PERFORMANCE ANALYSIS OF VIDEO TRANSMISSION PROTOCOLS OVER WIRELESS LOCAL AREA NETWORK

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PROJEK YANG DIKEMUKAKAN UNTUK MEMENUHI SEBAHAGIAN DARIPADA SYARAT MEMPEROLEHI IJAZAH SARJANA SAINS KOMPUTER

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DECLARATION

I hereby declare that the work in this research is an authentic study based on my work, except for quotation and summaries, which have been duly authorized and acknowledged.

29 January 2018 MOHAMED ALI MOHAMED
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ABSTRACT

With the development of various access technology and multimedia streaming over the internet and the demand of use internet for streaming the multimedia applications has increased significantly. Delivering an excellent quality of service requirements for different applications over wireless network environment is challenging. Transport layer protocols play a major role in affecting the delivered quality of service. The transport layer offers end-to-end data transport over the network environment. Nowadays, the most popular and recommended transport protocols that used over the internet for multimedia application are Transmission Control Protocol (TCP) and User Datagram Protocol (UDP). There are a number of protocols have been developed to meet the requirements of the video streaming applications and we have used two of them in this research which are Stream Control Transmission Protocol (SCTP) and Datagram Congestion Control Protocol (DCCP). In this research, we compare and evaluate the performance of TCP, SCTP, and DCCP transport protocols for video transmission over wireless local area network environment in order to find out which protocol performs well over the wireless network. The performance of an MPEG-4 video transmission is evaluated by using Network Simulation 3 as the simulator. We have analyzed the performance by using performance metrics like average throughput, packet loss, packet delivery and delay. The obtained results show that DCCP protocol performs better in terms of throughput and lesser delay but more packet losses when compared to other protocols. Based on the result, DCCP protocol is recommended to be used in video transmission over wireless local area network.

ABSTRAK

Dengan perkembangan kemudahan teknologi dan aplikasi penstriman multimedia yang semakin pesat melalui internet telah meningkatkan permintaan terhadap penggunaan internet untuk penstriman aplikasi multimedia tersebut. Oleh itu, untuk menyampaikan keperluan kualiti perkhidmatan yang baik bagi aplikasi yang berbeza melalui rangkaian tanpa wayar adalah sangat mencabar. Lapisan protokol pengangkutan memainkan peranan yang utama terhadap kesan kepada kualiti perkhidmatan video yang dihantar. Lapisan pengangkutan menawarkan pengangkutan data hujung-ke-hujung melalui persekitaran rangkaian. Pada hari ini, protokol pengangkutan yang popular dan dicadangkan dalam penggunaan aplikasi multimedia melalui internet adalah Transmission Control Protocol (TCP) dan User Datagram Protocol (UDP). Terdapat beberapa protokol yang telah dibangunkan untuk memenuhi keperluan aplikasi penstirman multimedia dan melalui kajian ini, kami telah menggunakan dua daripadanya iaitu Stream Control Transmission Protocol (SCTP) dan Datagram Congestion Control Protocol (DCCP). Dalam kajian ini, kami membandingkan dan menilai prestasi protokol pengangkutan TCP, SCTP, dan DCCP untuk penghantaran video melalui persekitaran rangkaian kawasan setempat tanpa wayar untuk mengetahui protokol mana yang berfungsi dengan baik melalui rangkaian tanpa wayar. Prestasi penghantaran video MPEG-4 dinilai melalui Network Simulator 3. Kami telah menganalisis prestasi berdasarkan metrik prestasi seperti purata daya pemprosesan, kehilangan paket, penghantaran paket dan kelewatan hujung ke hujung. Hasil keputusan yang diperolehi menunjukkan protokol DCCP berfungsi lebih baik berdasarkan nilai daya pemprosesan yang tinggi dan kelewatan penghantaran yang rendah tetapi mengalami kehilangan paket yang tinggi apabila dibandingkan dengan protokol yang lain. Melalui keputusan analisis yang telah dilaksanakan, protokol DCCP adalah dicadangkan untuk digunakan dalam penghantaran video melalui rangkaian kawasan setempat tanpa wayar.

TABLE OF CONTENTS

LIST OF TABLES

LIST OF FIGURES

LIST OF ABBREVIATIONS

ACK	Acknowledgement
ADSL	Asymmetric Digital Subscriber Line
AP	Access Point
ASF	Advanced System Format
AVI	Audio Video Interleave
B-frames	Bi-directionally predicted frames
CCID	Congestion Control Identifier
CWND	Congestion Window
DCCP	Datagram Congestion Control Protocol
FLV	Flash Video
GoP	Group of Pictures
GUI	Graphical User Interface
HSPA	High-Speed Downlink Packet Access
I-frames	Intra-coded frames
IP	Internet Protocol
LTE	Long Term Evolution
MPEG	Moving Picture Experts Group
MTU	Maximum Transmission Unit
NCR	National Cash Register
NS	Network Simulation
OMNET	Objective Modular Network Testbed
OPNET	Optimized Network Engineering Tools
OSI	Open System Interconnection
PC	Personal Computer
PDR	Packet Delivery Ratio
P-frames	Predicted frames
QoS	Quality of Service

QT	Quick Time
RTT	Round Trip Time
SACK	Selective Acknowledgment
SCTP	Stream Congestion Protocol
SYN	Synchronize
TCP	Transport Control Protocol
TRFC	Transactional Remote Function Call
UDP	User Datagram Protocol
VOIP	Voice Over Internet Protocol
WAN	Wide Area Network
WCDMA	Wideband Code Division Multiple Access
WLAN	Wireless Local Area Network
WMA	Windows Media Audio
WSN	Wireless Sensor Networks

CHAPTER I

INTRODUCTION

1.1 RESEARCH BACKGROUND

Recently, wireless internet technologies play a major role in global internet structure and wireless network environment have improved significantly. Also, it has changed from being costly technology for a few people can used in the previous and towards to today`s global system that the majority of people in the world can use it, and has drawn a marvelous attention from the industries and academies (Atiquzzaman & Ivancic 2003). Meanwhile, due to the suitability of wireless networks, more internet users choose to connect to the internet with wireless components for easy use and browse the internet, such as laptop computers and personal devices, etc. In addition, the demand for mobile multimedia streaming has increased rapidly on the internet, especially over a wireless network environment in recent years which is easier for the users to connect on it.

The challenging requirement of the multimedia applications streaming is achieving the good quality requirements as well as the best utilization of the network resources. In order to obtain the quality of service requirements and the effectual resource use, there is one of the significant factors that responsible to send the video to the application which is transport protocols. Typically, video transmission over a wireless network environment requires retransmit to deliver the video streaming to the receiver successfully in case of packet loss, being a cause to increased delay time for the data to reach at the receiver (Lee et al. 2005).

During video communication, typically the receiver displays the received video unceasingly. Like continuous display needs delivery of video data on time, and also convert to stringent (QoS) requirements such as bandwidth, delay, and packet loss on the underlying of the network.

There is a common requiring multimedia applications which is video as it imposes severe delay and bandwidth necessities at network. The multimedia file is encoded by using several video formats such as Moving Picture Experts Group (MPEG) standard, Audio Video Interleave (AVI) which typically offers less compression than MPEG. Moreover, there are some common type of formats that used for video streaming like windows media audio (WMA), and flash video (FLV) (Pornpanomchai et al. 2008; Vu et al. 2007).

1.2 PROBLEM STATEMENT

Many researchers have published several types of research which related to video transmission with using different transport protocols such as user datagram protocol (UDP), stream congestion transmission protocol (SCTP) protocols, transmission control protocol (TCP) and datagram congestion control protocol (DCCP). Moreover, they have implemented these protocols over different network types such as on wired, WiMAX, 4-G LTE and other environments.

Network sources play a major role that used to transport video from server to client over the wireless network. Nowadays the users expect to receive a high-quality service (QoS) of the video, with fast time to receive the video and with less time of delay and also with a low number of losing data packet while serving the video over the wireless network. One of the main problems faced by the users, that we have several transport protocols could be used when transmit a video from client to the server over the wireless network environment.

In order to choose the right protocol for video transmission has been always a debatable issue that influence by the performance metrics like average throughput, delay, packet loss ratio and packet delivery ratio and another metrics (Awang Nor et al. 2015).

The requirements for receiving a high quality of service of the video streaming are the average throughput should be high with less time of delay. The delay time affects the quality of service, especially for applications which share the preference of timelines over reliability.

1.3 OBJECTIVES

Our research is going to fill up the gaps between the previous research works where there was no comparison on the video transmission of TCP, SCTP, and DCCP transport protocols over wireless network environment have been performed and investigated.

In addition, this research will give a crystal-clear idea about the transport protocols and video transmission over the network wireless network environment and also help the researchers by choosing the right protocol for transmitting the multimedia over the wireless network.

Hence in an attempt to resolve the above statement, the specific task of this research outlined below:

i) To design and conduct the experiments between three transport protocols using the video transmission from the clients to the server over the wireless local area network by using the ns3 simulator.

ii) To evaluate and analyze the result by discovering each protocol in order to get the right protocol.

1.4 RESEARCH QUESTIONS

The research questions of this research are:

1. How to use NS3 to design the experiment and compare among TCP, SCTP, and DCCP transport protocols.

2. How to analyze the result of an experiment in order to find the best protocol by using the performance metrics.

1.5 SCOPE

This research deals with video transmission from the server to the client over a wireless local area network environment. We are not presenting new protocol, we only design the right type of experiments and evaluate the results that allow us to show the strength and weaknesses of each protocol in terms of throughput, delay, and packet loss and packet delivery.

This research to design and compare between three transport layer protocols which are transmission control protocol (TCP), stream control transmission protocol (SCTP) and datagram congestion control protocol (DCCP) to choose the right protocol for video transmission over wireless local area network environment by using the simulator.

1.6 THE RESEARCH PURPOSE

The purpose of this research is, first to give the users a full knowledge of each protocol in order to let the users decide which protocol will use for video transmission over a wireless network environment.

Furthermore, it helps the users to use the right protocol to send and receive a high quality of the video wherever they are, in their work environment or for academic usage such as teaching proposes.

1.7 RESEARCH METHODOLOGY

The methodology of this research is to point out the suitable method for conducting the research and in order to determine the correct procedure for answering the research questions and also to observe the objectives of the research. By choosing the suitable method is going to help in conducting the process of research and guide us to achieve our objectives.

The methodology chapter defines the usage of experiment methodology that is used in this research in order to meet the requirements of the research. The experimental study is going to be simulated by using a simulator which called Network Simulation 3 (NS3) to bring out a solution for the problem after analyzing the results of the experiment.

This research consists of four phases which are, the theoretical studies phase, the phase of development, the implementation and simulation phase, and analysis and testing phase.

1.8 RESEARCH ORGANIZATION

This thesis is divided into five main chapters which are:

Chapter 1 is the introduction of the research, which provides a brief introduction to the wireless network and video transmission, problem statement, our target objectives, the main questions of the research, scope and methodology of the research.

While chapter 2 presents a comprehensive literature review from the researchers which related to this research and comparing among them in detail. It starts by presenting wireless network overview along with wireless local area network, overview of transport layers protocol, characteristics of TCP, SCTP, and DCCP transport layer protocols. It also presents the features that supported by each of the three protocols and video streaming includes the video frame with the importance of the compression of video.

Chapter 3 describes the methodology that used to conduct the experiments, simulation setup that which tool we will use it for the simulation, scenario and explain the performance metrics that used for analysis the performance.

Chapter 4 presents the results of the simulation in terms of using the common performance metrics and provides general discussion about the results of our experiment.

Finally, Chapter 5 concludes the experiment and provides some recommendations for future works to help out the researchers to continue extend the work.

1.9 SUMMARY

This chapter outlines the summary of the complete research, which focused on performance analysis of transport layer protocols, DCCP, SCTP, and TCP in video transmission on the wireless local area network environment. The first part gave a brief introduction of the wireless network and video streaming and how the quality of video can be affected the communication and also have mentioned some types of video format. While the following part presents the main problem of the research which occurred to most users, those sending multimedia of data over a wireless network environment. The limitation of the research is the main part that explained what we are going to do in the experiment. Then, the research objectives are trying to resolve the problem. The questions of the research that define what we will do in this research. The purpose of the research for the users those using video transmission over the wireless local area network often have been described. The method of this research that we will use it for our experiment to simulate it that called NS3. Finally, the research organizations which consist of five chapters have been discussed in details.

CHAPTER II

LITERATURE REVIEW

2.1 INTRODUCTION

This chapter introduces the wireless local area network overview and video streaming over wireless. Moreover, this chapter dedicated the previous studies and our contribution, which are related to the study as basic conception of this thesis. This research is focused on the performance analysis of video transmission protocols over a wireless network environment. TCP, UDP, SCTP, and DCCP protocols are the main component of transport layer protocol. In addition, an MPEG and traffic module will also be shown here. Hence, the differences of transport layer protocols in terms of features and services will be discussed.

2.2 WIRELESS NETWORK

The wireless network was created under National Cash Register (NCR) in 1986 and was called as WaveLan product family. A wireless network is a workstation which allows the users to telecommunication networks and share resources over the wireless network by using data connections between two or more nodes (Chen et al. 2010).

Wireless networking is a scheme by which telecommunications network, business installations, homes use it to avoid the costly process of set up cables into buildings or make it as connection between several locations. Generally, wireless telecommunications networks implemented and managed by using radio communication.

Examples of wireless networks are wireless sensor networks (WSN), cell phone networks, terrestrial microwave networks, wireless local area networks (WLANs) and satellite communication networks (Abed et al. 2012; Lin et al. 2006).

2.2.1 Wireless Local Area Network (WLAN)

Wireless local area network (WLAN) connects between two or more device using high-frequency radio waves through a short distance that depend of the channel data rate by using the wireless distribution scheme, often offering a connection over access point (AP) to the internet. WLAN allows the users to have the movement around with the coverage area such as an office or home and still having a network connection. As shown in Figure 2.1 the connection of wireless LAN to local resources and to the internet (Chen et al. 2010).

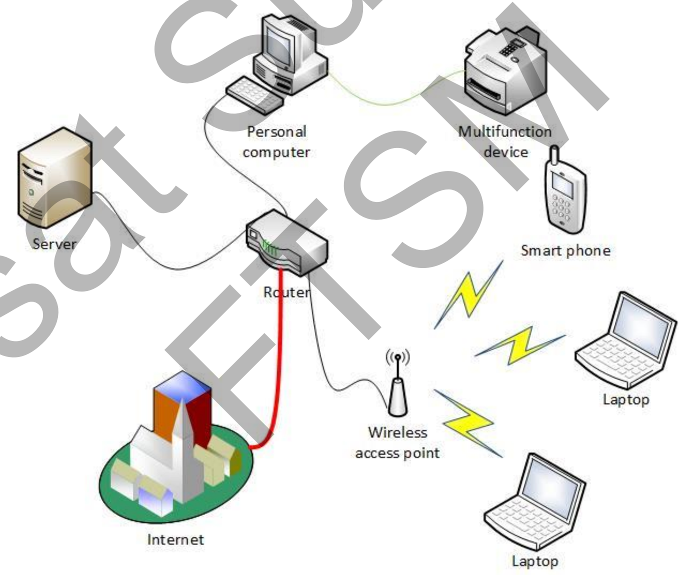

Figure 2.1 Wireless LAN

2.2.2 Advantages and Disadvantages of WLAN

Due to its nature, WLAN possesses the following advantages and disadvantages. One of the advantages of WLAN is that it provides the users with easy access. Users can access the network by connecting to the wireless router or wireless access point (AP) from any place within the AP coverage area without using any cables. Wireless AP has the ability to handle many connections simultaneously without any issues (S. Lee at al. 2004).

Moreover, WLAN hotspots can be setup almost anywhere, thus providing users the ability to access the network anytime and anywhere even when they are outside of their work environment for example, in the train or restaurant.

Despite the advantages, the usage of the wireless medium contributes to the following disadvantages of WLAN. Signals on wireless networks are noisier than signals on wired networks. WLAN AP has limited coverage area, thus more APs are needed if the coverage area needs to be extended. As WLAN uses radio waves to communicate, WLAN users also face interference issues due to radio waves signal coming from other devices (Li & Pan 2010) and the speed issue due to communication in WLAN is much slower than in wired networks.

2.3 TRANSPORT LAYER

Transport layer called as the abstract of approaches at the layered organization of protocols in network stack in the Internet Protocol (IP), and can be used for streaming the data from one device to another.

The transport layer is located at the third layer of TCP/IP model which it located between the internet layer and the application layer. One of the main objectives of the transport layer is to permit the devices on the source and the destination of the hosts to transfer data from one application to another on the wireless network.

The transport layer known as the level of services and status of the connections used when to transmit the multimedia on the layer. Furthermore, it delivers end to end communication services to the application layer and after getting the data from the application layer then sending the data to the lower layers for continues processing (Boussen et al. 2009; Shahid et al. n.d.).

Table 2.1 Comparison of the protocols

Feature	TCP	SCTP	DCCP
Packet size	20 bytes	12 bytes	12 / 16 bytes
Transport layer packet entity	Segment	Datagram	Datagram
Reliability	Yes	Yes	No
Flow control	Yes	Yes	No
Ordered data delivery	Yes	Yes	No
Multi-homing	No	Yes	No

Table 2.1 shows a comparison between the features of TCP, SCTP, and DCCP, which shows DCCP has different features in four points than from other two protocols and TCP has only one feature is different from others protocol. The first difference between them in DCCP is the header size of each packet, which takes a different form depending on the value of X field. The X field represents the Extended Sequence Number bit. If X equal to 0, the header takes 12 bytes long; if X is 1, the header is 16 bytes long. The only feature in TCP that different from others is the packet entity: while DCCP and SCTP send datagrams, TCP sends segments. The second difference is that DCCP does not ensure the delivery of messages, and provide best-effort of delivery without acknowledgments while TCP and SCTP generate the delivery of the message with acknowledged of data. The third difference is DCCP does not have flow control while TCP and SCTP have a flow control that needs three packets in order to make a socket connection before sending any data. The last difference is reordering: that DCCP does not ensure the packet reordering because reorder might reason of delay while TCP and SCTP have reordered all lost data automatically (Ali H Wheeb & Wheeb 2016). Figure 2.2 shows where the protocols are located on Open Systems Interconnection (OSI) model (Kamil & Nor 2015).

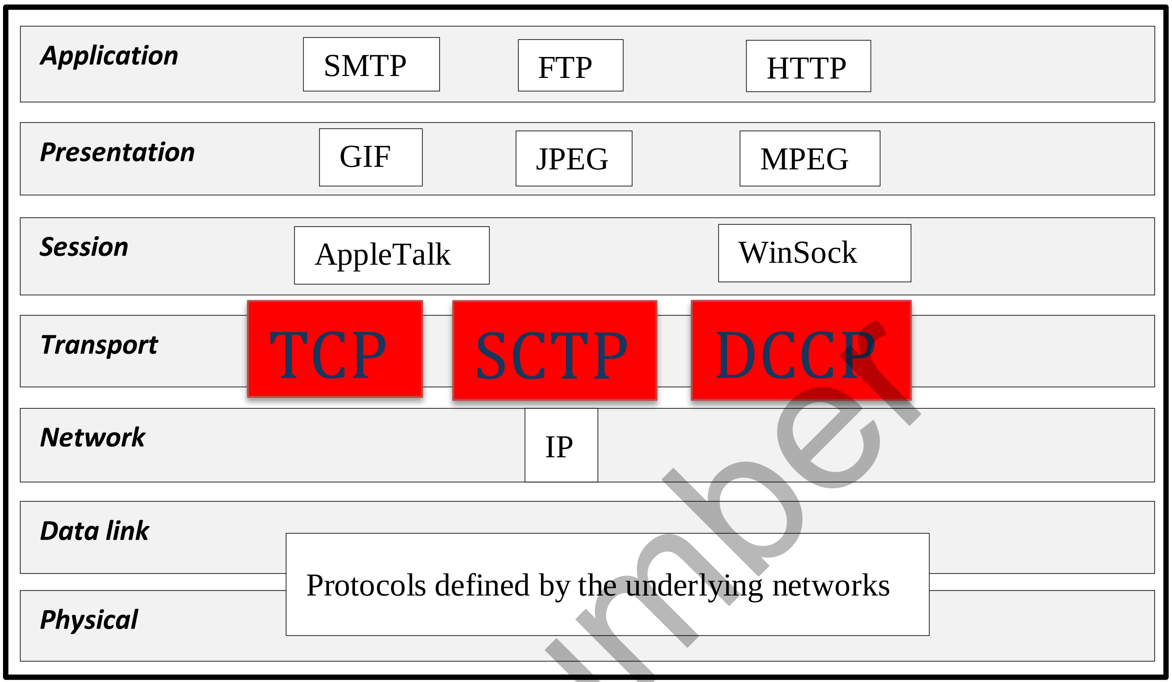

Figure 2.2 The protocols on OSI model

2.3.1 Aspects of Transport Layer

The following are the common features of transport layer that service for the applications:

i) **Connection-oriented communication:** Generally, it's such an easy for an application to read and understand the connection as a data stream instead of getting a deal with underlying connectionless datagram service, for example, the datagram model of the Internet Protocol (IP).

ii) **The same order delivery:** The packets of data in the network layer can not be guaranteed to reach at the destination in the same order that were sent from the sender, however, is a desirable feature as often. It can be done through the use of the segment numbering and the receiver sending them to the application in order.

iii) **Reliability:** The packets of data could be lost while the transmitting due to the error and network congestion, transport layer can check the data is not lost, and confirm the right receipt by sending acknowledgment message to the source. An automatic reply request methods might be used to resend the corrupted or lost data .

iv) **Flow control:** The rate of data transmission among the two hosts should be controled by the receiver to avoid fast sender when transmitting large amount of data than that receiver can buffer. The transport layer generates the expected data will not sending to the receiver later than the date that sent it (Alam et al. 2000).

v) **Congestion avoidance:** Control the congestion when the traffic entry into a communication network, so as to prevent the congestive corrupted with trying to prevent the over subscription of the link abilities of the middle nodes and also reduce the rate of transmitting packets. Such as an automatic resend requests might retain the network at a congested place. Here at this condition might be prevented by inserting a congestion avoidance to flow control include slow-start, at initial of transmission the rate will keep at a low level or after the packet of data retransmission.

vi) **Multiplexing and demultiplexing:** Ports offer several endpoint on a single host such as the name on a postal address. Each of computer appication listen for the application at their own ports which allow to use more than one network service simultaneously (Chughtai et al. 2009; Virani, Pradip G. Vanparia 2014; Xu et al. 2011).

2.4 TRANSMISSION CONTROL PROTOCOL (TCP)

Transmission Control Protocol (TCP) is defined as one of the main protocols of the (IP) suite. TCP is a connection-oriented protocol in transport layer which contains set of procedures and rules in order to control the congestion on the links to enhance the network throughput. TCP perform a high level of its functionality among two end-systems.

TCP was developed fast to achieve the growing to transmit the multimedia on the high speed links or over the wireless network. Moreover, TCP designed to be used over wired networks and also can use over wireless networks due to the important features with the same function such as, congestion control, reliability and flow control.

TCP provides important aspects like the reliability, congestion control, connection oriented, flow control and, the connection management for networks. Furthermore, best example of TCP application is, a web browser and another common example of TCP application is file transfer between two hosts and e-mail (Abed et al. 2012).

TCP divides the data into packets and confirms if the packets deliver to the receiver with the possibility of losing Internet Protocol (IP) packets, reorder, retransmit and control the network capacity in order to avoid the traffic congestions of the packets. TCP can detect these problems by rearranging the out of order data, request to retransmit the data which lost and reduce the network congestion to decrease the occurrence in future.

TCP works by exchanging the packets between two hosts, during the data transmission, TCP staying within internet protocol (IP) and application program. Since the packet transmits is not reliability until the receiver responds with ACK message that the data receives. This technique is called positive acknowledgment to guarantee the reliability of the packet transmits. It requires the receiver to reply within ACK message as the data have been received.

The sender set timer when the packet was transported and resends again the same packet when timer finishes for guarantee there is no lost or corrupted of the data during the transmission. Furthermore, the sender retains a record of the packet was sent and wait for an acknowledgement message from the receiver, when it receives successfully, the sender send the next packet (Awang Nor et al. 2015; Azad et al. 2009).

2.4.1 Features and Services by TCP

Transmission Control Protocol (TCP) is defined as one of the main protocols of the (IP) suite. It is called as a connection-oriented protocol in transport layer which covers set of procedures and rules in order to control the congestion on the links to enhance the network throughput.

The following are some of services and features that provided by TCP.

a. TCP packet

The format of TCP header`s packet shown in Table 2.2 with the explanation of each field in details.

Table 2.2 Header of TCP

16 bits				**16 bits**
Source Port				Destination Port
Sequence Number				
Acknowledgment Number				
Data Offset	Reserved	ECN	Control Bits	Window
Checksum				Urgent Pointer
Options and padding				
Data				

The source port identifies the sender of the packet while destination port classifies the receiver of the same packet and they consist of 16 bits for each port. Sequence Number identifies the packets uniquely in this segment. The sequence number bit set that is increased by one with every packet that sent by the source. The Acknowledgment number part has the value of next sequence number the source of segment is expecting to the destination that only occurred when acknowledgment number bit is set. Date offset filed is the offset that indicate from where the data starts and the TCP header is characterized in 32 bit words. The reserved filed is should be zero. (ECN) Explicit Congestion Notification which is used to protect against malicious concealment or the accidental from TCP source. Control bits filed contains of six bits which are an Urgent pointer, Acknowledgment number, Push, Reset connection, Synchronize sequence numbers and End of data flags. Window filed is called for the amount of data bytes start with one that specified in the acknowledgment part which the source of this segment is ready receives it. The checksum is the internet checksum of the TCP packet`s header and options, it is computed as the 16-bit. Also, it's depending on the checksum coverage. Urgent Pointer filed is indicated to the sequence number that from the last byte in the sequence. Options filed are taking a space at the end of the header and included in the checksum filed. If want to make TCP header`s length a several of 32 bits should be padded along with zero. And lastly, the data filed is a variable length (Chihani & Denis 2011; Khalid 2010).

b. Connection orientation

The connection in TCP is using three-way hand shake to establish the connection as shown in Figure 2.3. To initiate the connection in TCP, the client transmit Synchronize packet to the server where responds with Synchronize acknowledgment to client. Finally, the client confirms the delivery with ACK.

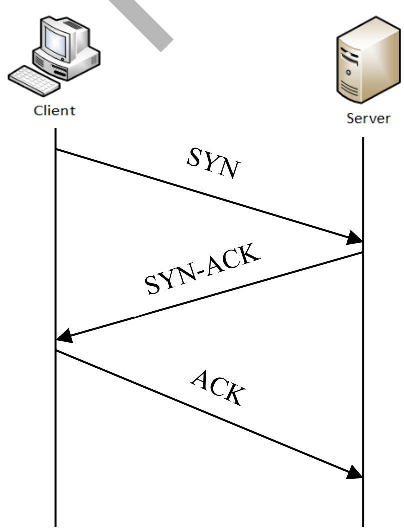

Figure 2.3 TCP 3-way handshake

c. Reliability

TCP provides a reliable transmission service by using the sequence number to identify the order of bytes that are transferred from the clients. Therefore, data could be recreated in the order, regardless of the packet loss that might happen in the transmission. It provides detection of data if duplicated, reordered or lost packet and retransmit the data.

d. Flow control

TCP provides an end-to-end flow control in order to avoid obligating sender sending the data faster for destination to receive it. It uses sliding widows flow control. In each segment in receive window part, the receiver specifies the amounts of the additionally received data that it is ready to storage for connection. The source can transfer up to that number of data only before it must waiting for ACK message from receiver as it receives.

e. Congestion control

TCP congestion control uses many schemes in order to obtain a good performance and also for avoiding the congestion failure, where the performance of the network could be dropped by many of large number of orders. These schemes in TCP used to control the data rate that enter in the network and retaining the data flow less a rate might be a cause of failure (Virani, Pradip G. Vanparia 2014).

2.5 STREAM CONTROL TRANSMISSION PROTOCOL (SCTP)

Stream Control Transmission Protocol (SCTP) is a message oriented data transport protocol, a reliable protocol that has been mainly proposed by researchers from universities and industries. However, before it was developed for telephone signaling over Internet Protocol (IP) based on the network and recently extended to be in a general proposing in transport layer for video transmission and text. SCTP is designed to operate over IP network and can be executed entirely at the operating system level (Boussen et al. 2009).

SCTP offers a reliable service and flow control methods similar to Transmission Control Protocol (TCP), it also can support unreliable transmission like User Datagram Protocol (UDP). STCP provides acknowledged, non-duplicated transmission of a message over links and error-free.

It also has many new interactive features compared to other protocols such as multi-homing services which can be helpful in mobility environment (Boussen et al. 2009; Tahir et al. 2012).

2.5.1 Features and Services by SCTP

SCTP is a reliable transport protocol and designed to operate over IP network and can be executed entirely at the operating system level. It provides features and services to its users and here is some of the services such as reliability, connection oriented and congestion and flow control.

a. SCTP packet

The format of SCTP packet is shown in Table 2.3 with explanation of each field in details.

Table 2.3 Header of SCTP

Bits 0-7	**8-15**	**16-31**
Source Port		Destination Port
Verification tag		
Checksum		
Chunk N-type	Chunk N flags	Chunk N length
Chunk N data		

The source and destination ports are already discussed early in section 2.4.1.a. The verification tag filed is used by receiver of a particular packet for authorizing the source of the same packet. Checksum filed has been explained clearly in section

2.4.1.a. The chunk type field is used to classify data available in the chunk data field. Chunk flags field are dependent on the chunk type, and they are set to zero when there is no determined by chunk type. Chunk length identifies chunk size in bytes. The last part is chunk data of the variable length (Nosheen et al. 2007).

b. Connection orientation

The connection in SCTP established by using four-way handshake as shown in Figure 2.4. Additional COOKIE method is used to prevent synchronizes (SYN) flooding attack. The connection in SCTP is called an association. The client starts the connection by sending INIT packet to the server. The server replies with INIT-ACK, which includes the cookie. Then the client responses with COOKIE-ECHO, that contains the exact cookie which had been sent by the server, then after the data exchanged between the client and the server, the association is completed.

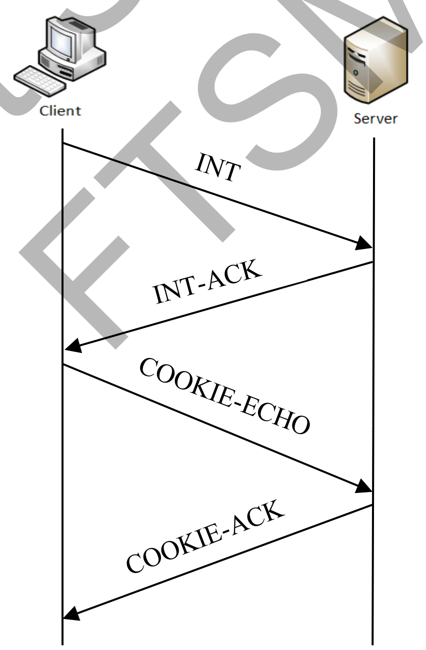

Figure 2.4 SCTP 4-way handshake

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c. Ordered data delivery

As SCTP a transport protocol and since it has multiple streams, the unordered data delivery is allowed as shown in Figure 2.5. For instance, when a stream gets affected, only that stream is temporarily blocked and the other streams will allow to pass (Chellaprabha et al. 2012).

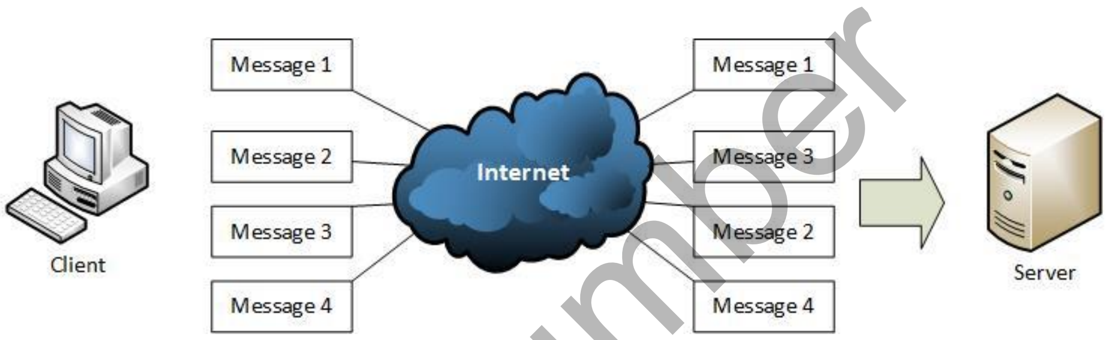

Figure 2.5 Unordered data delivery

Independent of message order, SCTP transfers reliably and processes them based on arrival sequence. This feature of SCTP minimizes the overhead caused by reordering messages on the server.

d. Reliability

SCTP provides a reliable transmission service similar to TCP and it guarantees the data is transmitted through the network sequentially with no errors. It also offers to detect the data when it is duplicated, reordered, corrupted or discarded and retransmit the data when it is damaged as necessary.

e. Flow Control

SCTP provides a method to perform flow control mechanism associated with TCP mechanism. The receiver window size has the ability in order to controls the rates that located on the receiver side and transport the data to the sender side and the receiver transmits the value of receiver window size with the all selective acknowledgment chunks. Sender retains a variable known as the Congestion Window. The bytes to be transmitted before they are ACK are be controlled by the congestion window. Data must be ACK after received and receiver could be waiting for a limit time around 300ms as usual.

f. Congestion Control

Serving a similar role to TCP and UDP, SCTP as a congestion control mechanism is not applied to individual streams but rather to the whole association. It has different modes for congestion control such as slow start and congestion avoidance. This congestion is controlled by using the CWND managed from the sender's end. The slow-start threshold is a variable at the sender end used to separate what type of congestion control mechanism to be employed in order to avoid congestion. In congestion avoidance mode, the CWND increases by one Maximum Transmission Unit per Round Trip Time (MTU per RTT). However, for slow-start mode, CWND increases much more rapidly estimated at about one MTU per SACK chunk that a sender receives.

g. Selective Acknowledgement (SACK)

For acknowledgments the data, SCTP uses SACK scheme to indicate the gaps in the transmission such as missing data blokes or disordered. SCTP only has the ability to report for large amounts of missed data packets in SACK.

h. Multi-homing

In SCTP multi-homing, the system has several interfaces. One interface marked as the primary interface and the rest as a secondary interface, for instance, wireless, and satellite, as shown in Figure 2.6. The communication is established via the primary interface and if the primary fails or inactive, the communication uses the secondary interface.

When the primary interface is available, the communication uses the primary interface. The mechanism that has the responsibility of point out which interface is fast and which interface is slow, or active or fails called as heartbeat (Khalid 2010).

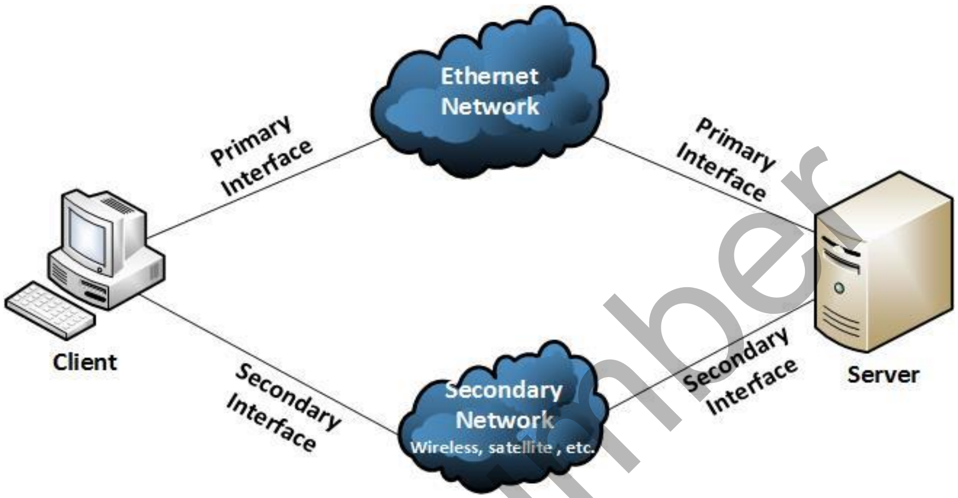

Figure 2.6 Multi-homing

2.6 DATAGRAM CONGESTION CONTROL PROTOCOL (DCCP)

Datagram Congestion Control Protocol (DCCP) is a transport protocol originally proposed by Kohler et al. in July 2001. It offers specific features intended to realize the problem with TCP and UDP transport protocols for media application requirements related to jitter and delay.

It is used to deliver congestion controlled flows for media applications, mostly over the internet but unreliable for data transmission with packet arrival acknowledgments (ACKs). DCCP offers built-in congestion control mechanism and avoids the long delay similar to TCP and provides unreliable data transmission associated with UDP. On the other hand, DCCP does not provide congestion control at the application layer and does not ensure the reliability.

DCCP has negotiation capability and reliable connection setup feature. Therefore it is suitable for the applications that transfer large amounts of data and prefer timeliness over require the reliability (Azad et al. 2009).

There are two congestion control mechanisms that supported by DCCP which are Congestion Control IDentifier 2 (CCID 2): TCP-Like Congestion Control that is useful for those applications which require a lot of bandwidth in the network and which can adjust to the changes of congestion control window, and the second mechanism is (Congestion Control IDentifier 3): TCP-Friendly Rate Control which used for applications that require to transmission the data at constant rates, as the applications are sensitive to change abruptly in the transmission rates. These mechanisms are suitable for applications where a stable rate of data transmission is obligatory instead of the reliability in transfer of packets (Bhatti et al. 2008).

2.6.1 Features and Services

DCCP has several features to service the applications which require large amount of data and prefer the timelines over require reliability. The following are some of the features and services that provided by DCCP.

a. DCCP packet

The format of DCCP packet is shown in Table 2.4 with explanation of each filed in details.

Table 2.4 DCCP header

<table>
<tr><th colspan="3">Bits 0-7</th><th colspan="2">8-15</th><th>16-31</th></tr>
<tr><td colspan="5">Source Port</td><td>Destination Port</td></tr>
<tr><td colspan="3">Data offset</td><td>CCVal</td><td>CsCov</td><td>Checksum</td></tr>
<tr><td>0</td><td>Type</td><td>X</td><td colspan="2">0</td><td>Sequence number</td></tr>
<tr><td colspan="6">Options</td></tr>
<tr><td colspan="6">Data</td></tr>
</table>

The source port and destination port are discussed earlier in section 2.4.1. Date offset is offset which begin from the start of the packet`s header till start of the data field that is characterized as 32-bit words. CCVal is the value of Congestion Control. CsCov is a checksum coverage which assigns the fields of packet that covered by checksum part and it includes header and the options. The checksum is discussed earlier in section 2.4.1.a. The type filed involves of unique values and each unique value represents unique activity such as DCCP-data, DCCP-Response, DCCP-request, which used for perform the actions. X filed is the extended sequence number. For using the extended generic header with 48 bits ACK and SYN the value of X set to 1. The packets on Sequence Number field are recognized uniquely and it increases by one with each packet that sent by the source. The data filed is a variable length (Nosheen et al. 2007).

b. Connection orientation

By using a three-way handshake, the connection creates in DCCP protocol is accomplished as shown in Figure 2.7. At this three-way handshake, the connection creates DCCP-Request that is sent to the server by the client and then the server sent back DCCP-Response to the client. Then, acknowledgment by DCCP-ACK will send back to the server by the client.

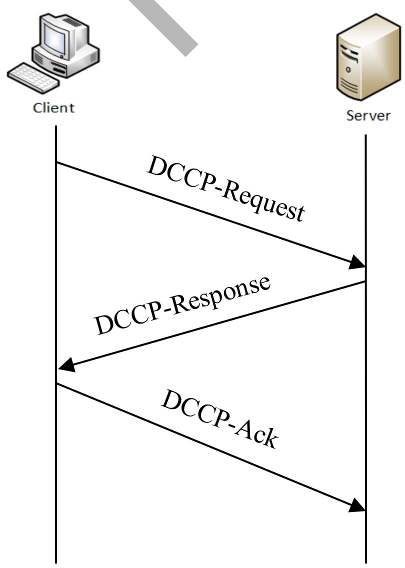

Figure 2.7 DCCP three-way handshake

c. Un-Ordered Data Delivery

DCCP as a transport layer protocol is employed for such applications which require data delivery in real-time. By reordering data, delays occur which are intolerable by multimedia applications. To avoid these delays, data using this protocol is not reordered.

d. Un-Reliability of DCCP

DCCP is referred to as an unreliable protocol intended for applications requiring real-time data delivery were delays are intolerable. As a result, this protocol does not retransmit dropped packets as applications favor most recent data over older ones.

e. Flow Control

DCCP in computer networking is regarded as a congestion control protocol rather than a flow control protocol. DCCP does not make use of flow control as transfer rates are affected by flow control limits. However, this is optional which can be implemented atop DCCP if required

f. Congestion Control

DCCP was designed to offers congestion control and it has built-in congestion control mechanisms and two kinds of congestion control mechanisms that provided by DCCP which are Congestion Control IDentifier 2 (CCID 2): TCP-Like Congestion Control t, and the second mechanism is (Congestion Control IDentifier 3): TCP-Friendly Rate Control. Congestion control IDs (CCIDs) used to select among the two mechanisms. These mechanisms are already discussed earlier in details in section 2.6 (Balan et al. 2007; Khalid 2010).

2.6.2 TCP-Friendly Rate Control (TFRC)

TCP-Friendly Rate Control (TFRC) provides a receiver-based congestion control mechanism where the sender is rate-limited by packets transmitted from the receiver with information for example, loss intervals, the time, and receive rate packets are retained in queues before being acknowledged (de Sales et al. 2008).

TFRC is one of feature that provided by DCCP and it used for applications those require to transmit the data at constant rates, as the applications are sensitive to change abruptly in the transmission rates (Ali Hussein Wheeb 2017).

The transmission rate is changed by varying the number of packets transferred and is not appropriate for the applications those share the preference of the variation in transporting rate by modifying the size of packet. In TFRC implementation, the sending rate is computed by analyzing the loss event rate based on a throughput equation. It supports ECN and, for verifying the receiver whether it reported an accurate loss event, moreover it reports the ECN Nonce Sum for all packets reported as received (de Sales et al. 2008).

2.7 VIDEO STREAMING

Video streaming consists of series of images that are transferred through the network in compressed form (encoding) and viewed by the clients after the decompressed process (decoding). Figure 2.8 displays the encoding video as sequence of images that transmitted on the wireless network to the computers and mobile devices, and playback the video after the decoding procedure (Rajarajeswari & Sutha 2013).

Video streaming applications contain the visual phone, video conference, and any multimedia applications which require large of many bandwidth, timeliness and orderliness. The sequence procedures of encoding and decoding attempt to reduce the bandwidth`s consumption for the video streaming and reduce the buffer size (Li & Pan 2010).

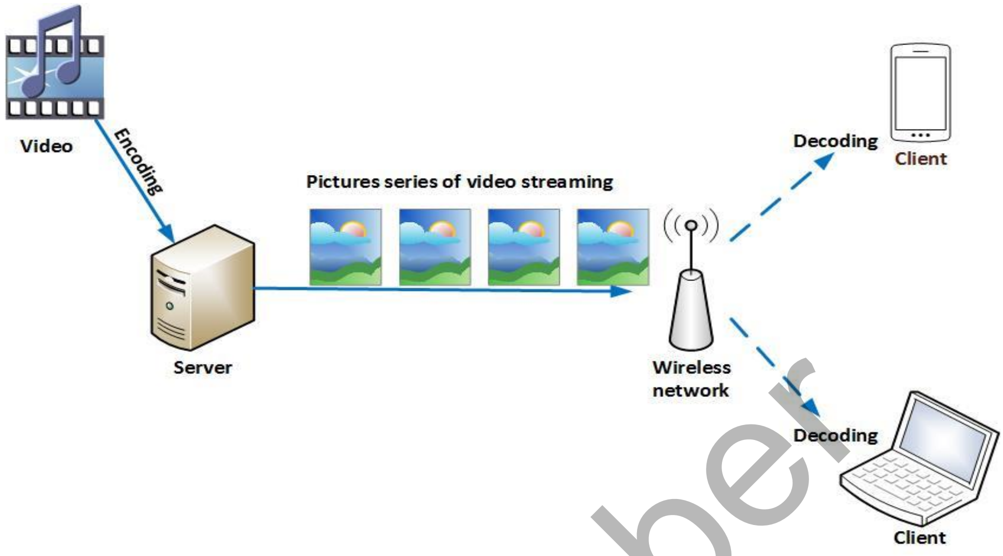

Figure 2.8 Pictures series of video streaming

The end-to-end streaming system needs some content design software. First is a streaming media server, and second is a client media player. Media clips could be designed with production tools that convert the video and audio to a format, for example, MPEG for the server to stream it. Streaming servers such as RealServer can be used to transport media clips to the clients such as RealPlayer (Lee et al. 2005; Lin et al. 2006).

2.8 MOVING PICTURE CODING EXPERTS (MPEG)

In this research, we use MPEG format for our experiment which comparing among transport layer protocols of video transmission over wireless network.

Moving Picture Coding Experts (MPEG) is a nickname for a group of international standard that was founded the group in January 1988. MPEG is used to coding visual and audio information in a digital compressed design.

For internet usage, MPEG4 is one of the most suitable as it focuses on low bit rates. Due to interaction with users, it allows for the co-existence of real-images and their computer-generated counterparts, so also the separation and receipt of different treatment. The main trait of importance MPEG4 has to the network is its capability of adaptive encoding in real-time, this in-turn enriches network utilization permits MPEG4 senders become more responsive network condition changes. This usually generates video in three types of frames; (I-frame, P-frame, and B-frame) which encodes different parts of the video signal at different levels of quality (Lin et al. 2006).

2.8.1 Traffic Module of Video

MPEG video frames are arranged into Group of Pictures (GoP) which contains of three different types of frames as described earlier in section 2.8. Each GoP consists of three frame types which are:

1. Intra-coded frames (I-frames) are encoded independently and decoded by themselves.
2. Predicted frames (P-frames) are encoded using predictions from the preceding I or P-frames and includes predicted signal data and the error of information.
3. Bi-directionally predicted frames (B-frames) are depending on both of the frames which the previous and next frames.

Normally the sequence of video is decomposed into smaller parts that can be coded together which is GoP as explained earlier. At the beginning of the sequence of GoP, I-frame is transmitted and after I-frame, a number of B-frames are transmitted and P-frames are inserted between B-frames. A standard sequence of frames, for instance, IBB–PBB–PBB (Lin et al. 2006).

GoP structure is described into two parameters, (M, N) M the distance from I-to-P frame and N is the total number of frames in the GoP which means the distance from I-to-I frame. For instance, as shown in figure 2.9, G (9, 3) which means the G has one I frame, two B frames and six P frames such as, IBB–PBB–PBB so the total is 9 frames and 3 frame intervals. Figure 2.9 shows the Sample of MPEG GOP (Takahashi 2002; Zikria et al. 2008).

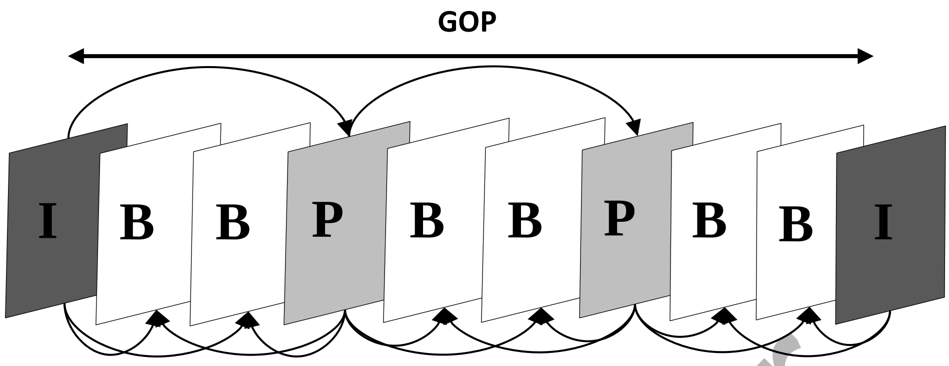

Figure 2.9 Sample of MPEG GOP

As shown in Figure 2.9, the quality of the multimedia will be affected if there is an error of I-frame data during the transmission, not only for the I-frame but all P-frames and B-frames which are predicted from I-frame because I-frame requires the larger data content than the other frames. P-frame requires a smaller numbers of bits than I-frame, and B-frame always requires the least number of bites between the frames (Chughtai et al. 2009).

2.8.2 Video Formats

Resolution is defined as number of distant pixels in one single video frame. The pixels are represented as number of rows and columns of pixels that create the display and each video formats have different resolutions. Defiantly the highest resolution of video delivers an excellent visual quality of the video. The ranges resolution of video start from 176 x 144 until 7680 × 4320 and 8K UHD has the highest resolution which is 7680 × 4320.

2.8.3 Pixel Color Depth

The color depth is the total of info about pixels that are stored and are specified in bits. It manages the pixel`s color how could be stated exactly. For better quality of the image it requires more bits in pixel information and for high color depth will be used more memory for storing the picture significantly. Moreover, for the video card to the process requirements more data to reduce the possibility of maximum refresh the rate. Table 2.5 illustrates some of the color depths and number of displayed color that are used nowadays in personal computers (PCs).

Table 2.5 Color depth that used today

Color Depth	No. of displayed color	Bytes of storage in pixel	The name of color depth
4 bit	16	0.5	Standard VGA
8 bit	256	1.0	256 color mode
16 bit	65,536	2.0	High color
24 bit	16,777,216	3.0	True color

2.8.4 Video Compression

Video streaming over the internet needs technology called as video compression because there are boundaries such as the bandwidth which applied if there is different in the streaming video. MPEG-4 is a standard for web compression and, multimedia compression (Khalid 2010).

2.8.5 Frame Inter-Arrival Rate

Frame inter-arrival rate is known as the rate at which frame can receive the frame of video and playback. The rate of frame inter-arrival from 10 - 30 frames per second (Hrudey 2008).

2.9 RELATED WORK

Several researchers published their works on different areas which are related to our research that based on performance of transport layer protocols. This work focus on TCP, SCTP and DCCP transport protocols over wireless local area network. In this section, we write about our works succinctly and some related papers to our study will be explained in details.

2.9.1 Comparison Of DCCP, SCTP, and UDP Using MPEG-4 Traffic Over WiMAX

Khalid (2010) compared and evaluated the performance based on the simulation of DCCP, SCTP and UDP transport protocols by using MPEG-4 video on mobile WiMAX. Considering mobile WiMAX network, some performance metrics like average throughput, packet loss, end-to-end delay and jitter that have been selected for the protocols over WiMAX network environment scenarios and typologies. After analyzed the performance, it has been found that SCTP and DCCP has better performance as compared to UDP. However, SCTP has a large number of packet loss ratio as compared to DCCP. Comparing DCCP and SCTP with UDP in performance of end-end delay and jitter, shows the jitter and end-to-end delay with UDP give low performance in throughput and a large numbers of packet loss, and cannot be the right protocol for transmitting the video over WiMAX. Based on the analysis, show that DCCP protocol gives the highest performance in terms of throughput, jitter, and delay among the three protocols and it is appropriate transport protocol of MPEG-4 video traffic on WiMAX wireless access technology. The reason behind why is DCCP has the best performance than other protocols because it has built-in congestion control mechanisms which have the ability to avoid the congestion in the network.

2.9.2 Performance Analysis Of TCP, SCTP and UDP, In Constant Traffic For VOIP Services

Gangurde et al. (2012) investigated the performance on SCTP, TCP, and UDP with traffic analysis over the Network Simulator 2 for voice over IP (VOIP) services and run the simulation with constant bit rates over the protocols. The results showed that TCP is execution the best with minimum numbers of packet losses when compared to SCTP and UDP. SCTP gives the best effort since it has multi-homing but it is packet delivery ACK which is time consuming. UDP retains a reliable behavior as the packets drop has no effect at it is an application. At the end, it is difficult to give a decision regarding choice of a transport protocol for VOIP traffic while SCTP cannot be used because it has higher loss rates that reduce its performance significantly compared to UDP.

2.9.3 Analysis The Performance Of SCTP, TCP and UDP, Over Wireless Sensor Network

Chellaprabha et al. (2012) evaluated the performance of SCTP, TCP and UDP protocols over congested wireless sensor network (WSN). The performance of the three protocols is measured and analyzed by using the performance metrics such as average throughput, packet delivery ratio and delay. The obtained result showed that, the throughput of UDP has an edge over its TCP and SCTP. UDP also is greatest reliable as the data packet loss is negligible at wireless sensor network as compared to TCP and SCTP. They had done it with low energy consumption and comparatively insignificant delay. After the result, it is found that UDP is the best transport protocol based among other protocols over WSN due to it is most reliable as the data packet loss is negligible when compared with TCP and SCTP. They have done the implementation by using NS2 as a simulator.

2.9.4 Simulation Of TCP, UDP, and TFRC Over Static Wireless Network

Rajaboina et al. (2015) compared the performance of TCP, UDP, and TFRC transport protocols and analyzed in wireless network. The objective of the comparison is to classify the limitation of these protocols over static wireless network. UDP sends the data at a constant rate independent of the state of the network. And the throughput and delay of UDP are high while the number of packets dropped is also high because it is lack of any congestion control mechanisms and long queuing delay at the router. TCP tries to exploit the availability of bandwidth at the initial phases and responds to congestion and reduce the data rate. The throughput of TCP is high and it oscillates and the loss rate is lower as compared to TFRC; however, the throughput of TRFC is smoother than of TCP. And, it decreases its data rate at the same time once the congestion occurs.

2.9.5 Performance Comparison Of UDP, TCP, and TFRC Over Wired Networks

Pakanati et al. (2015) analyze and compare the performance of transport layer protocols UDP, TCP and TFRC over wired networks. The reliability is the main demand in order to transfer files while delay and jitter are the primary necessities of multimedia applications. All layers are an aid to the successful operation on the Internet. They simulated first the three protocols independently and inter operation of TCP-UDP, UDP-TRFC, and TCP-TRFC is studied. TCP increases the data rate depends on the availability of the network bandwidth. UDP does not respond to the availability of the network bandwidth while it just sends everything that received the attached application. When the three protocols are interoperated, UDP affects badly both TRFC and TCP also the complete throughput. TRFC and TCP can interoperate; however, TRFC is comparatively less aggressive than TCP. Rightness is not the property handled by all the three protocols and to accomplish it suitable router mechanisms are required. After analyzing the results illustrate that TFRC requires being fine-tuned but before it`s implemented on the internet. They used Network Simulator 2 (NS2) for the implementation.

2.9.6 Analysis The Performance Of Transport Layer Protocols Using MPEG-4 Traffics On 4G Networks

Kamil & Nor (2015) analyzed and compared the performance of TCP, UDP, and DCCP transport protocols for streaming video over 4G-LTE technology. They used Network Simulation 3 (NS3) for implementation and analyses on varies performance metrics like average throughput, delay, and packet delivery ratio and packet loss. The obtained results show that DCCP gives high throughput, large numbers of packet losses and lesser delay as compared to TCP and UDP. TCP gave maximum packet delivery ratio and less packet loss as compared to other protocols. Based on the results, show that DCCP is the best suitable with high throughput for real-time video over LTE environment because it has congestions control built in that have ability to avoid the congestion of the large number of packets in the network.

2.9.7 Simulated Performance Of TCP, SCTP, DCCP and UDP Transport Protocols On 4G Networks

Awang Nor et al. (2015) analyze and evaluate the performance of MPEG video data transmission between TCP, UDP, SCTP and DCCP protocols over 4G-LTE environments. The performance metric used for evaluated the performance are throughput, jitter, delay, and packet loss ratio and evaluated at the base station by SCTP, DCCP, UDP and TCP protocols on the 4G-LTE environment. The output of the results shows that DCCP protocol performs as the best protocol in terms of throughput with the minimum of delay and jitter as compared to TCP, SCTP, and UDP. However, TCP provides highest packet delivery ratio with a minimum of packet loss ratio because it has connection orientation feature. Also for multimedia applications, the packet loss is kind of difficult to be controlled. Therefore DCCP protocol is the best choice with the highest throughput in MPEG-4 video data transmission over 4G-LTE environments among the three protocols. Because DCCP use congestion control for avoiding the large number of packet in the network and it is not reliable protocol which is suitable for video streaming. The simulator that they used to evaluate the result is Network Simulator 3 (NS3).

2.9.8 Performance Comparison Of Different Transport Protocols On Wired Network

Wheeb & Wheeb (2016) compared and evaluated the performance of the transport layer protocols UDP and TCP over a wired network. They used Network Simulator 2 (NS2) for analyzed the performance and compared the results since NS2 is common in the network community for analysis. The result has been evaluated according to performance metrics such as average throughput, end to end delay, packet delivery ratio and packet loss ratio. Constant Bit Rate (CBR) had been used for both UDP and TCP protocols.

The summary of the literature review is shown in Table 2.6.

Table 2.6 Summary of literature review

Author/year	**Network type**	**Protocols**
Khalid (2010)	WiMAX network	DCCP, SCTP and UDP
Gangurde et al. (2012)	Wireless network	SCTP, TCP, and UDP
Chellaprabha et al. (2012)	Wireless Sensor Network	TCP, UDP, and SCTP
Rajaboina et al. (2015)	Ad hoc network	TCP, UDP, and TFRC
Pakanati et al. (2015)	Wired network	TCP, SCTP, and DCCP
Kamil & Nor (2015)	4G networks	TCP, UDP, and DCCP
Awang Nor et al. (2015)	4G LTE	TCP, UDP, SCTP, and DCCP
Wheeb & Wheeb (2016)	Wired network	UDP and TCP

Based on Table 2.6 presented the previous researches on different topics of the performance analysis in transport layer protocols. The researchers did their experiments on different network types like WiMAX, wireless sensor, wired networks. Likewise, with different protocols such as DCCP, SCTP, UDP, TCP and TRFC with different performance metrics used for analysis the result.

In this research, we compare the performance of video transmission among TCP, SCTP, and DCCP transport protocols over local area wireless network environment and analyze the result in terms of many performance metrics to find the best protocol.

2.10 SUMMARY

This chapter discussed the transport layer protocols overview and video streaming. It also discussed the characteristics and applications of the TCP, SCTP and DCCP protocols with some related protocols. Furthermore, the performance of video transmission over wireless local area network has been discussed briefly. Lastly, the review of others research papers closely related to the performance of transport layer protocols was discussed.

CHAPTER III

METHODOLOGY

3.1 INTRODUCTION

This research has been done by using Network Simulation 3 (NS-3.26) as a simulator tool for the implementation of the scenario and presented the general methodology of the performance. With compare between TCP, SCTP, and DCCP transport layer protocols in video transmission over wireless local area network environment. The performance of the protocols is evaluated in terms of performance metrics like average throughput, PDR, packet loss ratio and end to end delay. Simulation method, parameters of the simulation are presented as well. The results of the simulation determine the right transport protocol to deliver high quality of service (QoS) for video transmission over a wireless network.

3.2 OVERALL METHODOLOGY

The main objective of this research is to investigate that which protocol delivers better performance by comparing the performance of TCP, SCTP, and DCCP for video transmission over wireless local area network environment, which consists of 10, 20, 30, 40 and 50 nodes. We have selected these protocols because they are the popular protocols in transport layer for multimedia streaming, and our scenario is implemented in WLAN. Different number of nodes has been executed for getting the accurate results. Figure 3.1 shows an overview of the phases of our study steps. The research starts with reviewing the latest related literature review and analyzes the data identify the problem statement to find out the objectives for our simulation. Next, we implemented the protocols using NS-3 simulator. Finally, we examined the performance of the protocols by analyzing the final results and discuss about them with our contribution.

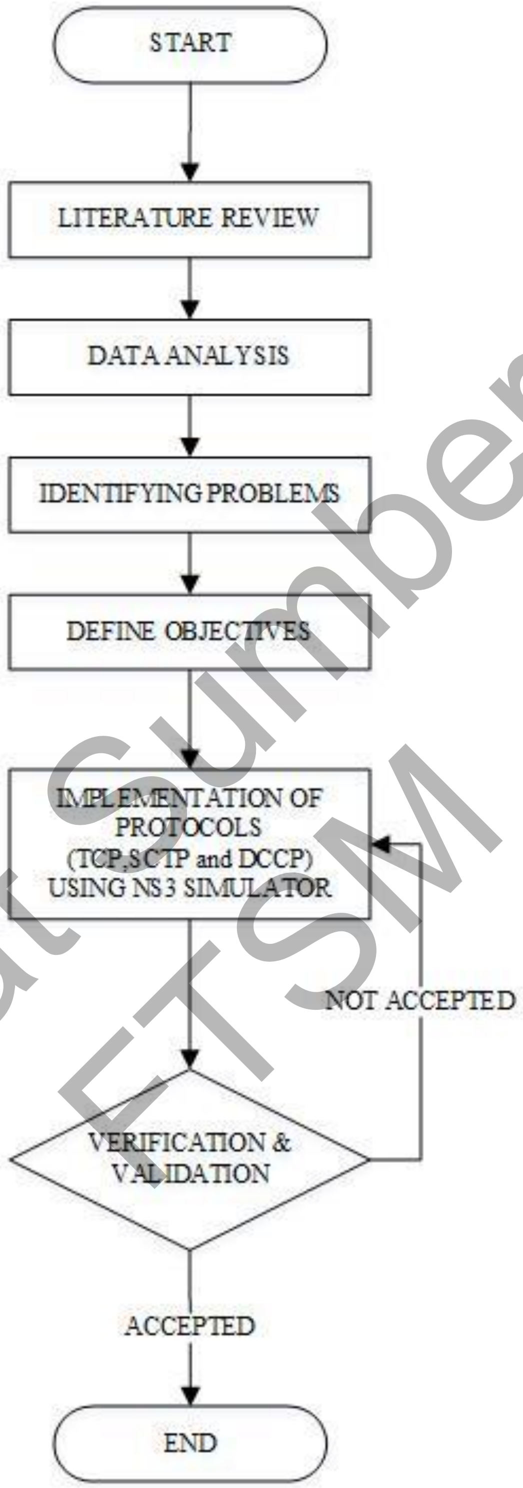

Figure 3.1 Overall methodology

3.3 EXPERIMENT METHODOLOGY

The development of our simulation is divided into five main stages as shown in Figure 3.2. It shows the simulation development process of each phase of our simulation until analysis and discusses the results.

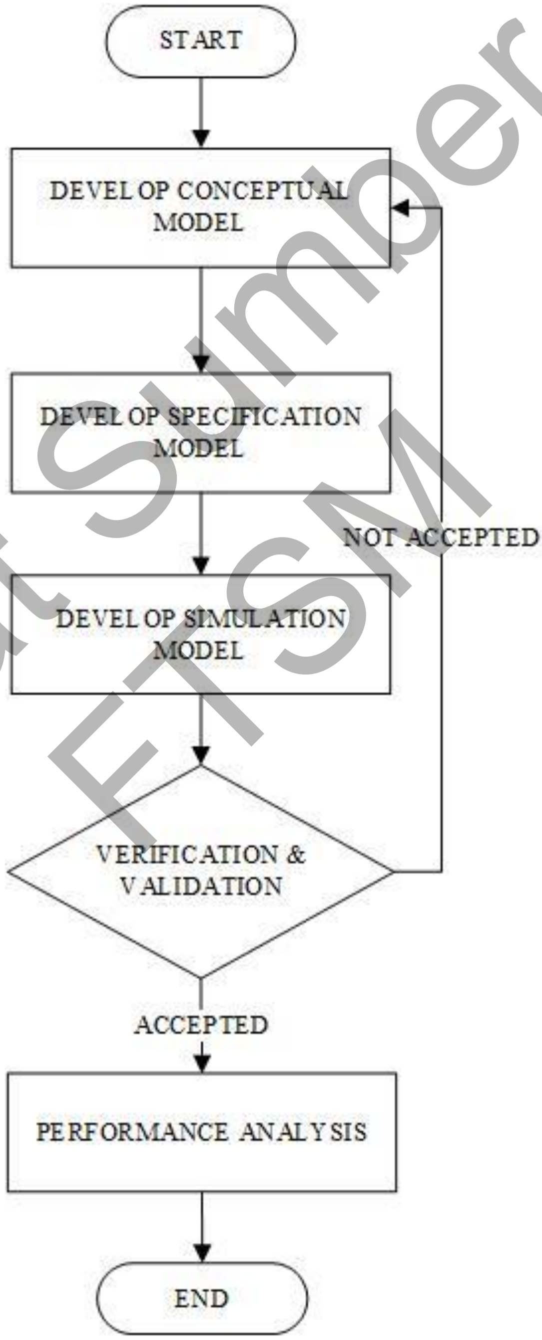

Figure 3.2 Simulation development process

a. Develop conceptual model

The conceptual model is defined as the abstracted model of the target real system. In our simulation the main conceptual model consists of one server, numbers of clients which are called nodes and they are connected to one access point (AP).

b. Develop specification model

The specification model is the details of the conceptual in order to populate the model by collecting the data. In our simulation the channel parameters are the specification of the model that are chosen as the packet size is set as 1000 Kbyte which is a suitable to measures the performance result. The transport protocols TCP, SCTP, and DCCP are placed over all the nodes, and these protocols are selected because they are the common protocols in transport layer that are used for video streaming.

c. Develop a simulation model

The simulation model is the tool that used to simulate the experiment. We used ns-3 as a simulator for our experiment which is based on C++ programming language or python.

d. Verification and validation

The verification is the process of ensuring that input to the system as per the design model specified by the conceptual and specification model. The validation is the process of ensuring that the output the system is complying with expected results that is been anticipated based in the conceptual and mathematical model. In our scenario, the verification is established by using the three protocols with the same number of nodes and environment which is ensured by the NS-3 simulator. While the validation ensured by the result comparison with the expected result of the experiment.

e. Performance analysis

The performance analysis is the last step of the processes and comes after the simulation have been verified and validated. In this experiment we analyzed the performance of the purposed transport protocols by using the common performance metrics like the average throughput, packet delivery ratio, packet loss ratio and end to end delay.

3.4 NETWORK SIMULATION

In the real world, the experiments could be expensive and complex often for the large scale multimedia streaming in physical implementation. Based on the study of multimedia streaming, there are few Network Simulation software were used for multimedia streaming system which used by the research community (Vineeth & Guruprasad 2015).

There are several Network Simulators that are used by network investigator in order to implement and evaluate the simulation scenario such as OMNET++, LTE, OPNET, NS-2 and NS-3 (Afanasyev et al. 2012).

Network Simulation 3 (NS-3) is a discrete event simulator, which could be used to implement several applications. The NS-3 project has started as an open source and it is freely available to research community and students to be used as the basis for the implementation. There are many external tools and animators in NS-3. The platform of the simulation offers users with a single, integrated graphical user interface (GUI) environment, visualization and data analysis. Recently, it has become one of the common simulator in the network methodology (Vineeth & Guruprasad 2015).

In this research, we have used Network Simulator tool as NS-3.24 to simulate our experiment, which is C++ based, and it is a discrete simulator for the education and network research evaluation. It was established early in 2006 and released in 2008 (Afanasyev et al. 2012). Figure 3.3 shows the demonstration of simulation.

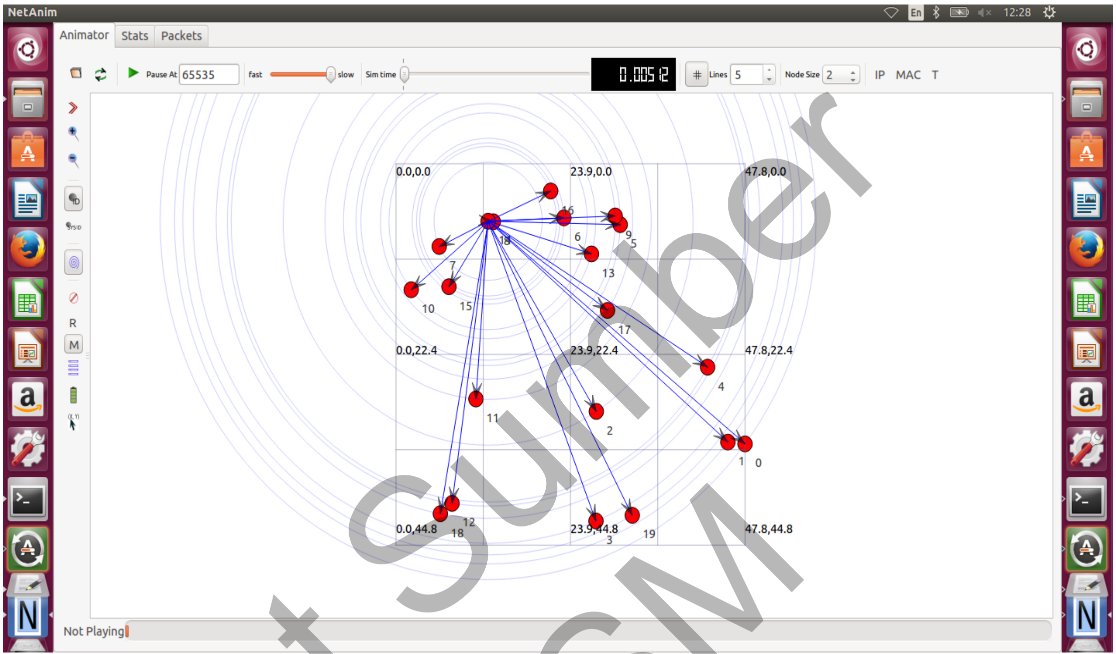

Figure 3.3 The demonstration of simulation

Previously, the users focused on the wired network for video transmission and recently several models like LTE, WiMAX and Wi-Fi models are added for improving the contribution of NS-3. Figure 3.4 illustrates the basic structure of Network Simulation-3. First, It shows the simulation script which written in C++ with optional python, executes the script by using shell command on the terminal to get the output. Finally, the graphical results in NS-3 can be interpreted by using open source software which called NetAnim that shows the animation and the movement of the experiment (Chihani & Denis 2011).

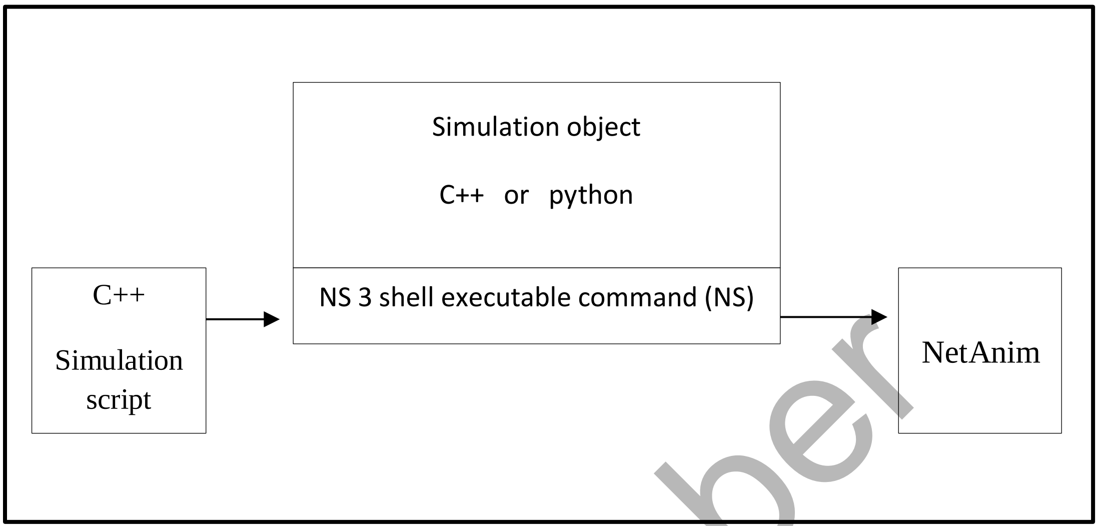

Figure 3.4 Basic structure of Network Simulator-3

3.5 SIMULATION ENVIRONMENT

This section explains the analysis and testing of the environment for the performed implementation. The structure of network shows how it models the normal network is evaluated and testing the performance. When simulate a network environment is for constructing to design a real world system that can present server-client video streaming.

Our simulation scenario is implemented in wireless local area network environment based on IEEE 802.11b has throughput up to 11 Mbps which is suitable for the scenario that we have designed. We designed the scenario topology of 10, 20, 30, 40 and 50 nodes to find out a clear different with many nodes in terms of average throughout, packet delivery ratio, packet loss ratio and end to end delay. The nodes are connecting with one access point (AP) and we assume that there is a wired link between the video server and the wireless access point. We used point to point as connection channel due to it provides the majors of what our simulation requirements. Packet size is 1000 Kbytes has set for the experiment which is a clip of the video for transmission. As shown in Figure 3.5 the target scenario of our simulation is described.

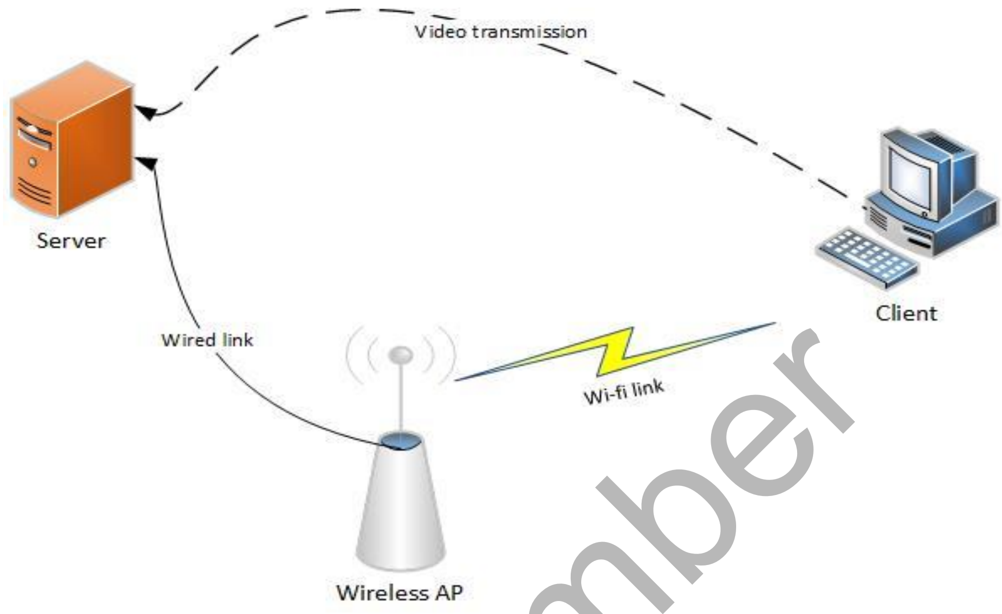

Figure 3.5 Simulation scenario

The experiment starts when the streaming server transmits a video to the clients and the server connects with the wireless access point by wired link while the transmitting between the server and clients occurred in wireless local area network (WLAN).

As shown in Table 3.1 the summary of model parameters that we have used for the proposed simulation experiment.

Table 3.1 The simulation parameters

Parameters	**Value/Name**
Network Simulation	NS-3
Protocol	TCP, SCTP, DCCP
Simulation time	80 s
Wireless network	IEEE 802.11b
No. of node	10, 20, 30, 40 and 50
Channel data rate	11 Mb/s
Packet size	1000 Kbyte

3.6 PERFORMANCE MEASUREMENT

The quality of service can be measured by using the performance metrics which is based on the designing of the network. Four performance metrics have used in this research like the average throughput, packet loss ratio, packet delivery and end to end delay for transport layer protocols which are TCP, SCTP, and DCCP. In case of packet loss cause affect the performance of video transmission over a wireless network environment (Kim et al. 2010).

3.6.1 Throughput

Average throughput is defined as the total amount of transmission data between the source and the distention which dividing through the simulation time. It is measured in term of bit per second as similar as the bandwidth is measured. The throughput in network is affected negatively through different elements like the traffic, the propagation losses and etc. The higher rate of throughput is better for the congestion control (Sales et al. 2008). The Equation 3.1 is used to calculate the value of throughput:

$$Throughput = \frac{8 \times N_{pr}}{T_{Sim}} \quad \text{... (3.1)}$$

Where

N_{pr}: The successful received number of packets.

T_{Sim}: The simulation time in second.

8: To change type of packet from bit to byte

3.6.2 Packet Loss Ratio

Packet loss ratio (PLR) refers to the ratio of the total amount of the packets lost from source to the destination during video transmission. When the packet loss rate is low that means the performance of the protocol is good (Bernardo et al. 2009). The packet loss value can be calculated by using the Equation 3.2:

$$\text{Packet loss ratio} = \frac{\sum N_{pt} - \sum N_{pr}}{\sum N_{pt}} \times 100 \qquad \dots (3.2)$$

Where

$\sum N_{pr}$: The successful received number of packets.

$\sum N_{pt}$: The successful transmitted number of packets.

3.6.3 End to End Delay

End to end delay (E2E) is defined as the average time for a packet to be sent over the network from sender to the destination. It also includes the delay occurred due to another reason such as processing delay, propagation in delay and queuing delay. It can be calculated by taking the difference of the transferred time and the received time of each packet (de Sales et al. 2008). The Equation 3.3 shows the formula to calculate the end to end delay of the packet:

$$\text{End to end delay} = \frac{\sum_{i=1}^{n}(R_i - S_i)}{\mathrm{n}} \qquad \dots (3.3)$$

Where

i: The data packet index

R_i: The time of received data packet

S_i: The time of sent data packet

n: The total number of data packets

3.6.4 Packet Delivery Ratio

Packet delivery ratio (PDR) refers to the number of received data packets to the number of transported which arrive at an endpoint successfully in comparing with the number of packets that transmitted by sender. The performance of experiment is good when the packet delivery ratio is high (Nosheen et al. 2007). It can be calculated the value of PDR by using the Equation 3.4:

$$\text{Packet delivery ratio} = \frac{\sum N_{pr}}{\sum N_{pt}} \times 100 \qquad \dots (3.4)$$

Where

$\sum N_{pr}$: The successful received number of packets.

$\sum N_{pt}$: The successful transmitted number of packets.

3.7 SUMMARY

This chapter outlines the overview of the methodology in this research which are the scenario of simulation, parameters of the simulation, and the performance metrics that use to analysis the performance. The NS-3 has been explained in details. Moreover, the chapter presented the all associated figures, the flow charts for the research have been explained, simulation methodology and performance metrics for analyzing the result.

CHAPTER IV

RESULTS AND DISCUSSION

4.1 INTRODUCTION

This chapter discusses the results of simulation based on the experiment. The performance of TCP, SCTP and DCCP transport protocols is reviewed and analyzed on video transmission over the wireless network. The final results of this scenario simulation will discover which protocol delivers better performance. The evaluation of the protocols will be explained in the rest of this chapter. Lastly, the results of our experiment are presented and general discussion about the result.

4.2 THE RESULT OF EXPERIMENT

In this part, we compare and evaluate the performance of SCTP, TCP, and DCCP transport protocols by using the results from our experiment. The experiment evaluated the average throughput, packet loss ratio, packet delivery ratio and end to end delay will be used in this scenario with a different number of nodes to study the impact of the performance.

4.2.1 Average Throughput

The average throughput refers to the total amount of transmission data rate between the source and the destination over a communication channel. Figure 4.1 shows that DCCP protocol has the highest throughput value in the wireless network among the other protocols. The scenario assumes that all the nodes are sending the video at the same time to base station node.

Furthermore, we can illustrate that when the number of nodes increases from 10 to 20, 30, 40 and 50 it shows that DCCP protocol still has the good throughput in the wireless network environment. The output confirms that the stability of DCCP protocol even with the number of video files transmission increasing when increasing in a number of nodes can make a chance of the impact of the protocol. Also, as the number of nodes increased the average throughput of the network will get enhanced. The steady growth of graph shown that network has the ability to handle the nodes. To get the highest performance, there is no bottleneck up to this limit of a number of nodes. That means if the number of nodes increases, the average throughput also increases. DCCP protocol has the highest throughout among TCP and SCTP protocols because DCCP has two mechanisms built inside it which are the congestion control and flow control. Congestion control in DCCP is different when compared with other protocols because in DCCP control the number of packets that have sent to the network when the amount of packets are larger than the availability in the network`s capacity. Flow control in DCCP protocol is different in comparing with other protocols because in DCCP only is to control the size of traffic once the sender sent up to the max limits, then the sender will receives respond from the receiver.

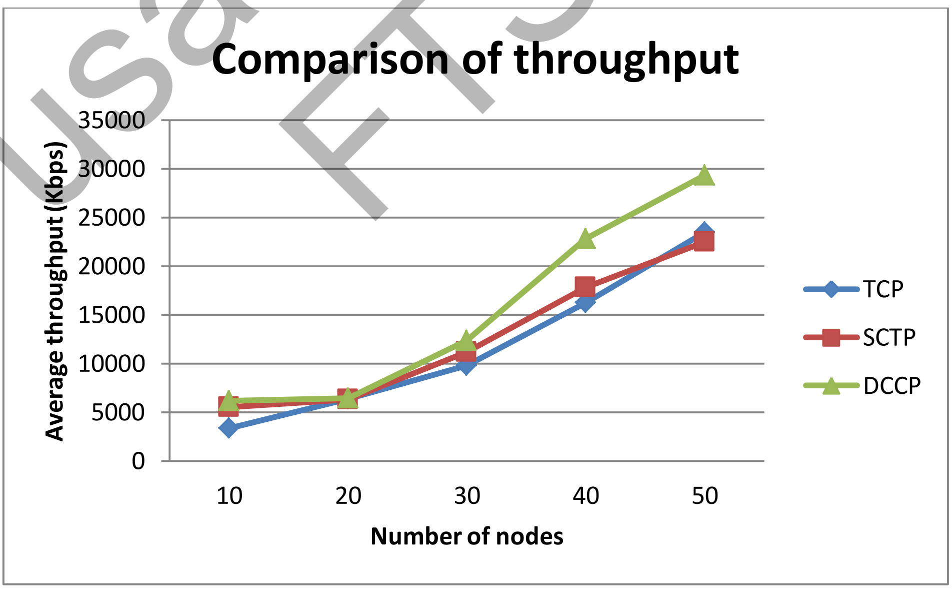

Figure 4.1 The comparison of throughput

4.2.2 Packet Loss Ratio

Packet Loss Ratio (PLR) refers to the ratio of the total number of packets lost from the source to the destination during video transmission. The packet loss always occurs in wireless network environment more than the wired network due to sharing the medium between the nodes.

Figure 4.2 shows that TCP protocol has the best result whereas DCCP protocol has the worst in packet loss for 10 nodes. As the graph shows that when increasing the nodes to 20, 30, 40 and 50 will still there is no difference. The number of loss packets increases when the number of nodes increases. The reason behind that is due to the base station became a bottleneck because got affected by a large number of nodes in the network topology since the nodes send packets at the same time to one base station only.

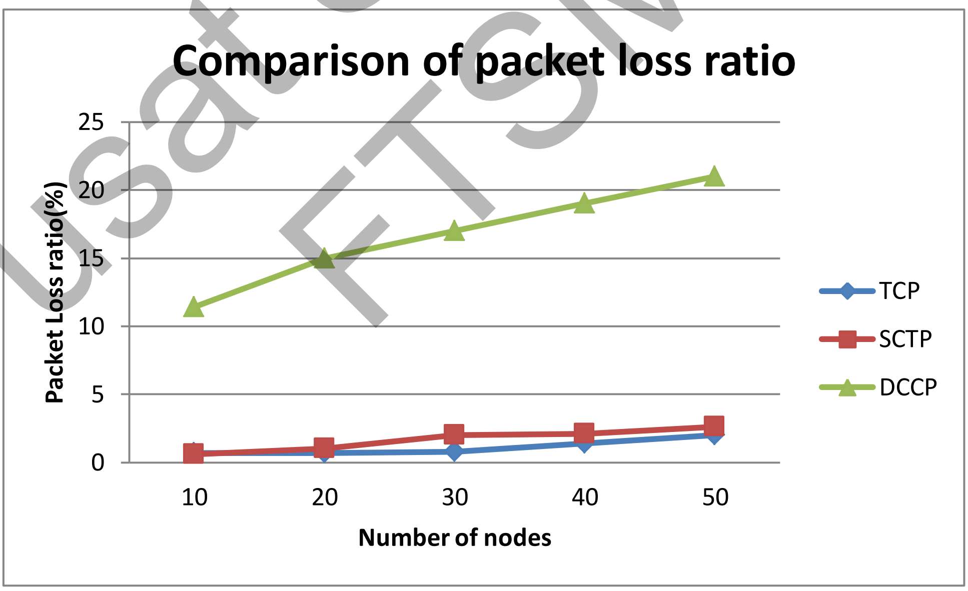

Figure 4.2 The comparison of packet loss

4.2.3 End To End Delay

End to end delay (E2E) refers to the average time for a packet to be delivered over the network from the sender to the destination. As shown in figure 4.3, DCCP protocol achieves the best result among TCP and SCTP because delay time is less compared to other protocols. TCP and SCTP protocols require more time at the beginning in order to establish the connection because they use reliability which affect the number of nodes. When the number of nodes increases, the delay increases which happens more in a wireless network than wired network because it uses media share. The average delay for TCP, SCTP and DCCP protocols on all three scenarios show that consistently more delay because the connecting less flow of data on network. TCP has the highest delay time due to TCP`s reliability consumes time more than in SCTP and DCCP in terms of time. DCCP protocol outperforms connection-oriented protocol and convocational connection-less protocols in terms of delay time. Comparative examination of TCP, SCTP and DCCP protocols the all nodes scenario shows DCCP protocol has the best performance regarding to the delay time due to it has the congestion control which has been explained in details earlier in this thesis.

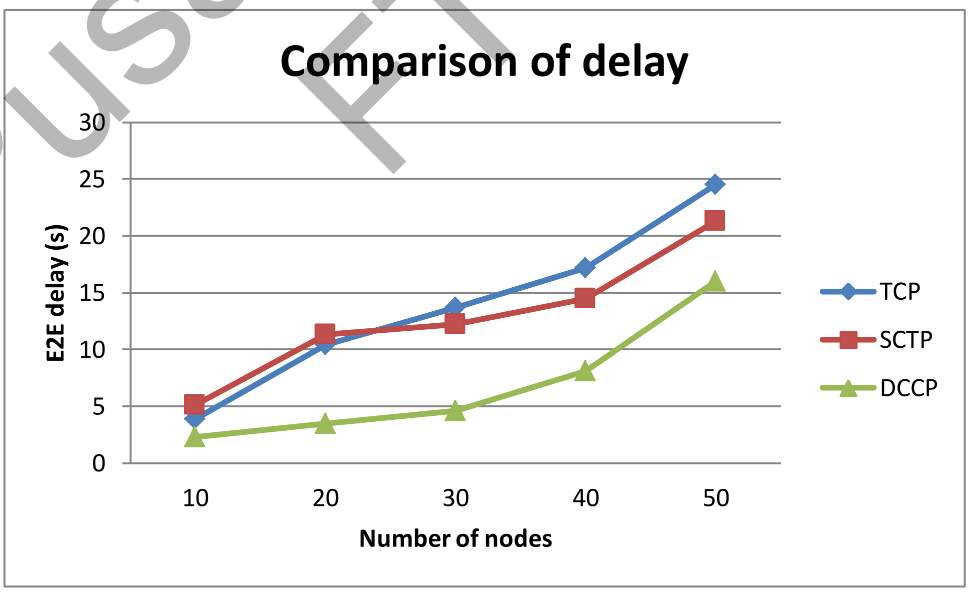

Figure 4.3 The comparison of Delay

4.2.4 Packet Delivery Ratio

Packet delivery ratio (PDR) refers to the number of the received packets to the number of transferred ones which arrive at the destination successfully in comparing with the number of packets that transmit by the source. TCP protocol has the best PDR in compared with SCTP and DCCP as shown in Figure 4.4 because it uses acknowledgment (ACK) while begins the connection. Which respond a massage to the source that it receives the packet. DCCP protocol has the worst result of packet delivery ratio and the result will be different if we have a big memory to buffer the overcome of packet loss. The results are related to congestion concept that is occurring if send more packets than what the receiver can retain. DCCP provides a scheme to access congestion control scheme without executing them at the application layer.

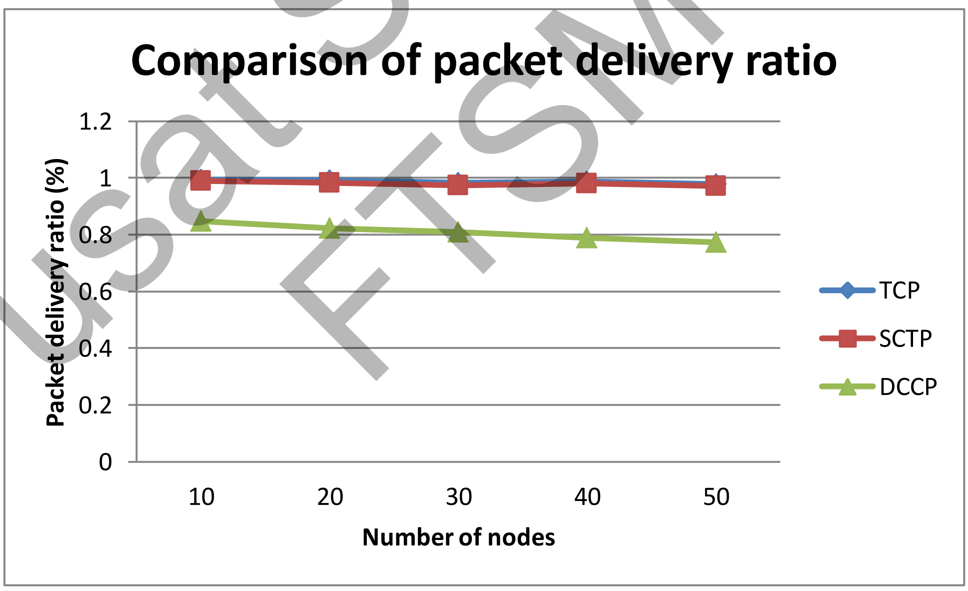

Figure 4.4 The comparison of PDR

4.3 DISCUSSION

The results of the network overall performance of the given scenario shows that DCCP protocol achieves the best average throughput even when the number of nodes increases from 10 to 50 nodes. However, TCP and SCTP protocols have a less average throughput when compared to DCCP. Moreover, TCP and SCTP protocols have the highest end to end delay time when compared to DCCP. While in the cases of packet delivery ratio and packet loss ratio, TCP protocol outperforms the highest ratio in compared with SCTP and DCCP protocols because the reliability that TCP provided. That`s means the connection in TCP started and waiting till the messages between the server and client be exchanged and checks if there is errors in the packet will send requests to the sender for retransmit the packet to receiver again.

4.4 SUMMARY

This chapter clarifies the final simulation`s results for the scenario that proposed through a different number of nodes. TCP, SCTP, and DCCP transport protocols are compared and analyzed by measuring the performance metrics like the average throughput, packet delivery ratio, packet loss ratio and end to end delay. Likewise, the chapter discusses the comparison of the results between the three protocols in order to explore the best transport protocol for video transmission.

CHAPTER V

CONCLUSION AND FUTURE WORKS

5.1 INTRODUCTION

This chapter completes the flow of methods on this thesis. The summary of the process have explained and determined in section 5.2 and the following section proposed the future works which can be implemented by extend the works for other researchers.

5.2 CONCLUSION

In this research, we have focused on performance of three transport protocols which are SCTP, DCCP, and TCP which are evaluated by using video transmission over wireless local area network environment. In order to determine the right transport protocol which can meet high quality of service requirements of video streaming and various of performance metrics are used like the average of throughput, packet delivery ratio, packet loss ratio and end-to-end delay to measure each protocol`s performance.

The comparison between three transport protocols in terms of the average throughput indicated that DCCP outperforms in terms of throughput. However, at the same time it has the worse protocol in terms of number of packet loss and packet delivery ratio compared to TCP and SCTP. When the number of nodes increased due to DCCP has two mechanisms built in. These mechanisms are the congestion control (controlling the number of packets that have sent to the network if the amount of packets are larger than the availability in the network`s capacity and this the main purpose of DCCP protocol which is control the congestion in the network), and the flow control (control the size of traffic once the sender sent up to the max limits, then the sender will receives respond from the receiver).

By evaluating the performance of the three protocols in terms of end to end delay, it is showed that the DCCP protocol has the best result compared to other protocols. TCP and SCTP protocols need more time to establish the connection between the communications because of the reliability feature that consumes more time at the beginning due to the server is still idle. For example, when the client sends the video to server, it will take more time to reply with acknowledgment message to the sender.

Based on the analysis of TCP, SCTP, and DCCP transport protocols for video transmission over wireless area network where packet loss is difficult to be controlled. The result shows DCCP is the right protocol to fulfill the quality of service requirement in terms of throughput, packet loss, packet delivery ratio and delay for efficient video transmission. The reason behind that is because DCCP gives good throughput and high packet loss in comparison with TCP and SCTP due it has congestion control scheme that aid the protocol to avoid the congestion in the network. These two schemes are TCP-Like Congestion Control that is useful for those applications which require the much possibility bandwidth in the network and which can adjust to the changes of congestion control window, and the second scheme is TCP-Friendly Rate Control which used for applications that require to transmission the data at constant rates, as the applications are sensitive to change abruptly in the transmission rates. These mechanisms are suitable for the applications where a stable rate of data transmission is obligatory instead of the reliability in order transfer of packets.

5.3 FUTURE WORK

With the issue of time constraint, still there are some issues can be extended for future works such as:

1) By evaluating some new transport protocols such as multiple path tranmission prototcol (MPTCP).

2) The work can be extended for a wider range of use other multimedia applications like voice over IP (VOIP) and real audio or real video.

3) The experiment can extend by implement the applications over different wireless network technologies such as wideband code division multiple access (WCDMA), LTE and High-Speed Downlink Packet Access (HSDPA) etc. The execpected result mght be perform better if using another technology than that implemeneted in this thises.

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