Research Journal of Applied Sciences, Engineering and Technology 11(10): 1048-1057, 2015
DOI: 10.19026/rjaset.11.2118
ISSN: 2040-7459; e-ISSN: 2040-7467

Submitted: May 1, 2015 Accepted: June 14, 2015 Published: December 05, 2015

Research Article

Performance Evaluation of TCP, UDP and DCCP Traffic Over 4G Network

[1, 2]Wisam Abduladheem Kamil, [1]Shahrudin Awang Nor and [1, 3]Raaid Alubady
[1]School of Computing, College of Arts and Sciences, University Utara Malaysia, 06010 Sintok, Kedah, Malaysia
[2]Computer Science Department, University of Thi-Qar, Thi-Qar,
[3]Computer Science Department, University of Babylon, Babylon, Iraq

Abstract: Fourth Generation (4G) mobile systems 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. This study tries to evaluate if the old protocols suit with this new technology. And which one has the best performance and which one has the greatest effect on throughput, delay 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 Simulation-3 (NS3), the performance of transporting 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 Evaluation (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, congestion control, DCCP, LTE, TCP, transport protocol, UDP

INTRODUCTION

The trend of 1G till 4G nowadays are the boiling connection over airwaves (Xue *et al*., 2014). The demand of the 4G has increased widely thought the most spread smart phone (Shukla and Khare, 2013), for example, I-phones and Samsung. Nevertheless, the performance of the multimedia stream will not completely fit the merit of end-user satisfaction. In the transport layer of the OSI model, User Datagram Protocol (UDP) and Transmission Control Protocol (TCP) are the most recommended and widely used protocols (Abeta, 2010). However, both of them have a few performance shortcomings. In UDP, the transmission is unreliable due to the lacking of acknowledgment for received data stream. In contrary, in TCP, the transmission is more reliable at the expense of the time cost. To achieve a kind of trade-off clogging control system with reasonable conveyance is needed. The existing transport protocols, e.g., UDP, TCP and DCCP (Datagram Congestion Control Protocol), do not propose a generic solution to the said dilemma. The functional drawbacks distributed among the above protocols is Lack of 4G performance, especially, when DCCP transport video such as MPEG-4 over transport layer (Varet and Larrieu, 2014). Therefore, in this study we will study the analysis and will compare the Internet protocols which are used for streaming video, such as MPEG-4 over LTE infrastructure technology to show the strength and weakness of DCCP, TCP and UDP by simulating it in the latest NS3- repository and gives the intensive results based on simulations each protocol spread.

The main scope of this research is to compare three important protocols - UDP, TPC and DCCP based on four metrics, i.e., Delay, Packet loss, Packet delivery ratio and throughput. After that we will design a separate three main scenarios for each protocol based on selected metrics with coverage. Then, we expect through our result to see that which protocol will perform better than other, especially, when there are payloads through MPEG-4 over slandered LTE stations.

WORK MOTIVATION

This research is significant because we can study the effectiveness of the 4G through video traffic stream, especially the delay and video transmission time. The performance of DCCP, TCP and UDP protocols will be

Corresponding Author: Wisam Abduladheem Kamil, School of Computing, College of Arts and Sciences, University Utara Malaysia, 06010 Sintok, Kedah, Malaysia

studied, supporting to determine which one is better in 4G environment. In fact, the vast majority of people nowadays have used smart phone and they are defiantly looking for 4G supportive devices. Then, they will search the performance of the video payload. Therefore, our study will evaluate the video traffic and will reveal that which protocol will be more useful for the 4G Smart phones and their applications. However, the vast studies emphasis its performance separately.

A payload video traffic would cause video latency, or even lost 4G signaling in some cases. Then, the advantages of MPEG4 are slightly reducing the traffic. But, what about the other holding protocols that MPEG4 went through 4G LTE. Therefore, we have evaluated UDP, TCP and DCCP protocols. Also we have provided a comparison study of all UDP, TCP and DCCP traffic over transport layer by sending MPEG4 video over 4G network.

LITERATURE REVIEW

Japan had invented the cellular communication system in late 1970s and it was the main paradigm in that time (Bamidele Moses, 2014). While 4G is the name to port generation of wireless technologies. Moreover, the mobile devices that the user used to communicate with each other like telephone calls, e-mails, Internet access and GPS signals are using these networks. These technologies are faster and have more mobility than the old wired network technologies (Shukla *et al*., 2014). However, in 1980s, a modern and faster analog telecommunication was brought for the wireless technologies at that time. Then, with the modernization of cellular network generations starting from 0G to 4G, have distributed widely for new telecommunications world. Nevertheless, the paradigm of mobile telephone services can be characterized as: Mobile Telephone Service, IMTS (Improved Mobile Telephone Service) AMTS (Advanced Mobile telephone System) (Rumney, 2013).

While, in the range of 28Kbit/s to 56Kbit/s it would be the speed performance for the 1G. However, from 2.9KB/s to 5.6KB/s. 2G is the actual standard downloads speed (Shukla and Khare, 2013). 3G technologies allow network operators to offer customers a wider variety of more cutting-edge facilities for attaining better network, where capacity can be improved via spectral efficiency. On the other hand, IEEE 802.11 (Wi-Fi and WLAN), 3G covers a wider area with high bit rate. The speed is up to 5.8Mbit/s in the uplink and 14.4Mbit/s in the downlink. UMTS (Universal Telecommunication System), CDMA 2000, W-CDMA (Wideband-CDMA), GSM EDGE (Mobile Enhanced Data Rates for GSM Evolution) and WIMAX (Worldwide Interoperability for Microwave Access) are main standards in 3G (Garg, 2000). However, 3.5G includes HSDPA (High-speed Downlink Packet Access) up to 8- 10Mbit/s in the downlink. 3.75G is HSUPA (High-speed Uplink Packet Access) up to 1.4Mbit/s in the uplink (Kolding *et al*., 2002).

Nonetheless, in 2006 it came out with the emergence of 3G. Then, after four years pre-4G system had come out (Dzebo and Mutapcic, 2013). Further, LTE has bright later three years (Long Term Evolution) and it has been more significant. Because of this the coverage of LTE is more friendly and established.

Fourth Generation blankets over billions of supporters overall or more than 80% of the worldwide versatile business sector (Dahlman *et al*., 2013). However, the number of worldwide subscribers, in 2008, utilizing High-Speed Packet Access (HSPA) networks surpassed 70 million (Khan, 2009). Then, HSPA is a 3G evolution of GSM that supports high-speed data transmission by means of WCDMA technology. While the global use of HSPA technologies among clients and businesses have accelerated, representing continuous traffic growth for high-speed mobile 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 the Internet traffic (Godwin-Jones, 2014).

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 (Abeta, 2010). However, the spectral efficacy target for the LTE scheme is three to four times more than the existing HSPA scheme (Shukla and Khare, 2013). 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 (Khan, 2009). For the wireless, broadband data speed transaction, Fig. 1 explained how the wireless data transfer grows from 384 kbps till LTE. Therefore, this project intends to use LTE for the

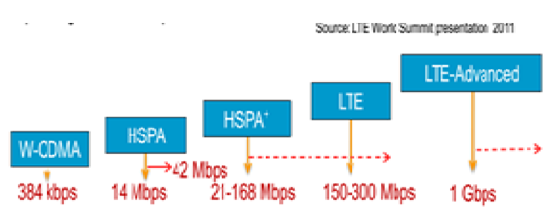

Fig. 1: The grown of the telecommunication

Table 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
Message-oriented	No	Yes	Yes

testing, explaining the performances over previous survived data (Ramli *et al*., 2014).

METHODOLOGY

Transport layer protocols for multimedia applications: The services and features of some transport layer protocols, i.e., UDP, TCP and DCCP are shown in Table 1, all of them have their own features and relevance for particular application under specific environments.

Transmission Control Protocol (TCP): TCP is another 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 (Verma and Dhawan, 2014). Whereas, this protocol also performs some management tasks, such as controlling rate and message during regulating traffic congestion and communication.

TCP acts as a transport layer that hides the underlying systems administration points of interest from correspondence provisions. One of the best cases of TCP applications is the web browser (Vetro *et al*., 2011). Then, other common main applications include, web server, e-mail and file transfer.

User Datagram Protocol (UDP), The UDP has structured by Postel (1980) and it considers the backbone for the Internet Protocol Suite (IPS) (Alferness *et al*., 1997). 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 (Zheng and Boyce, 2001). 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. So it is much suited for small applications which do not need reliable connection. The most common use of UDP is in 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 DHCP (Lemon *et al*., 2002), RIP (Hedrick, 1988) and VoIP (Goode, 2002) etc.

Nevertheless, Time-sensitive and Real-time applications, for example, video traffic and voice, are using UDP due to the dropping packets, which 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 (Edelman *et al*., 2007).

Datagram Congestion Control Protocol (DCCP): The DCCP is a convention of the transport layer with dependable association setup, blockage control and characteristic transaction competence (Kohler *et al*., 2006). 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, for example, streaming media which can be assisted from manipulation over the adjustments between delay and reliability 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 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 agreement. Several advantages of a DCCP association are coordinated by characteristics, for example, congestion control mechanism in use on 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. 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.

Simulation setup and metrics: We have used Linux Ubuntu 12.04 as operating system, because the Network Simulation 3 (NS3) www.nsnam.org,

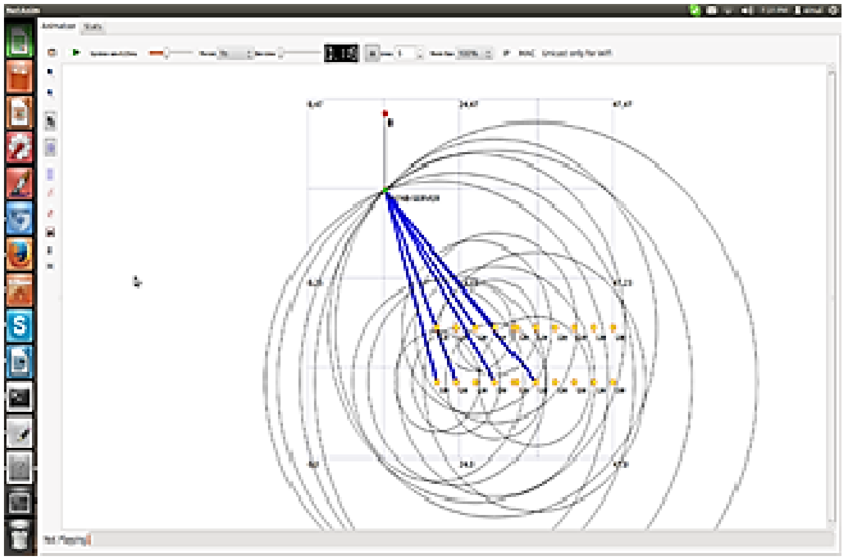

Fig. 2: Transmission between eNB server and ue nodes

(Henderson *et al*., 2006) works with high efficiency in Linux environment than other operating system. And the hardware computer CPU is core I5 with memory size 4 Gigabyte. Our scenario is connected to a number of nodes Ue with one base station eNB. Then the base station connect to server node as point to point. Just to read the network performance from server node. Besides, we have implemented visualization graph for that scenario using NetAnim tool as shown in Fig. 2.

The main traffic metric that is used for the LTE network by using UDP, TCP and DCCP protocols are:

Throughput: Defines the rate of 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 (Chughtai *et al*., 2009). Throughput is usually measured either bits per second (bit/s or bps), or data packets per second (p/s or pps). It refers to the performance of network, as shown in the Eq. (1):

$$\text{Throughput} = \frac{\text{Numberof ReceivedPackets}}{\text{LastpacketsentTime} - \text{FirstpacketsentTime}} \quad (1)$$

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 is happened. 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 (Chughtai *et al*., 2009; Alubady *et al*., 2015) Eq. (2):

$$\text{Packetloss} = \sum \text{packets send} - \sum \text{packets received} \quad (2)$$

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 (Alubady *et al*., 2015). It means that the total number of arrived packets is divided by the total number of sender packets. See Eq. (3):

$$\text{PDR} = \frac{\sum \text{Total number of Received Packets}}{\sum \text{Total number of Send Packets}} \quad (3)$$

Delay: This matric is also important to check network performance. Let explain how by instance, with a live audio stream, it is far 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 do not care about the real time delivery. Such a network might use control protocol for congestion management, adding even more complexity, as a consequence give more delay (Chughtai *et al*., 2009). Delay is the time faced by a packet to move or travel across the network from one node to another. See the Eq. (4):

$$\text{Delay} = \text{Tr} - \text{Ts} \quad (4)$$

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 Eq. (5):

$$\text{Mean delay} = \frac{\text{TotalDelay}}{N} \quad (5)$$

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

EXPERIMENTAL AND RESULTS WITH DISCUSSION

In this research, we show the results of TCP, UDP and DCCP protocol using NS3. NS-3 supports a graphical tool gnuplot. All the graphs are generated by gnuplot to show the results of NS-3 simulation for each protocol. We give the graphical analysis of the protocol performance metrics like delay, throughput, Packet Delivery Ratio and packet loss. We have shown previously that the network topology consists of three parts. The mobile unit call (Ue) 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 Ue, 20 Ue and 30Ue connect directly to one eNB and the eNB connected to server node.

Network performance measurement: Most importantly, through our research, we have found that there are two research questions here. Why the number of nodes (Ue) 10, 20 and 30 Ue specifically. And what is the effect of the different distance between nodes (Ue) and base station (eNB). These are so important questions that the researcher must be concerned about them, especially when design the topology and write code. Therefore to be more fairly, we have implemented easy and dynamic C++ code that could help us to find the answers for those questions. So inside the code we have changed the number of nodes for several different numbers. We have taken (5 nodes, 8 nodes, 10 nodes 16 nodes... etcetera) as a different program running scenario in the end we have got different results. After that we have studied there are different results by analysis and compare. Then we have ignored all the similarity result from our research. Therefore we have chosen the (10 20 and 30) nodes. We have implemented also different scenarios based on different distances.

The second question regarding the impact of the distance on the network performance. The distance from Ue to eNB is (50, 100, 150 and 250 m). But we have discovered that different distance does not affect so much of the network performance. That is why we ignored.

Comparison analysis for throughput: One of the main solutions to improve the performance of the new wireless communication systems is by improving the protocols used over these networks such as UDP, TCP and DCCP protocols. This section investigates the comparative performance of UDP, TCP and DCCP 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, perhaps the delivery over a physical or logical link. The measurement unit of throughput is usually either bits per second (bit/s or bps), or data packets per second (p/s or pps).

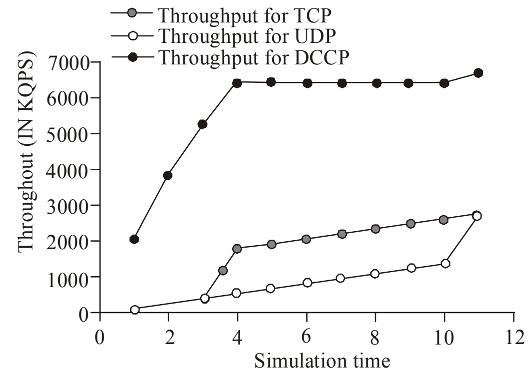

Fig. 3: Throughput of TCP/UDP/DCCP for 10 nodes

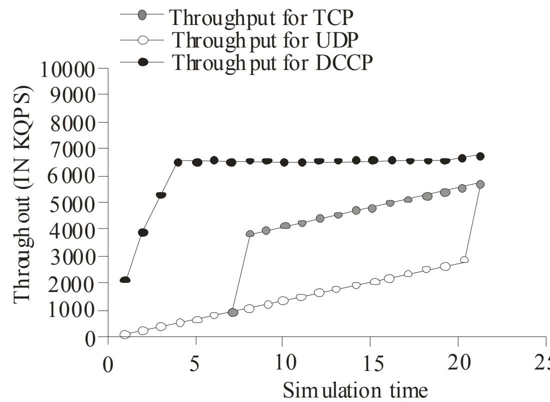

Fig. 4: Throughput of TCP/UDP/DCCP for 20 nodes

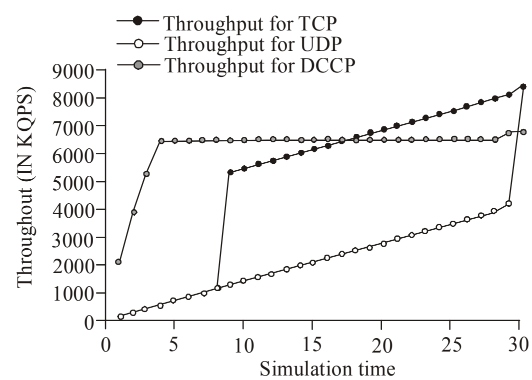

Fig. 5: Throughput of TCP/UDP/DCCP for 30 nodes

Figure 3 shows that the DCCP protocol has a good throughput in the environment of the LTE network of 10 nodes. The scenario here supposes all 10 nodes send file video at a same time to the server. If we increase the number of nodes in the LTE network, for instance, let suppose 20 nodes, then still the DCCP protocol is better than other protocols as the results showed in Fig. 4. All these results are taken from NS3 simulator, which is already valid and we have made the node number as 30 nodes. The result is shown in Fig. 5.

Furthermore, here as the number of nodes increases the throughput of complete network will get 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 too high. From the total throughput result, as the number of nodes increases the throughput gets double approximately from 2805.15 to 5593.25 Kbps.

The DCCP protocol has good 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 DCCP provides less delay. DCCP supports delay-sensitive streaming over UDP without TCP's delay inducing reliability. Moreover, the TCP protocol is suitable for wire connection not adaptive or designed to work in the wireless environment. Therefore the TCP's disadvantage protocol has been overcome by new protocol (DCCP) which is adaptive and design for wireless environment.

Comparison analysis of delay: Delay is one of the important metrics to check network performance. Before we proceed further, let us explain how by instance, with a live voice stream, it is more vita to send recent packets faster than to assure that stale packets are finally sent. 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 Fig. 6.

The DCCP protocol has 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.

Figure 6 and 7 show the result for 10 nodes and 20 nodes, respectively. We see the result is a 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 good result with 20 node scenario.

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. 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

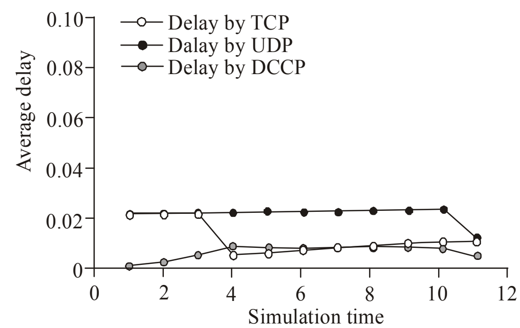

Fig. 6: Average delay of TCP/UDP/DCCP for 10 nodes

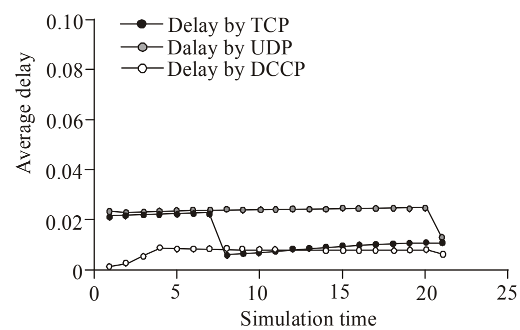

Fig. 7: Average delay of TCP/UDP/DCCP for 20 nodes

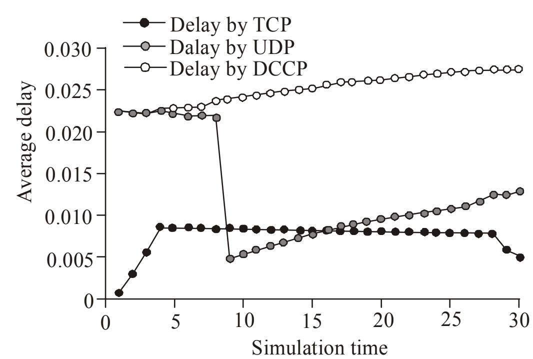

Fig. 8: Average delay of TCP/UDP/DCCP for 30 nodes

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 that the DCCP protocol is the best protocol regarding to delay time.

Figure 8 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.

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

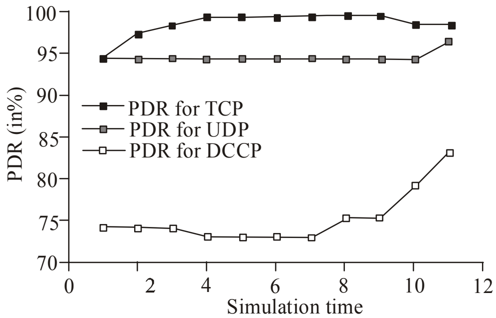

Fig. 9: PDR of TCP/UDP/DCCP protocol for 10 nodes

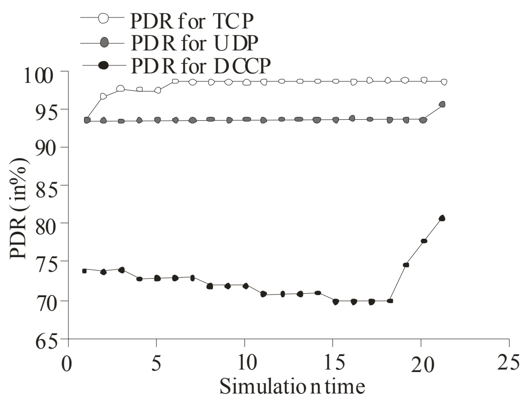

Fig. 10: PDR of TCP/UDP/DCCP protocol for 20 nodes

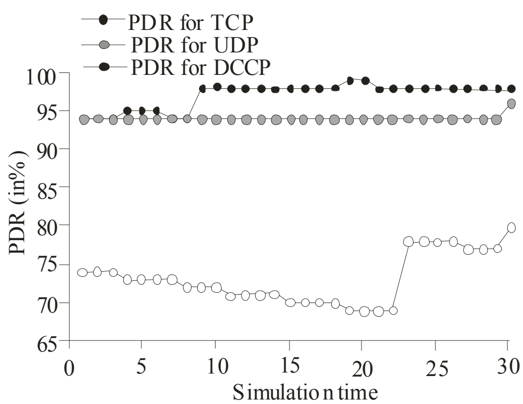

Fig. 11: PDR of TCP/UDP/DCCP protocol for 30 nodes

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 maximum 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 and eNB base station is low. Packet Delivery Ratio for TCP socket varies, i.e.,

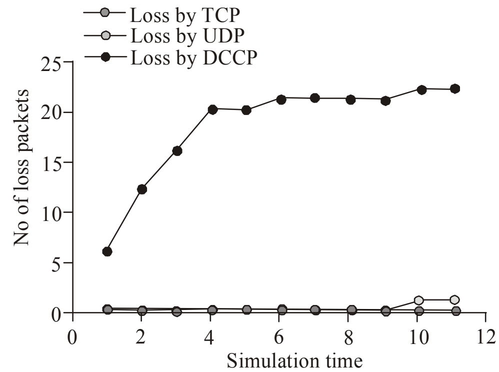

Fig. 12: Packet loss of TCP/UDP/DCCP for 10 nodes

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 good Packet Delivery Ratio.

Also, the result shows the UDP protocol has good Packet Delivery Ratio if the number of nodes is 10, as shown in Fig. 9. 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. Figure 10 and 11 do not have much difference from the Fig. 9, which is already discussed above; therefore, no need to further explain it.

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 Fig. 12. That TCP protocol has good 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 increase the nodes to 20 nodes. But we have to explain Fig. 13. 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 Fig. 14 that through the time is running the amount of loss packets

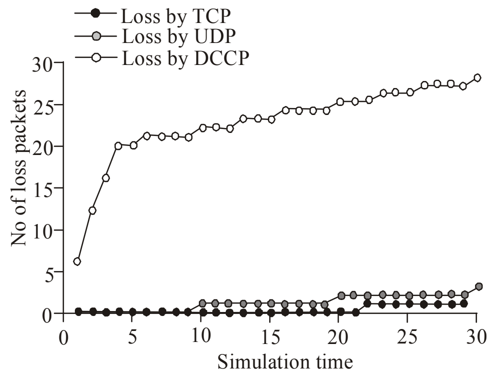

Fig. 13: Packet loss of TCP/UDP/DCCP for 20 nodes

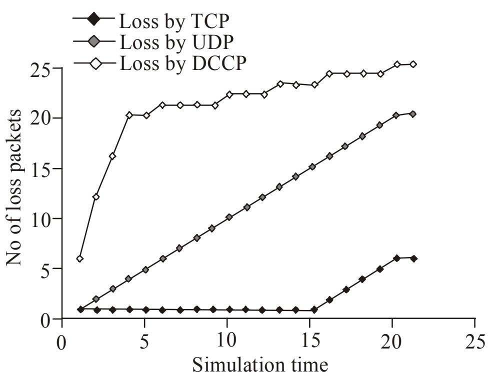

Fig. 14: Packet loss of TCP/UDP/DCCP for 30 nodes

```
Config::SetDefault ("NS3::UdpClient::MaxPackets",
UintegerValue (125000));
Config::SetDefault ("NS3::LteMacQueue::MaxSize",
StringValue ("1024"));
Config::SetDefault ("NS3::DropTailQueue::MaxPackets",
UintegerValue (10));
```

Fig. 15: Packet loss of TCP/UDP/DCCP for 30 nodes

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.

Comparative analysis of TCP, UDP and DCCP: DCCP offers a method to overcome network load by congestion control methods if the sender delivers more packets than the receiver can keep. 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 applications consist of multiplayer online games, streaming media and Internet telephony. 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 NS3 gives us facilities to make that in easy way. Figure 15 presents the NS3 script for transferring the video streaming file sized (128 MB).

Our result shows that the DCCP protocol has a good throughput when the number of Ue becomes 10 & 20. But the UDP and TCP protocol have less throughput if compare with DCCP. The difference 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 Ue becomes 30. That's because the default maximum number in the LTE is 22 Ue. We brock this exception to see what is the result. 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 (Table 2).

As we saw 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, 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.

Table 2: UDP/TCP/DCCP protocols based LTE environment with 10, 20, 30 nodes

	Throughput (in Kbps)			Packet loss		
Protocol	10Ue	20Ue	30Ue	10Ue	20Ue	30Ue
UDP	2762.19	5524.38	8286.56	3%	3%	3%
TCP	2805.16	5593.25	8397.78	1%	1%	1%
DCCP	6699.34	6715.28	6731.22	16%	17%	19%
	Packet delivery ratio			Delay		
Protocol	10Ue	20Ue	30Ue	10Ue	20Ue	30Ue
UDP	96%	96%	96%	0.01243390	0.01341040	0.01463610
TCP	98%	98%	98%	0.01139420	0.01246730	0.01106310
DCCP	83%	82%	80%	0.00522303	0.00511127	0.00500419

CONCLUSION

In this research we analyzed the performance of transport layer protocols on LTE network. As the capabilities of network layer changes with high potential of network, our concludes the performance at transport layer by analyzed the TCP, UDP and DCCP protocols, on various performance metrics like delay, throughput, packet delivery ratio and packet loss. As our simulation results on network simulator, DCCP protocol outperforms the other conventional connection-oriented and connection-less protocols in delay and throughput. While due to its connection oriented architecture, TCP give maximum packet delivery ratio and minimum packet loss count. So for the applications where we can't handle packet loss, we must go for TCP else DCCP is best suited for real time applications with good throughput. We have applied the scenarios to send traffic video stream with size 150 MB by using DCCP, TCP and UDP protocols. The performance metrics of bandwidth throughput, packet loss and delay will be used to set the benchmark of the 4G network performance. In future DCCP can also be improved to reduce the packet loss and also to be suited for the applications which are very critical to packet-loss.

REFERENCES

Abeta, S., 2010. Toward LTE commercial launch and future plan for LTE enhancements (LTE-advanced). Proceeding of the IEEE International Conference on Communication Systems (ICCS, 2010), pp: 146-150.

Alferness, M.H., P.B. Criswell, D.R. Johnson and J.R. McBreen, 1997. Method for Generating an Internet Protocol Suite Checksum in a Single Macro Instruction. Publication No. US5701316 A, Google Patents.

Alubady, R., M. Al-Samman, A. Habbal, S. Hassan and S. Arif, 2015. Performance analysis of reactive and proactive routing protocols in MANET. J. Eng. Appl. Sci., 10(3): 1-11.

Bamidele Moses, K., 2014. Mobile communication evolution. Int. J. Mod. Educ. Comput. Sci., 6(1): 25-33.

Chughtai, H., S. Malik and M. Yousif, 2009. Performance evaluation of transport layer protocols for video traffic over WiMAX. Proceeding of the IEEE 13th International Multitopic Conference (INMIC, 2009), pp: 1-6.

Dahlman, E., S. Parkvall and J. Skold, 2013. 4G: LTE/LTE-Advanced for Mobile Broadband. 2nd Edn., Academic Press, pp: 544, ISBN: 9780124199859, Safari Book.

Dzebo, A. and H. Mutapcic, 2013. Regulatory aspects of mobile broadband technologies. Proceeding of the 36th International Convention on Information and Communication Technology Electronics and Microelectronics (MIPRO).

Edelman, B., J. Gay, S. Lozben and P. Shetty, 2007. Real-time Priority-based Media Communication. Google Patents, Publication No. US 7272658 B1.

Garg, V.K., 2000. IS-95 CDMA and CDMA 2000-Cellular/PCS Systems Implementation. Prentice Hall PTR, Upper Saddle River, NJ.

Godwin-Jones, R., 2014. Emerging technologies. Int. J. Comput. Sci. Bus. Inform., 9(1): 66-79.

Goode, B., 2002. Voice over Internet Protocol (VoIP). P. IEEE, 90(9): 1495-1517.

Hedrick, C.L., 1988. Routing Information Protocol. RFC Editor, United States.

Henderson, T.R., S. Roy, S. Floyd and G.F. Riley, 2006. NS-3 project goals. Proceeding of the Workshop on NS-2: the IP Network Simulator, pp: 1-13.

Khan, F., 2009. LTE for 4G Mobile Broadband: Air Interface Technologies and Performance. Cambridge University Press, Cambridge, New York.

Kohler, E., M. Handley and S. Floyd, 2006. Datagram Congestion Control Protocol (DCCP). RFC 4340 (Proposed Standard). Retrieved from: https://tools.ietf.org/html/rfc4340.

Kolding, T., F. Frederiksen and P. Mogensen, 2002. Performance aspects of WCDMA systems with High Speed Downlink Packet Access (HSDPA). Proceeding of the IEEE 56th Vehicular Technology Conference (VTC, 2002-Fall), pp: 477-481.

Lemon, T., S. Cheshire and B. Volz, 2002. The Classless Static Route Option for Dynamic Host Configuration Protocol (DHCP): Version 4. Network Working Group. RFC 3442, (Proposed Standard).

Postel, J., 1980. User datagram protocol. Technical Report RFC 768, Information Sciences Institute.

Ramli, H., K. Sandrasegaran, A. Ismail, S. Latif and I. Farah, 2014. A simulation tool for downlink long term evolution-advanced. Res. J. Appl. Sci. Eng. Technol., 8(19): 2032-2041.

Rumney, M., 2013. LTE and the Evolution to 4G Wireless: Design and Measurement Challenges. John Wiley and Sons, New York.

Shukla, S. and V. Khare, 2013. Comparative study of 1G, 2G, 3G and 4G. J. Eng. Comput. Appl. Sci., 2(4): 55-63.

Shukla, S., C. Shingau, A.S.W. Tam, A. Gupta, X. Yang and H.J. Chao, 2014. TCP PLATO: Packet labelling to alleviate time-out. IEEE J. Sel. Area Comm., 32(1): 65-76.

Varet, A. and N. Larrieu, 2014. Realistic network traffic profile generation: Theory and practice. Stud. Comp. Intell., 7(2): 1-16.

Verma, A.V. and S. Dhawan, 2014. Comparison and assessment of TCP and UDP protocols in different network scenarios. Int. J. Softw. Hardw. Res. Eng., 2(2): 2-5.

Vetro, A., T. Wiegand and G. Sullivan, 2011. Overview of the stereo and multiview video coding extensions of the H. 264/MPEG-4 AVC standard. P. IEEE, 99(4): 626-642.

Xue, W., S. Ramamurthy and J.D. Cole, 2014. Data Session Continuity between Wireless Networks. US 8639233 B1, Google Patents.

Zheng, H. and J. Boyce, 2001. An improved UDP protocol for video transmission over internet-to-wireless networks. IEEE T. Multimedia, 3(3): 356-365.