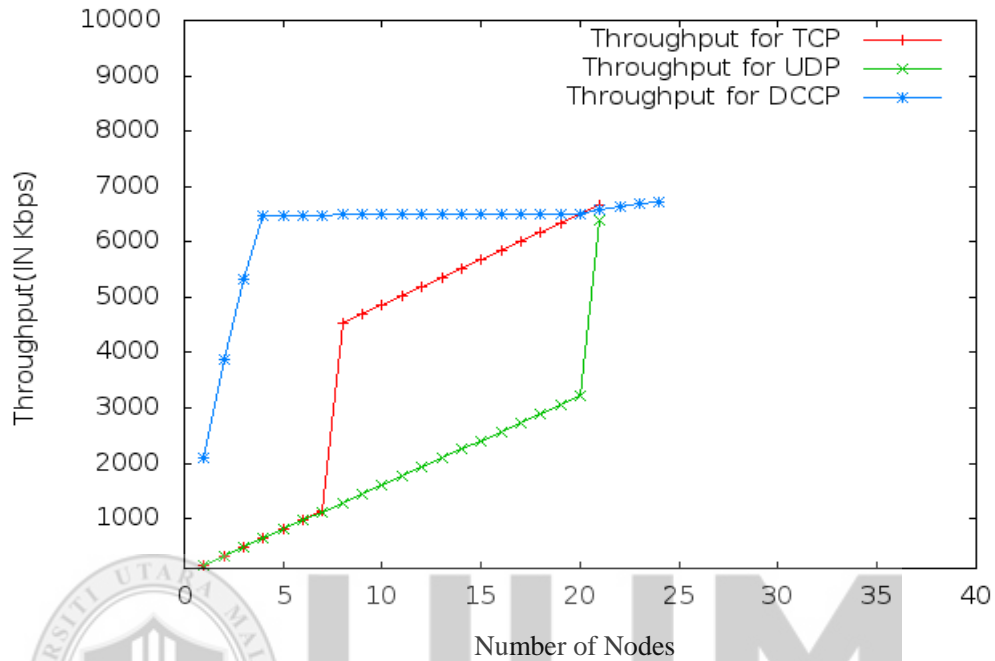


Figure 5.2: The results of comparing the throughput of TCP/UDP/DCCP protocol for 20 nodes.

The DCCP protocol has the best throughput because it uses congestion-controlled schemes with Explicit Congestion Notification. DCCP provides with two diverse congestion control techniques containing TCP-Like and TCP friendly rate control. Also it provides less delay and supports delay-sensitive streaming over UDP without TCP's delay inducing reliability. In contrast, the TCP protocol is suitable for wire connection and not designed to work in the wireless environment. Therefore the TCP's disadvantage protocol has been overcome by DCCP which is designed for wireless environment.

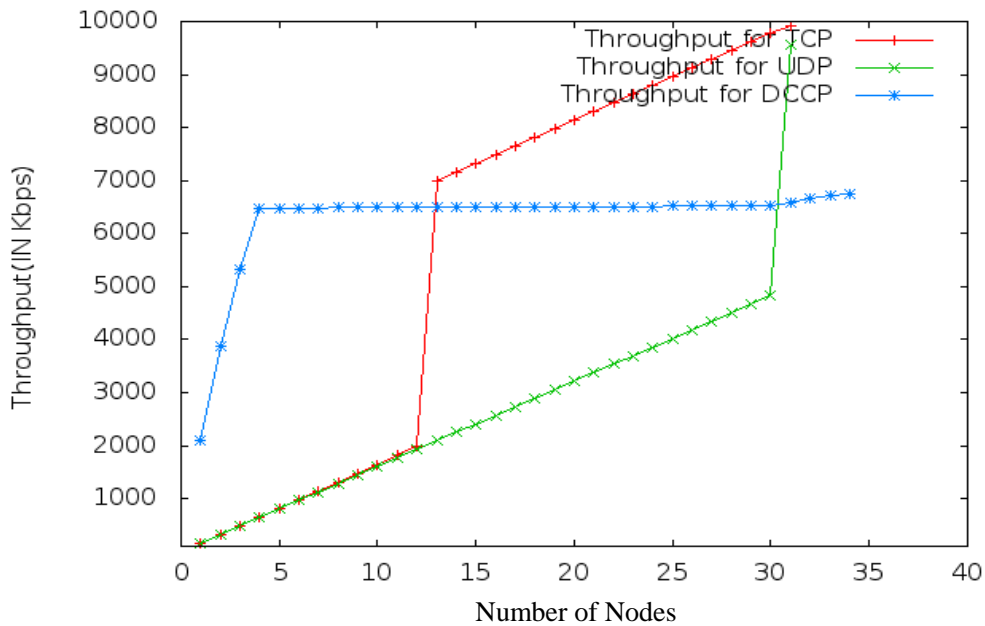


Figure 5.3: The results of comparing the throughput of TCP/UDP/DCCP protocol for 30 nodes.

The DCCP protocol has good throughput than TCP and UDP protocols because it has two mechanisms employed inside it, which are: congestion control (controlling the packets sent to the network when it became greater than available network capacity) and flow control (controlling the traffic size when the sender being sent up to the limit it receives response from the receiver). It can be seen that DCCP has congestion control mechanism that point a threshold for window size growing as exemplified in time (4) in Figures 5.1-5.3 where the throughput quickly increased from 2000 Kbps to 6500 Kbps. Therefore, the congestion mechanism will force the protocol to send packets in steady state phase. The same behaviour can be found in TCP except that TCP try to investigate a new threshold by increase the window size slowly until the time (12) where the window size increasing until reach the new threshold by using slow start mechanism. As a result from this mechanism, the throughput will jump from 2000 Kbps to 7000 Kbps (no congestion occurs). After

that, the avoidance mechanism increases window size a bit from time to time to investigate the new threshold and so on. The same concern goes to UDP except that in UDP there is no congestion control (there is no acknowledgement indication).

In general, the throughput is high as the LTE environment is characterized with high bandwidth where the implemented bandwidth in Chapter Four is 100 MB/s. The impact of LTE bandwidth even dominated the increment in number of nodes: the throughput increases when number of nodes are increased. Hence, the DCCP behaves better than others due to the fact that DCCP is congestion protocol not flow control protocol. This enable it to send the data in the same frequency. Further, the packet header of DCCP can be used to carryout 1024 B in each transmission which can effect link utilization.

### **5.2.2 Comparison Analysis of Delay**

Delay is one of the important metrics to check network performance. Some of protocols give high priority for packet delivery guarantee. And do not care about the real time delivery, such as TCP protocol. In the end the congestion management, adding even more complexity, as a consequence gives more delay. So for that reason the TCP protocol has long delay time. As shown in the Figure 5.4.

The DCCP protocol has the best result because the delay time is less than the other protocols. This result is for 10 nodes. Again, all these are node sending file stream video at a same time to reach the server. The server must be behind the eNB.

The Figures 5.4 and 5.5 show the results for 10 nodes and 20 nodes respectively. We see the results are the same except in the beginning of figures for UDP and TCP protocols. The small difference is that the TCP protocol needs at the beginning more



time to establish the connection. Also, this establishment of connection affects the number of nodes. To be fair, the DCCP protocol also has best result with 20 node scenario.

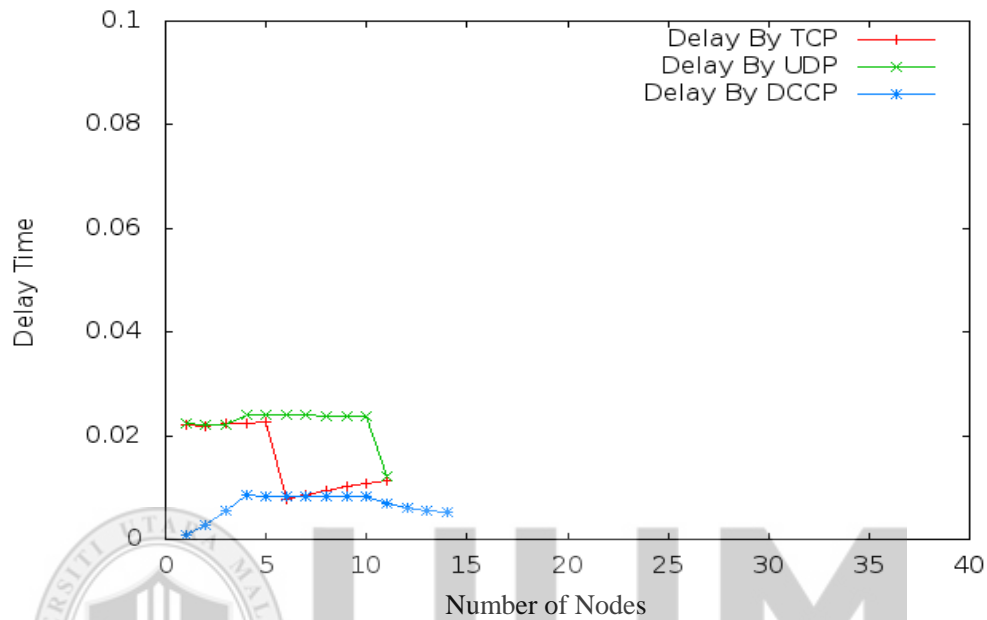


Figure 5.4: The results of comparing the average delay of TCP/UDP/DCCP protocol for 10 nodes.

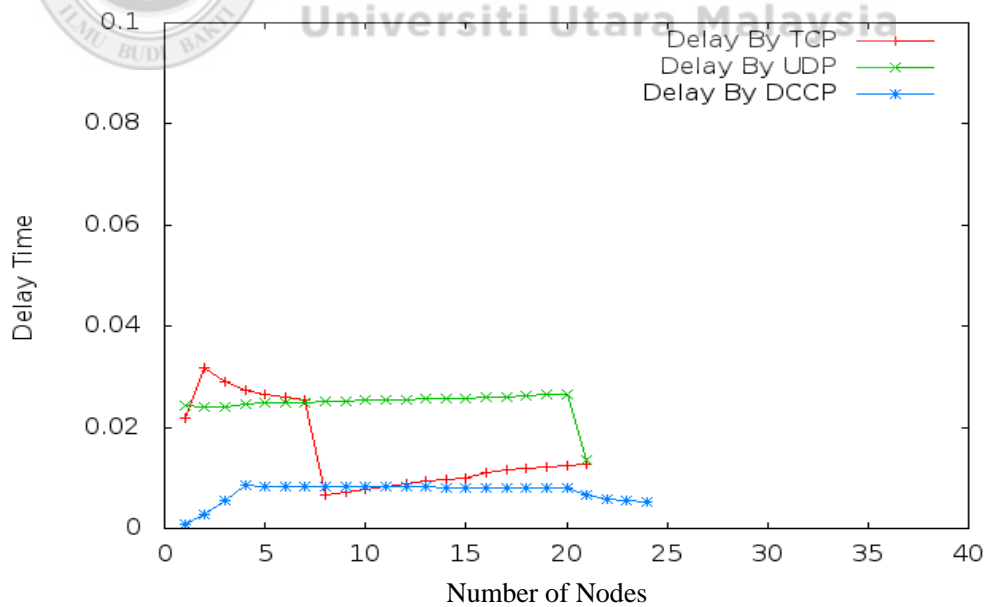


Figure 5.5: The results of comparing the average delay of TCP/UDP/DCCP protocol for 20 nodes.

Figure 5.6 shows the results for the 30 node scenario. Because the number of nodes increases definitely the time delay also increase. This increase happens more in the wireless than wire because the layer two in the wireless needs acknowledgement (ACK) the RTS/CTS as well as layer three (ACK). Besides, wireless network uses media share not like wire. Compared to all three scenarios for Average Delay time for TCP, UDP, DCCP, the UDP protocol shows consistently higher delay due to less flow of the data over the network. In TCP, first the delay is more during the connection establishment phase, but once the connection has been established, TCP increases its window size delay drops sharply in the data flow as shown in the diagram. And DCCP outperforms these both conventional connection-less and connection-oriented protocols in case of delay. Comparative Analysis of TCP / UCP / DCCP protocols for 30 node scenario shows the DCCP protocol is the best protocol regarding to delay time.

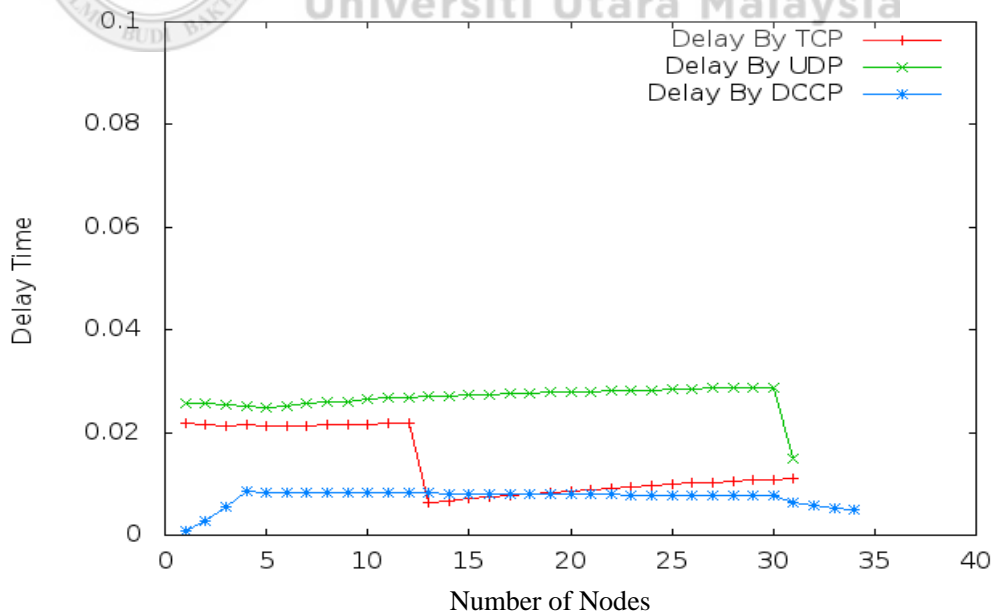


Figure 5.6: The results of comparing the average delay of TCP/UDP/DCCP protocol for 30 nodes.

The DCCP protocol has better results than TCP and UDP because the delay time is less than the other protocols. TCP protocol needs at the beginning more time to establish the connection. Also, this establishment of connection affects the number of nodes. Because of when the number of nodes is increased definitely the time delay also increases. Compared to all three scenarios for Average Delay time for TCP, UDP, DCCP, the UDP protocol shows consistently more delay due to connecting less flow of the data over the network. For TCP, in the first period the TCP has a specific window size with specific delay then the TCP behaviour needs to investigate new threshold from time to time. After the new threshold established, the window size increased immediately to reach the new threshold. This caused decrease the delay in times 5, 7 and 13 of Figures 5.4, 5.5 and 5.6 respectively. While the DCCP use slow start mechanism to increase the number of sent packets. This causes increasing the delay in the first time of DCCP behaviour. After that, DCCP threshold induces the stability in sending packets therefore the delay will be in steady state the rest time. Since the UDP try to send as much as possible according to the limit of transmission rate (because the LTE has high bandwidth). Hence, the UDP delay seems in steady condition most of time.

After examining the different scenarios, it can be seen that the load of TCP is proportional to the number of nodes: the increased in number of node will cause a high load. Beside, sending large amount of data (MPEG-4) comparing to the amount of available bandwidth (link capacity) is another property. Since the MPEG-4 is characterized with a high frame rate (30 frame/sec) as implemented in Chapter Four, it results a delay to transmit the video. Moreover, the TCP protocol retransmitted the

dropped packets which will cause traffic delay. These two kinds of delay are justified the shape of TCP behaviour.

While in DCCP protocol, although there are no packet level retransmission DCCP will not suffer from this kind of delay. The only delay that can be noticed is the delay that caused by MPEG-4 frame rate. Therefore, the behaviour of DCCP is better than TCP.

For UDP, it has high delay because of the network characteristic and the network traffic. Considering the scenario in which the network is LTE, UDP does not have congestion any avoidance or congestion control mechanism and eNB using UDP sends the data at the same rate continuously without using the available bandwidth. For this end, DCCP is the best protocol among all in terms of delay.

### **5.2.3 Comparison analysis Ratio for Packet Delivery Ratio**

It refers to the amount of packet, effectively sent to a receiver compared to the amount of packets that have been delivered by the transmitter, means the total number of arrived packets divided by the total number of sent packets. Packet Delivery Ratio for TCP socket is varies; minimum 94 % to 99% approximate which is quite good result for any Network. The packet delivery ratio is the rate of packets arrived at the receiver node in comparison to the total number of packets sent from the sender node. The Packet Delivery Ratio is maximizing up to 99% showing that the network performance is good quality.

The result shows the number of loss packets is only (4 packets) and its loss ratio is only 1%. So the lost ratio between Ue node & eNB base station is low. Packet Delivery Ratio for TCP socket varies, i.e., minimum 94 % to 99% approximately,

which is quite good result for any network. The TCP protocol uses (ACK) while establishes the connection that is why it has the best Packet Delivery Ratio.

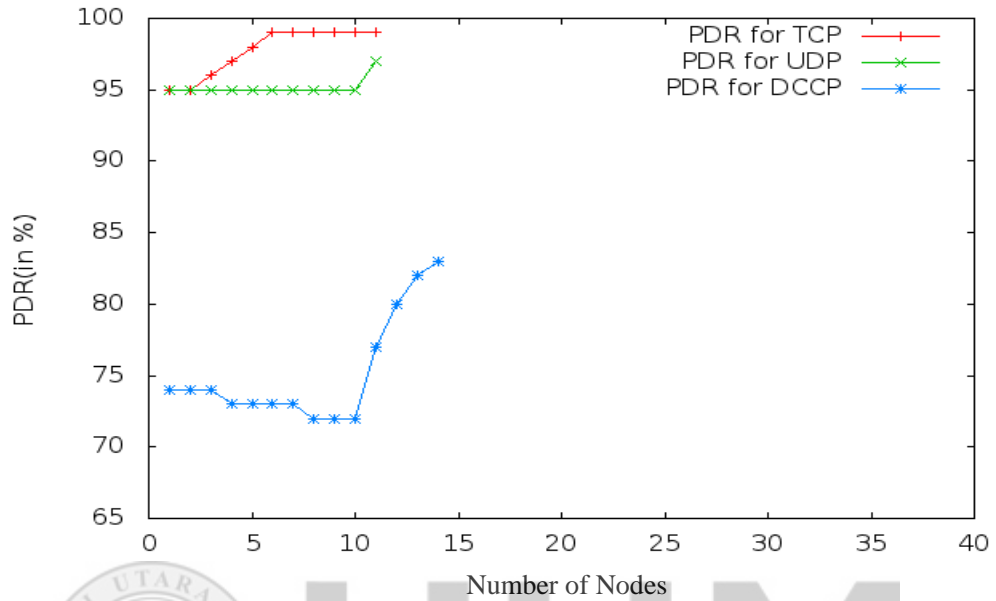


Figure 5.7: The results of comparing the PDR of TCP/UDP/DCCP protocol for 10 nodes.

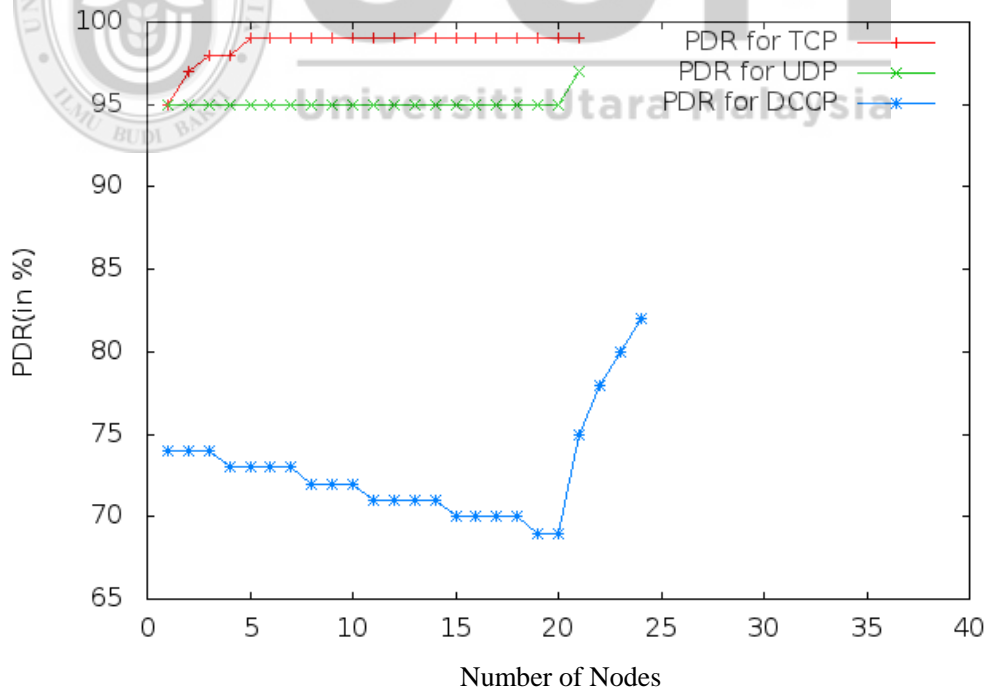


Figure 5.8: The results of comparing the PDR of TCP/UDP/DCCP protocol for 20 nodes.

Also, the result shows the UDP protocol has the best Packet Delivery Ratio if the number of nodes is 10, as shown in Figure 5.7. The DCCP protocol is the worse if we measure Packet Delivery Ratio, i.e., it is about 75%. Therefore we must improve (i.e., minimize) the packet loss for this protocol in the future work. This result would be different if we remove the constraint. This leads us to make the component of hardware which will have a big memory buffer to overcome the packet loss. And nowadays memory is available in terabytes, so it is not an issue at all. The Figures 5.8 and 5.9 do not have much difference from the Figure 5.7, which is already discussed above.

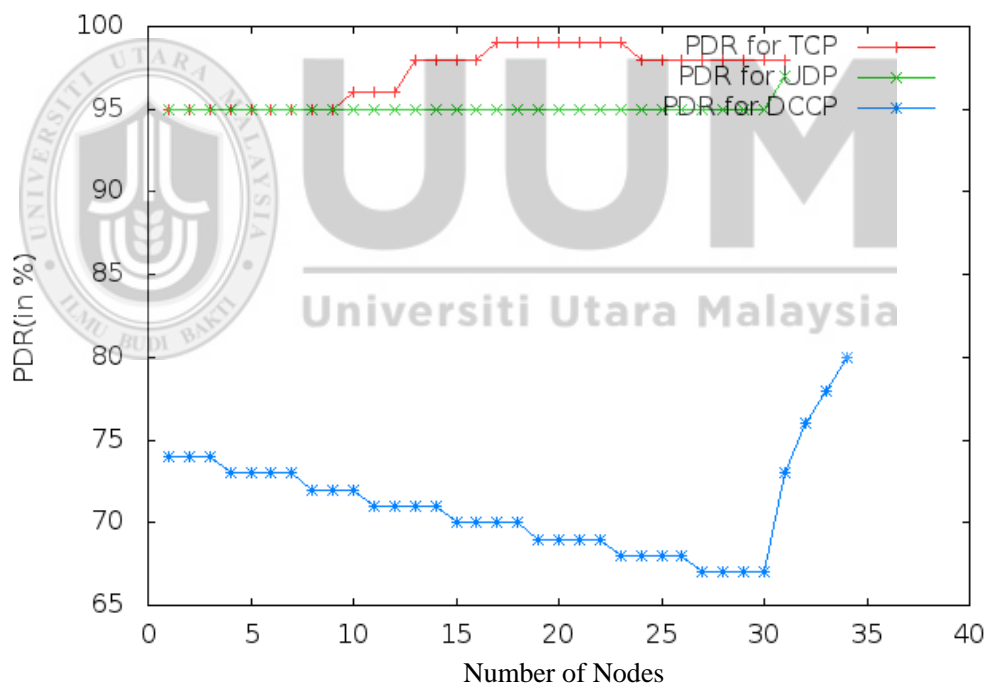


Figure 5.9: The results of comparing the PDR of TCP/UDP/DCCP protocol for 30 nodes.

From Figures 5.7-5.9, it can be seen that they reflect similar behaviours of protocols in Figures 5.1-5.3. For TCP, PDR increases proportionally with throughput. For UDP, PDR continues with 95% most of the simulation time. By this, it shows a more

stability behaviour comparing with others due to the stability of packets loss. For DCCP, the packets loss entails decreasing PDR. Consequently, the congestion control mechanism recovers the steady state by increasing window size. The packet delivery ratio is the rate of packets arrived at the receiver node in comparison to the total number of packets sent from the sender node. PDR for TCP socket varies, i.e., minimum 98 % to 99% approximately, which is quite good result for any network. The TCP protocol uses (ACK) while establishes the connection that is why it has good Packet Delivery Ratio. Also, the result shows the UDP protocol has good PDR than DCCP. Hence, the DCCP protocol is the worse if we measure PDR. This result would be different if we remove the constraint. This leads us to make the component of hardware which will have a big memory buffer to overcome the packet loss. And nowadays memory is available in terabytes, so it is not an issue at all.

Overall, the results are highly related with the congestion concept that is happening if the sender delivers more packets than the receiver can keep. DCCP offers a method to achieve access to congestion control methods without implementing them at the application layer of the OSI model. It has license the flow-based semantics such as TCP, but does not offer reliable in-order transmission. DCCP is helpful for applications with timing restrictions on the data transmission. Such applications consist of multiplayer online games, streaming media, and Internet telephony. At present, such application has regularly either settled for TCP or used UDP and employed their own congestion control methods, or has no congestion control at all.

#### 5.2.4 Comparison Analysis for Packets Loss

This section focuses on how many packets drop before reach the destination, (in our scenario the server). For one reason or another, when the packet drops from the node, this causes unreliable delivery in the network. If you have anything less than complete success in transmitting and receiving packets, then packet loss is happening in the end the video stream becomes interrupted. It can mean much slower download and upload speeds, poor quality VoIP audio, pauses with streaming media. Packet loss is a metric where anything greater than 0% should cause concern. Moreover the packet loss happens in the wireless network more than wire network because of sharing media among nodes.

The result is shown in the Figure 5.10. That TCP protocol has the best result while the DCCP protocol has the worst. We have already explained that in the above point. This results for 10 nodes broadcast file video to the server at the same time. Also, there is no big difference when we increase the nodes to 20 nodes. But we have to explain Figure 5.12. The packet loss happens for different reasons. We don't care about the other reasons because it is out of the scope of this research. We have to focus and show here in the Figure 5.12 that through the time is running the amount of loss packets increase. Because of the eNB become the bottleneck in our network topology. All nodes send packets at a same time to one base station. And the overload will be happened in the eNB base station through time. This is our explanation.



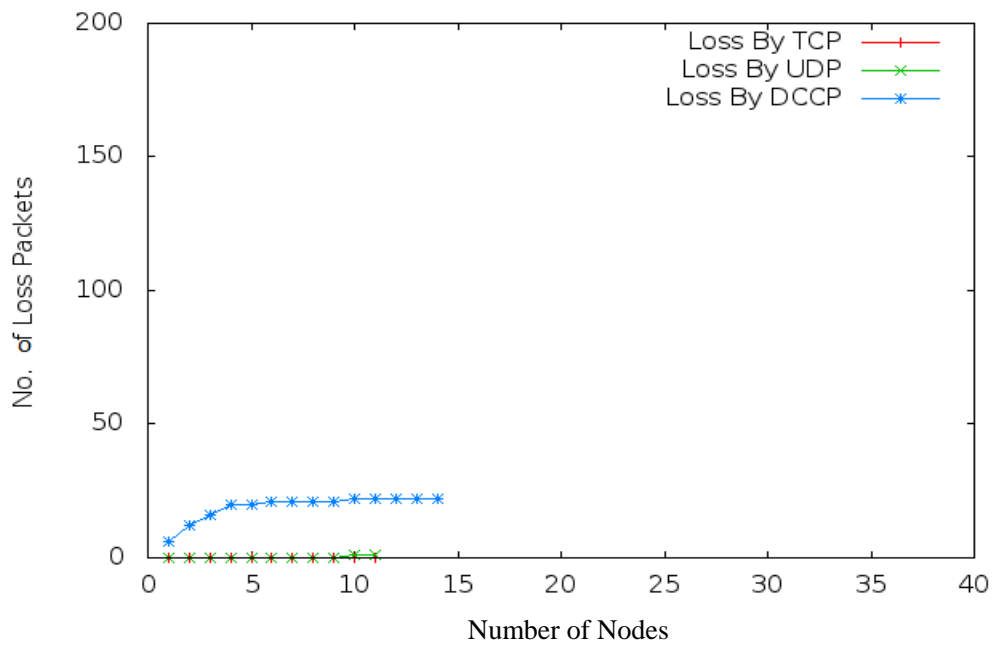


Figure 5.10: The results of comparing the packet loss of TCP/UDP/DCCP protocol for 10 nodes.

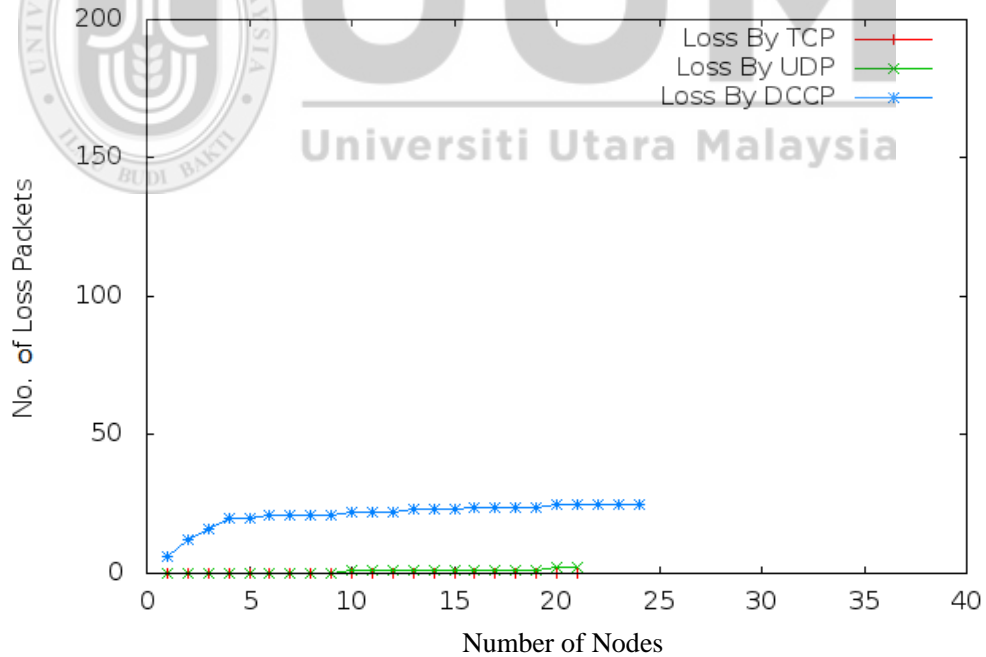


Figure 5.11: The results of comparing the loss packets of TCP/UDP/DCCP protocol for 20 nodes.

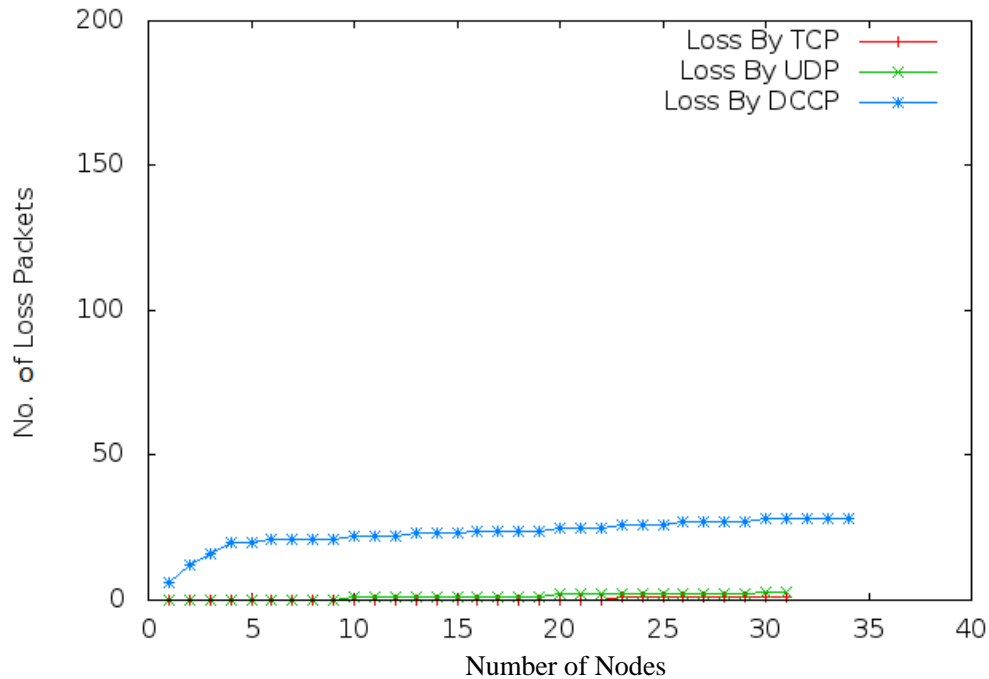


Figure 5.12: The results of comparing the loss packets of TCP/UDP/DCCP protocol for 30 nodes.

The TCP protocol has good result while the DCCP protocol has the worst. Figures 5.8-5.10 shows that the DCCP has more packets loss due to the high sending rate without full control of windows size. For UDP and TCP, they utilize all the LTE network bandwidth to send all data of video streaming.

As a result, the amount of lost packets increases as the number of nodes increased. This is due to the base station became the bottleneck which is affected by the number of nodes and transmission rate, as implemented in Chapter Four. This can be seen clearly in our network topology since all the nodes send packets at the same time to one base station. According to our topology which uses LTE environment, where the packet loss happens in the wireless network more than wire network because of the media sharing among nodes. The TCP has less packet loss because the effect of congestion controls mechanism to reduce transmitted rate. While, DCCP protocol

can detect many types of drop due to network environment, some of this drop not related with congestion, this makes DCCP keep transmitted rate as normal even when a lot drop packet happen.

Overall, the results are highly related with the congestion concept that is happens if the sender delivers more packets than the receiver can keep. DCCP offers a method to achieve access to congestion control methods without implementing them at the application layer of the OSI model. It permits the flow-based semantics like in Transmission Control Protocol (TCP), but does not offer reliable in-order transmission. Sequenced delivery within multiple streams as in the Stream Control Transmission Protocol (SCTP) cannot be offered by DCCP. DCCP is helpful for applications with timing restrictions on the transmission of data such as MPEG-4 application. At present, such applications have regularly either settled for TCP or used User Datagram Protocol (UDP) and employed their own congestion control methods, or have no congestion control at all.

DCCP has been developed to afford nominal functionality of unreliable data transport with congestion control and therefore attempts to deploy that only. It does not offer any flow control as offered by TCP. It also does not have support for multicasting. There is no sequenced delivery like SCTP, therefore streams are to be layered on top of DCCP. It offers the unreliable transport needed by modern day real-time applications and streaming media while running congestion control techniques. TCP utilizes a network congestion-avoidance algorithm. There are two variants proposed by TCP, i.e., Tahoe and Reno. Before we proceed further, let us know why the result in this section is different from the above section. Actually to

measure the congestion we have to use a stander algorithm with the limitation of the buffer queue. The NS-3 gives us facilities to make that in easy way.

Results show that the DCCP protocol has the best throughput when the number of Ue becomes 10 & 20. But the UDP and TCP protocols have less throughput if compared with DCCP. The difference of throughput between UDP and TCP is small difference even with this small difference the TCP is better than UDP protocol. The DCCP protocol has fewer throughputs when the number of UEs becomes 30. Also the result shows the TCP, then UDP protocol have less loss packets. Because the TCP protocol is connection oriented. Therefore DCCP uses to transfer video, voice due to real time transfer, as shown in the Table 5.1.

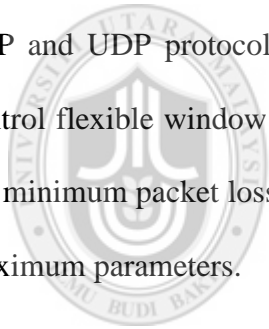
*Table 5.1. UDP/TCP/DCCP protocols based LTE environment with 10, 20, 30 nodes.*

Protocol	Throughput (Kbps)			Packet loss (in %)			Packet Delivery Ratio (in %)			Delay (second)		
	10Ue	20Ue	30Ue	10Ue	20Ue	30Ue	10Ue	20Ue	30Ue	10Ue	20Ue	30Ue
UDP	3264.69	6529.38	9794.06	2%	2%	2%	97%	97%	97%	0.0123676	0.013488	0.0146983
TCP	3299.44	6649.12	9914.12	0%	0%	1%	99%	99%	98%	0.0113728	0.012711	0.0110855
DCCP	6699.34	6715.28	6731.22	16%	17%	19%	83%	82%	80%	0.00522303	0.00511127	0.00500419

### 5.3 Summary

As seen in the above scenario for Average Delay for all three protocols, in TCP, first the delay is more during connection establishment phase, but once the connection has been established and TCP increased its window size, delay drops sharply in the data flow. But The UDP protocol shows consistently more delay due to connection less flow of the data over the network. And DCCP outperforms these both conventional connection-less and connection-oriented protocols in case of delay.

Similarly more comparison graphs are given for TCP, UDP, and DCCP for throughput, delay, PDR, and packet loss. In throughput also DCCP outperforms the TCP and UDP protocols. TCP outperforms in case of PDR due to its congestion control flexible window mechanism. Due to controlled window size TCP also gives the minimum packet loss as compared to DCCP and UDP. So seems TCP is better in maximum parameters.



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## **CHAPTER SIX**

### **CONCLUSION**

Video streaming demands more bandwidth and high quality of communication. The main contribution in terms of bandwidth is the development of coding standards for video comparison. The main contribution in terms of the quality of communication is the development of LTE technology which helped in increasing the data throughput and decreasing the latency. Through LTE environment the transportation layer protocols are the dominant players of the latest advancements in multimedia applications. These advancements motivate the need for evaluating the performance of the prominent protocols, i.e. TCP, UDP and DCCP, in MPEG-4 video data transferring in LTE. Because of the shortage in literature review in analysing these protocols for transporting this kind of data in this important environment, this thesis came to produce new experimental insights, conclusions, clarifications and results to fill this research gap. Chapter One drew a roadmap to connect the main stations in this thesis. In Chapter One, the problem statements have been identified and the research questions have been raised rationally. The significance of answering these questions has been highlighted before the thesis organization is presented.

On the problem being studied, Chapter Two is placed this thesis in its proper context with respect to the literature. Indeed, before starting the experiments, detailed discussions about related work together with supporting conceptual comparisons among existing analysing studies are provided. This chapter has been designed to know what has been done and to save this research from doing uninteresting experiments. Exploring the literature helped in suggesting important questions for

designing appropriate performance analyzing to the TCP, UDP and DCCP protocols for transferring MPEG-4 video data in LTE network. What protocol behaviours need to be analysed? What are the promising protocols to be further improved? What the evaluation metrics that are needed to clarify such behaviours? What is the coding standard of video data that is suitable in such performance analysing?

The research methods to answer the questions that emerged from studying the literature are outlined in Chapter Three with concise description about each method. Chapter Three provided two phases to achieve the research objectives. These are the scenario experiment phase to be detailed in Chapter Four and the performance evaluation phase to be detailed in Chapter five. Furthermore, the chapter presented the video data properties to be considered in the development phase and the performance metrics to be considered in the evaluation phase. The TCP, UDP, and DCCP protocols are analyzed in Chapter Five on various performance metrics they are delay, throughput, packet's loss and packet's delivery ratio. As per the simulation results on network simulator, DCCP protocol outperforms the other conventional connection-oriented and connection-less protocols in delay and throughput. The TCP performance gave maximum packet delivery ratio and minimum packet loss count due to its connection oriented architecture. As final conclusion for the multimedia applications where the packet's loss difficult to be handled, the developers should go to TCP otherwise DCCP is the best suited for real time applications with good throughput in MPEG-4 video streaming over LTE environment.

## **6.1 Contribution of the Study**

The main contribution in terms of the quality of communication is the development of LTE technology, which helped in increasing the data throughput and decreasing

the latency. Through LTE environment the transportation layer protocols were the dominant players of the latest advancements in multimedia applications. These advancements motivate the need for evaluating the performance of the prominent protocols, i.e. TCP, UDP and DCCP, in MPEG-4 video data transferring in LTE. There is a shortage in the literature review in analyzing these protocols for transporting this kind of data in this important environmental, this study came to produce new experimental insights, conclusions, clarifications and results to fill this research gap. The TCP, UDP, and DCCP protocols are analyzed on various performance metrics such as delay, throughput, packet loss and packet delivery ratio.

## **6.2 Future work**

There are several directions of this study. Firstly, because of the shortcoming of DCCP in packet loss in spite of its outperformance over other protocols, the future work will be focused on reducing the packet loss. Secondly, new horizons need to be studied in terms of performance evaluation of video transmission such as studying its performance evaluation in wireless sensor networks (WSN) because of its efficiency in energy management and investigating the effect of other coding video standards on the performance of layer protocols transforming. Furthermore, 4G technologies is very promising in wireless sensor networks due to the widening of its applications nowadays however it is suffering from problems related with the WSNs properties i.e, energy, memory and bandwidth limitation. Therefore, more efforts needed from the researcher community to reduce the energy consumption and prolong the network lifetime. Thirdly, In LTE environment, future work will be focused on its mobility challenges.



## REFERENCES

- [1] E. Dahlman, S. Parkvall, and J. Sköld, *4G: LTE/LTE-Advanced for Mobile Broadband*, Academic Press, ISBN: 012385489X, 2011.
- [2] S. S. Muttagi, S. D. Biradar, and D. T. Kushnure., “5G : A Digital Society,” in *Electrical, Electronics, Signals, Communication and Optimization (EESCO)*, 2015.
- [3] S. Singh, O. Oyman, A. Papathanassiou, D. Chatterjee, and J. G. Andrews, “Video capacity and QoE enhancements over LTE,” in *IEEE International Conference on Communications*, 2012, pp. 7071–7076.
- [4] J. Postel, “User datagram protocol,” DARPA Network Working Group Rep. RFC-768, U.S.C. Inform. Sci. Inst., Aug. 1980
- [5] J. B. Postel, “Transmission control protocol,” RFC, Information Sciences Institute, Marina del Rey, CA, vol. RFC-793, Sept. 1981.
- [6] E. Kohler, M. Handley, S. Floyd, and J. Padhye, “Datagram congestion control protocol (DCCP),” *Netw. Work. Gr. RFC 4340*, pp. 1–130, 2006.
- [7] S. Abeta, “Toward LTE commercial launch and future plan for LTE enhancements (LTE-Advanced),” in *Communication Systems (ICCS), 2010 IEEE International Conference on*, 2010, pp. 146–150.
- [8] P. Seeling and M. Reisslein, “Video transport evaluation with H.264 video traces,” *IEEE Commun. Surv. Tutorials*, vol. 14, no. 4, pp. 1142–1165, 2012.
- [9] A. Varet and N. Larrieu, “Realistic Network Traffic Profile Generation: Theory and Practice,” *Comput. Inf. Sci.*, vol. 7, no. 2, pp. 1–16, Feb. 2014.
- [10] H. V. Balan, L. Eggert, S. Niccolini, and M. Brunner, “An Experimental Evaluation of Voice Quality Over the Datagram Congestion Control Protocol,” *IEEE INFOCOM 2007 - 26th IEEE Int. Conf. Comput. Commun.*, 2007.
- [11] H. M. O. Chughtai, S. A. Malik, and M. Yousaf, “Performance evaluation of transport layer protocols for video traffic over WiMax,” in *INMIC 2009 - 2009 IEEE 13th International Multitopic Conference*, 2009.
- [12] M. A. Azad, R. Mahmood, and T. Mehmood, “A comparative analysis of DCCP variants (CCID2, CCID3), TCP and UDP for MPEG4 video applications,” in *2009 International Conference on Information and Communication Technologies, ICICT 2009*, 2009, pp. 40–45.

- [13] L. M. de Sales, H. Oliveira, A. Perkusich, and A. C. de Melo, "Measuring DCCP for Linux against TCP and UDP With Wireless Mobile Devices," in *Ottawa Linux Symposium*, 2008, pp. 163–177.
- [14] C. A. Froidi, N. L. S. Da Fonseca, C. Papotti, and D. A. G. Manzato, "Performance evaluation of the DCCP protocol in high-speed networks," in *2010 15th IEEE International Workshop on Computer Aided Modeling, Analysis and Design of Communication Links and Networks, CAMAD 2010*, 2010, pp. 41–46.
- [15] I. S. Chowdhury, J. Lahiry, and S. F. Hasan, "Performance analysis of Datagram Congestion Control Protocol (DCCP)," in *ICCIT 2009 - Proceedings of 2009 12th International Conference on Computer and Information Technology*, 2009, pp. 454–459.
- [16] P. Hofmann, C. An, L. Loyola, and I. Aad, "Analysis of UDP, TCP and voice performance in IEEE 802.11 b multihop networks," in *13th European Wireless Conf*, 2007, pp. 1–4.
- [17] S. A. Nor, S. Hassan, O. Ghazali, and A. S. M. Arif, "On the performance of TCP pacing with DCCP," in *Proceedings - 2nd International Conference on Network Applications, Protocols and Services, NETAPPS 2010*, 2010, pp. 37–41.
- [18] S. Nosheen, S. A. Malik, Y. Bin Zikria, and M. K. Afzal, "Performance evaluation of DCCP and SCTP for MPEG4 video over wireless networks," in *INMIC2007 - 11th IEEE International Multitopic Conference*, 2007.
- [19] H. K. Rath and A. Karandikar, "Performance analysis of TCP and UDP-based applications in a IEEE 802.16 deployed network," *2011 14th Int. Symp. Wirel. Pers. Multimed. Commun.*, pp. 1–5, 2011.
- [20] R. Sunkara and A. Markov, "Communication system." Google Patents, 2014.
- [21] M. H. Alferness, P. B. Criswell, D. R. Johnson, and J. R. McBreen, "Method for generating an internet protocol suite checksum in a single macro instruction." Google Patents, 1997.
- [22] Z. Haitao and B. Jill, "An improved UDP protocol for video transmission over Internet-to-wireless networks," *IEEE Trans. Multimed.*, vol. 3, no. 3, pp. 356–365, 2001.
- [23] T. Lemon, S. Cheshire, and B. Volz, "The Classless Static Route Option for Dynamic Host Configuration Protocol (DHCP) Version 4." RFC Editor, United States, 2002.

- [24] C. L. Hedrick, "Routing Information Protocol." RFC 1058, United States, 1988.
- [25] S. Zeadally and F. Siddiqui, "Voice Over Internet Protocol," *Handb. Comput. Networks*, vol. 2, pp. 468–487, 2011.
- [26] B. A. Edelman, J. Gay, S. Lozben, and P. Shetty, "Real-time priority-based media communication." Google Patents, 2014.
- [27] A. V. Verma and S. Dhawan, "Comparison and Assessment of TCP & UDP protocols in Different Network Scenarios," *Int. J. Softw. Hardw. Eng.*, vol. 2, no. 2, pp. 2–5, 2014.
- [28] A. Vetro, T. Wiegand, and G. J. Sullivan, "Overview of the stereo and multiview video coding extensions of the H. 264/MPEG-4 AVC standard," *Proc. IEEE*, vol. 99, no. 4, pp. 626–642, 2011.
- [29] E. Kohler, M. Handley, S. Floyd, and J. Padhye, "Datagram congestion control protocol (DCCP)," 2006.
- [30] F. Khan, *LTE for 4G mobile broadband: air interface technologies and performance*. Cambridge University Press, 2009.
- [31] M. A. R. Khan and M. S. Iqbal, "Emerging Technologies: LTE vs. WiMAX," *Int. J. Comput. Sci. Bus. Informatics*, vol. 9, no. 1, 2014.
- [32] S. Abeta, "Toward LTE commercial launch and future plan for LTE enhancements (LTE-Advanced)," in *2010 IEEE International Conference on Communication Systems (ICCS)*, 2010.
- [33] S. Shukla, V. Khare, S. Garg, and P. Sharma, "Comparative Study of 1G, 2G, 3G and 4G," *J. Eng. Comput. Appl. Sci.*, vol. 2, no. 4, pp. 55–63, 2013.
- [34] A. S. Viji, "Advanced high-resolution, low-delay medical ultrasound video communication using H. 264/AVC over LTE network," in *India Conference (INDICON), 2014 Annual IEEE*, 2014, pp. 1–6.
- [35] R. K. Ahir, "Improving a performance of MPEG video streams with different UDP variants," *Int. J. Adv. Reseach Comput. Commun. Eng.*, vol. 2, no. 12, p. 4, 2013.
- [36] N.-E. Rikli and S. Almogari, "Efficient priority schemes for the provision of end-to-end quality of service for multimedia traffic over MPLS VPN networks," *J. King Saud Univ. Inf. Sci.*, vol. 25, no. 1, pp. 89–98, 2013.
- [37] S. D. Strowes, "Passively measuring TCP round-trip times," *Commun. ACM*, vol. 56, no. 10, pp. 57–64, 2013.

- [38] R. Koenen, "Overview of the MPEG-4 Standard," *Int. Organ. Stand. Organ. Int. Norm. Iso/Iec Jtc1/Sc29/Wg11 Coding Mov. Pict. Audio*, pp. 1–74, 2002.
- [39] O. Landsiedel and G. Minden, "MPEG-4 for Interactive Low-delay Real-time Communication," no. ITTC-FY2004-TR-23150–10, p. 78, 2003.
- [40] F. Pereira and T. Ebrahimi, *The MPEG-4 Book*. Preason, 2002.
- [41] J. Watkinson, *The MPEG Handbook MPEG-1, MPEG-2, MPEG-4, Second Edition*. Taylor & Francis, 2004.
- [42] T. Sikora, "The MPEG-4 video standard verification model," *IEEE Trans. Circuits Syst. Video Technol.*, vol. 7, no. 1, pp. 19–31, 1997.
- [43] M. N. Khalid, "Simulation Based Comparison of SCTP, DCCP and UDP Using MPEG-4 Traffic Over Mobile WiMAX/IEEE 802.16 e," *Sch. Comput. Blekinge Inst. Technol. SE-371*, vol. 79, 2010.
- [44] H. W. H. Wang, Y. J. Y. Jin, W. W. W. Wang, J. M. J. Ma, and D. Z. D. Zhang, "The performance comparison of PRSCTP, TCP and UDP for MPEG-4 multimedia traffic in mobile network," *Int. Conf. Commun. Technol. Proceedings, 2003. ICCT 2003.*, vol. 1, 2003.
- [45] M. Al-akaidi and R. Hamzaoui, "TRANSPORT AND MAC CROSS-LAYER PROTOCOL FOR VIDEO SURVEILLANCE OVER WIMAX," 2012.
- [46] a Vetro, T. Wiegand, and G. J. Sullivan, "Overview of the Stereo and Multiview Video Coding Extensions of the H.264/MPEG-4 AVC Standard," *Proc. IEEE*, vol. 99, no. 4, pp. 626–642, Apr. 2011.
- [47] J. Joint Video Team (JVT) of ISO/IEC MPEG and ITU-T VCEG, *Recommendation and final draft international standard of joint video specification (ITU-T Rec. H. 264/ ISO/IEC 14496-10 AVC)*. 3003.
- [48] S. Suherman, M. Al-akaidi, and N. Mubarakah, "A Transport Layer Protocol for Uplink WiMAX Video Streaming," vol. 10, no. 1, pp. 19–32, 2015.
- [49] A. K. Chaurasia and A. K. Jagannatham, "Dynamic parallel TCP for scalable video streaming over MIMO wireless networks," *Proc. 2013 6th Jt. IFIP Wirel. Mob. Netw. Conf. WMNC 2013*, 2013.
- [50] S. M. A. Kazmi and S. M. H. Zaidi, "Concurrent Multipath Transfer in Fi-Wi Access Networks," pp. 209–213, 2013.
- [51] A. Talukdar, B. Mondal, M. Cudak, A. Ghosh, and F. Wang, "Streaming video capacity comparisons of multi-antenna LTE systems," in *IEEE Vehicular Technology Conference*, 2010.

- [52] L. Zhang, T. Okamawari, and T. Fujii, "Performance evaluation of TCP and UDP during LTE handover," in *IEEE Wireless Communications and Networking Conference, WCNC*, 2012, pp. 1993–1997.
- [53] B. Singh and R. Hans, "TCP and UDP Based Performance Analysis of AODV, DSR and DSDV Routing Protocols Under Different Traffic Conditions in Mobile AdHoc Networks," vol. 8, no. 2, pp. 73–92, 2015.
- [54] N. Nyame-Asiamah, Frank; Patel, "Research methods and methodologies for studying Organizational Learning," *Eur. Mediterr. Conf. Inf. Syst.*, vol. 2009, p. 15, 2009.
- [55] J. van Aken and A. Romme, "A design science approach to evidence-based management," *Rosseau, Denise M*, pp. 140–184, 2012.
- [56] V. Vaishnavi and B. Kuechler, "Design Science Research in Information Systems Overview of Design Science Research," *Ais*, p. 45, 2004.
- [57] X. C. X. Chang, "Network simulations with OPNET," *WSC'99. 1999 Winter Simul. Conf. Proceedings. 'Simulation - A Bridg. to Futur. (Cat. No.99CH37038)*, vol. 1, 1999.
- [58] A. Varga, "OMNeT++," in *Modeling and Tools for Network Simulation*, 2010, pp. 35–59.
- [59] T. Issariyakul and E. Hossain, "Introduction to Network Simulator NS2," *Network*, vol. 2, pp. 1–16, 2009.
- [60] T. R. Henderson and G. F. Riley, "Network Simulations with the ns-3 Simulator," in *SIGCOMM'08*, 2006, p. 527.
- [61] G. Piro, N. Baldo, and M. Miozzo, "An LTE module for the ns-3 network simulator," in *Proceedings of the 4th International ICST Conference on Simulation Tools and Techniques*, 2011, pp. 415–422.
- [62] M. Vasiliu, *Le Language C++*, Pearson Education France, vol. 43, no. 4. 2005.
- [63] G. F. Riley and T. R. Henderson, "The ns-3 network simulator," in *Modeling and Tools for Network Simulation*, 2010, pp. 15–34.
- [64] M. Mezzavilla, "Communication protocols and simulation tool development for multimedia traffic optimization in LTE networks," *2011 IEEE Int. Symp. a World Wireless, Mob. Multimed. Networks*, pp. 1–3, Jun. 2011.
- [65] R. Rosen, "Layer 4 Protocols," in *Linux Kernel Networking*, Springer, 2014, pp. 305–344.

- [66] J. Abraham, "NetAnim 3.105," *www.nsnam.org*, 2015. [Online]. Available: [https://www.nsnam.org/wiki/NetAnim\\_3.105](https://www.nsnam.org/wiki/NetAnim_3.105).
- [67] G. Carneiro, P. Fortuna, and M. Ricardo, "FlowMonitor - a network monitoring framework for the Network Simulator 3 (NS-3)," *Proc. 4th Int. ICST Conf. Perform. Eval. Methodol. Tools*, vol. 3, p. 10, 2009.
- [68] Nsnam.org, "Design Documentation," 2015. [Online]. Available: <https://www.nsnam.org/docs/models/html/lte-design.html>.
- [69] L. Valtulina, "MANAGEMENT ( DMM ) SOLUTION IN," no. November, 2013.
- [70] O. Oyman, J. Foerster, Y. J. Tcha, and S. C. Lee, "Toward enhanced mobile video services over WiMAX and LTE," *IEEE Commun. Mag.*, vol. 48, no. August, pp. 68–76, 2010.

