Voice Transmission over Transport Layer

Pushpender Sarao

Abstract: Transport layer having the big responsibilities in network models. It plays a vital role during process-to-process delivery. To perform process-to-process delivery, several network protocols are used in transport layer. TCP (Transmission Control Protocol), UDP (User Datagram Protocol), SCTP (Stream Control Protocol) are the protocols that support for process-to-process delivery. In this paper, we have analyzed the transport layer protocols in various aspects. Because some network protocols are reliable while other is unreliable. Identifying best suitable protocol and service is required. For performance analysis, several parameters like throughput, jitter, latency, delay, end-to-end delay has been used. Here, we are trying out to find out the suitable protocol for different type of data transmission in several different network-scenarios. To ensure the congestion control and to reduce the end-to-end delay, this analysis work will be very useful. This evaluation work will help for identifying and enhancing the existing transport protocols and algorithms. All transport protocols have simulated and evaluated over network simulator and graphical presentation is performed with the help of JTarana, TraceGraph202, and Xgraph.

Index Terms: Jitter, End2End delay, packet size, throughput, packet lost, multi-homing, multi-stream;

I. INTRODUCTION

Transport layer is very important layer of the OSI reference model. In this layer, so many protocols are working to complete the functionality and services provided by this layer. Transmission control protocol (TCP), user data gram protocol (UDP), stream control transmission protocol (SCTP), TCP friendly rate control (TFRC), and data gram congestion control protocol (DCCP). Generally, for short term communication, UDP protocol is used; but when transmission data and time is long, TCP and SCTP transport protocols are used. Actually, it depends on the network scenario, data type, and size of data. Congestion control and packet lost is big issue here, when transmitting the data in transport layer. SCTP and TCP protocols are connection oriented protocol i.e. connection establishment is mandatory before transmitting the data. in case of UDP, guarantee of data delivery and ordering of data is not there. For sending short messages, UDP protocol is suitable. Overall, services provided by UDP are unreliable. SCTP provides 32-bit end-to-end checksum facility. Services provided by SCTP and UDP are full duplex. Ordered and unordered data delivery is possible in SCTP transport protocol. SCTP provides flow control, congestion control, ECN capable, selective acknowledgements, and path MTU discovery facilities; while on the other hand these facilities are not provided by the UDP protocol in transport layer. Preservation of message boundaries is possible in both protocols i.e. SCTP and UDP protocols. Pseudo-header for checksum facility is available in UDP, but not in SCTP protocol.

TCP[13]:

- ✓ When TCP is used with HTTP, port number 80 is used while port number 20 and 21 are used in case of FTP for data and connection control respectively.
- ✓ As TCP, UDP, and SCTP are process-to-process (program-to-program) delivery protocols, port numbers are used for ensuring different services.
- ✓ Process-to-process communication, stream delivery service, sending and receiving buffers, segments, full-duplex communication, connection-oriented, reliable services are provided by TCP protocol.
- ✓ TCP having same features like number system, byte number, sequence number, acknowledgement number, flow control, error control, congestion control.
- ✓ In case of TCP, a packet is known as segment.
- ✓ In TCP segment, source port address and destination port number address, both having 16 bits size while 32 bits size is reserved for sequence number and acknowledgement for each.

UDP[13]:

- ✓ In case of UDP, packets are known as user datagrams.
- ✓ User datagram having a header of 8 bytes(fixed size).
- ✓ In header of user datagram of UDP protocol, Source port number address, destination port number address, total length, check sum, each having 16 bits size.
- ✓ Port number 161 and 162 are used when UDP works with SNMP, while port number 111 is used in case of RPC.
- ✓ Encapsulation and de-capsulation process on message is performed to send a message from one process to another.
- ✓ UDP is suitable for simple request-response communication, internal flow control, internal error control, multi-casting, process management, route uploading.

SCTP[13]:

- ✓ SCTP and UDP are message-oriented protocols while TCP is byte-oriented protocol.
- ✓ Best features of TCP and UDP protocols are included in SCTP protocol.
- ✓ Process-to-process communication, multiple streams,

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multi-homing, full-duplex communication, connection-oriented services, reliable service, are the services provided by SCTP protocol.

- ✓ SCTP uses different number of ports depends on the type of SCTP application (for example port number 9990 with IUA SCTP application).
- ✓ The best features of SCTP are: transmission sequence number, stream identifier, stream sequence number, packets, acknowledgement number, flow control, error control, and congestion control.
- ✓ Source port address and destination port address each having 16 bits size while check sum and verification tag having 32 bits.

II. RELATED WORK

Adnan Ghani [2016] [5]et al has evaluated the TCP and DCCP protocols in MANET. To overcome the drawback of UDP in respect of traffic management, DCCP protocol was used. Like TCP, and SCTP, DCCP is not a reliable protocol. To ensure the smoothly multimedia application transmission over the network, TFRC protocol is mostly used. Performance of TFRC is not well in multi-hop networks. Performance metrics like throughput, end-to-end delay, average throughput and average end-to-end delay were used. A comparative study was carried out for DCCP, TCP, and TFRC protocols. DCCP was declared as best protocol as compared to TCP. Overall throughput of TCP was better due to its reliability property in network environment. TFRC is not a complete protocol, but works like a congestion control mechanism for transport layer [RFC 3448].

Ali Hussein Wheeb[1] has worked for evaluating the UDP, DCCP, SCTP and TFRC protocols. Data traffic, video traffic, and VoIP traffic scenarios were used for simulating and evaluating of protocols in wired network. CBR traffic was generated in 50 seconds simulation time for 4 network nodes. Throughput, average throughput, End to end delay, and packet loss ratio were the performance parameters used for evaluation in NS 2.35 network simulator. It was declared that SCTP outperforms as compared to TFRC, UDP, and TCP protocols. For video traffic, TFRC works well. End to end delay of DCCP is very less for data transmission. UDP works better for video traffic transmission.

Pallavi Gsngurde et al [2] have compared SCTP, TCP, and UDP protocols for video over IP (VoIP). Network simulator NS 2.35 for SCTP protocol was used for simulating the protocols in different network scenarios. Delay, average delay, throughput, number of packet send, number of packet received, packet lost performance metrics were used for comparing the protocols. TCP performs better in respect of packet lost as compared to SCTP, UDP protocols. Due to multi-homing and multi-streaming features, SCTP works well as compared to TCP protocol.

Ravonimanantosa[2016] [3]et al have analysed the TCP and UDP transport protocol with voice over IP(VoIP). TCP and UDP protocols were simulated in NS 2.35 scenario. Packet size 128, 1040, and 40 octets was decided for UDP, TCP, and ACK respectively. End-to-end delay, jitter, flow rate, packet loss were considered as the performance parameters. It was claimed that TCP is not compatible with VoIP. VoIP works well with UDP in respect of reliability. For multimedia application transmission, TCP is not so compatible.

sielayGebru et al [4] evaluated and compared the SCTP and UDP protocols over VoIP. Services and features of both protocols have been elaborated in detail. For simulating the protocols, NS2.35 was used. Performance evaluation work was carried out by considering the throughput, instant throughput packet loss, latency, and jitter metrics in wired network environment. Jitter and the latency values were variable in case of SCTP. It was concluded that SCTP outperforms than UDP in respect of packet loss and throughput. SCTP does not perform well in respect of jitter and latency.

Shiferaw Hermela et al[7] proposed a modified version of existing SCTP transport protocol in transport layer of OSI/TCP network model. To overcome the drawback of SCTP protocol, it was modified in respect of jitter, latency, and delay. The main aim for enhancing the SCTP was to enhance this over wired VoIP (voice over IP). The simulation work and performance work was implemented over NS-2 simulator. Detailed working architecture of SCTP was discussed in detail. Jitter performance with respect to packets was graphically presented and discussed. Packet wise and bitwise performance was evaluated. It was claimed that modified version of SCTP outperforms as compared to existing SCTP protocol in respect of delay, jitter, and latency.

III. PERFORMANCE ANALYSIS OF TRANSPORT LAYER PROTOCOLS

We have used network simulator 2.35(NS-2.35) [14] for simulating the TCP, UDP, SCTP, DCCP, and TFRC protocols. NS-2.35 is very popular simulator which is mostly used to experiment the wired and wireless network environment in several different scenarios. Besides the network simulator, we have applied different tools like JTarana, NSG2, NSWireless, Tracegraph202, GNUPlots, XGraph. We wrote five trace files for TCP, UDP, SCTP, TFRC, and DCCP protocols. We decided following parameters for simulation purposes, as shown in table 1:

Table 1: simulation parameters

Parameters	description
Protocols	TCP, UDP, SCTP, TFRC, DCCP
Simulation time	10s
Simulator	NS 2.35
Number of Nodes	2
Colour for nodes	Yellow, green
Label on node1	Voice1
Label on node2	Voice2
Traffic type	CBR
Packet size	128
Interval	0.020
Link type	Duplex link
Capacity	256 Kb
Propagation delay	50 ms
Connection	Two way VoIP connection
Queue size	50
Queue type	Drop tail

Metrics used for Performance analysis:

The following metrics were used for the performance analysis of the transport routing protocols:

a) Throughput of sending bits Vs average simulation jitter
b) Throughput of generating packets

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c) Throughput of receiving packets
d) Jitter of all the receiving packets Vs sequence number
e) Packet size Vs maximum throughput of generating packets
f) Packet size Vs average throughput of receiving packets
g) Throughput receiving bits Vs minimal simulation End2End delays
h) Throughput of receiving bits Vs average simulation jitter
i) Throughput of forwarding bits Vs average simulation processing time
j) Packet size Vs average throughput of receiving packets
k) Packet size Vs average simulation E2E delay
l) Cumulative sum of all the received bytes
m) Delay Vs simulation time
n) Packet IDs Vs delay
o) Average delay
p) Instant delay
q) Instantaneous jitter
r) Instantaneous throughput
s) Simulation time Vs jitter
t) Packet IDs Vs jitter
u) Latency
v) Packet IDs Vs End2End delay
w) Traffic Vs simulation time
x) Throughput
y) Generated packet IDs Vs simulation time
z) Packet size Vs delay
aa) Throughput of receiving bits
bb) Generated packets IDs Vs simulation time

We executed tcl files one by one in network simulator. As a result trace and nam files were generated. We wrote awk scripts for getting metrics values for each transport protocol. Xgraph, tracegraph and gnuplot tools were used for creating graphs with respect to network parameters like throughput, delay and jitter. Total simulation time was decided only for 10 seconds. All network environments were created as a wired network with only two nodes.

IV. RESULTS AND DISCUSSION

When we executed the tcl files, we got several values for network traffics generated in different network scenarios for different transport protocols as shown in table 2.

Table 2: Generated network traffic details

Parameters	SCTP	TCP	UDP
Simulation length in seconds:	10.397875	10.38875	9.954
Number of nodes:	2	2	2
Number of sending nodes:	2	2	2
Number of receiving nodes:	2	2	2
Number of generated packets:	204	674	992
Number of sent packets:	204	638	992
Number of received packets:	202	634	992
Minimal generated packet size:	36	40	128
Maximal generated packet size:	1472	1040	128
Average generated packet size:	739.0	1037.0	128.0
Number of generated Bytes:	150808	698960	126976
Number of sent Bytes:	150808	661520	126976
Number of received Bytes:	150712	657360	126976

With the help of trace files and awk scripts, we got results for average delay, as shown in table 3 and graphical representation in figure 1.

Table 3: average delay

Average delay[ms]	
TCP	611.108
UDP	54
SCTP	81.9691
TFRC	82.25
DCCP	52.75

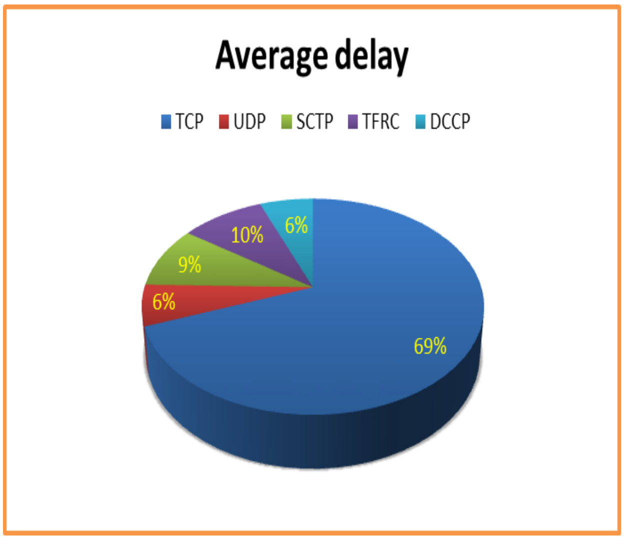

Figure 1: Average delays for TCP, UDP, SCTP, DCCP, TFRC

We have analysed that DCCP and UDP have lowest average delay while TCP have highest (as shown in **figure 1**). UDP is unreliable protocol while SCTP supports for multi-homing and multi-streaming characteristics in different network scenarios.

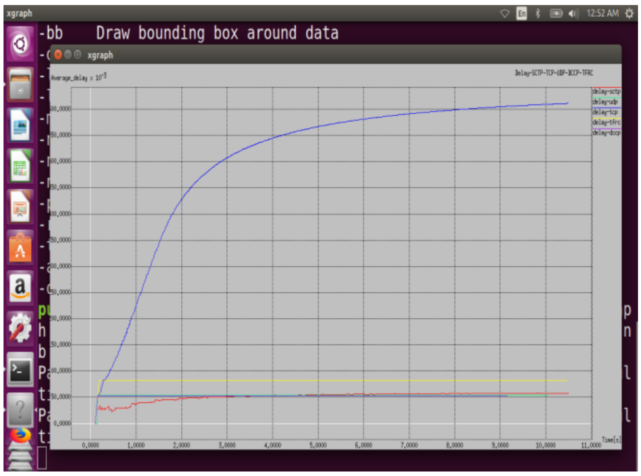

Figure 2: Average delay Vs Network simulation time

Average delay: Average delay for TCP is increased with respect to the simulation time. Overall average delay for TCP is highest as compared to SCTP, UDP, DCCP, and TFRC network protocols (**figure 2**). Average delay of UDP, TFRC, and DCCP is approximately same as the simulation time is increased. But, it is totally different for SCTP network protocol. Average delay fluctuates at the low simulation time. When simulation time is increased, fluctuation is reduced and overall average delay is little bit high as compared to UDP and DCCP network protocols. SCTP outperforms for small time data transmission as well as for long time data transmission.

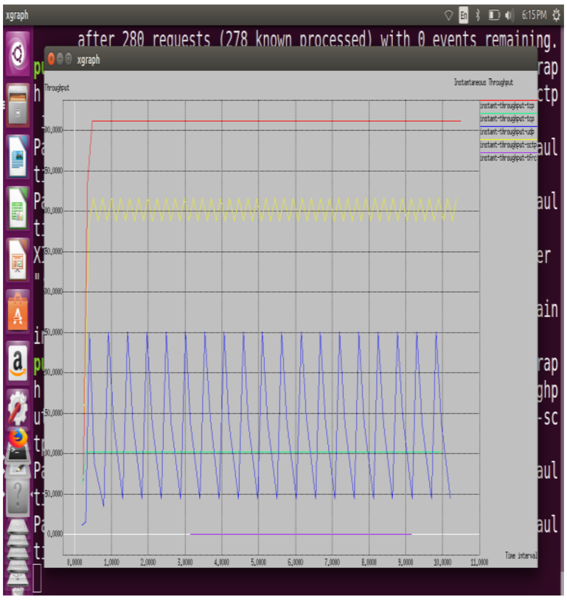

Figure 3: Instant throughput Vs Variant time

Instant throughput for TCP is highest as compared to DCCP, UDP, TFRC, and SCTP (**figure 3**). It remains mostly same with respect to variant times. Instant throughput for SCTP and UDP is fluctuated with respect of variant time. This fluctuation is high for SCTP. For SCTP instant throughput is lowest as compared to SCTP, TCP, DCCP and UDP network protocols. It remains approximately constant for DCCP and RFTC protocols. At initial stages of variant timings, instant throughput for TCP and SCTP are approximately same.

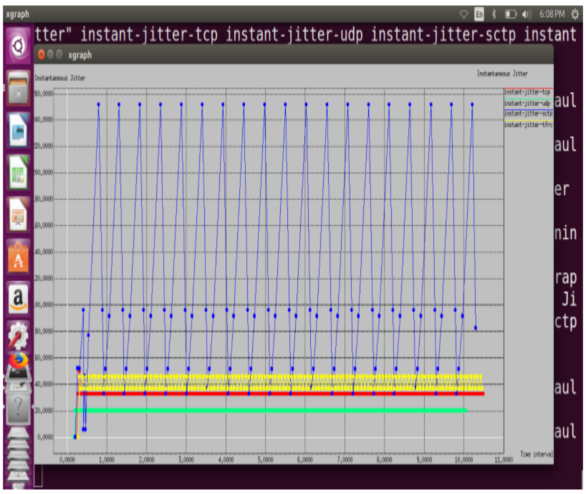

Figure 4: Comparison for Instant Jitter Vs Variant time

Performance analyses of network protocols-variant time w.r.t. instant jitter: instant jitter for SCTP is highest as compare to TCP, UDP, and TFRC network protocols **(figure 4**). It also highly fluctuated with respect to the variant timings. For UDP network protocol, instant jitter is very low as compare to TCP, SCTP, and TFRC protocols. Instant jitter remains approximately same for TFRC, TCP and UDP protocols with respect to the variant timings.

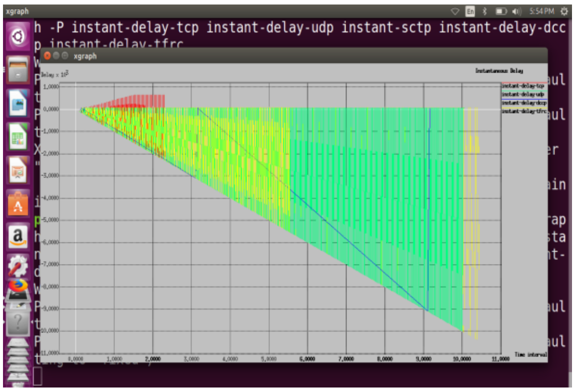

Figure 5: Comparison w.r.t. Instant delay Vs Variant time

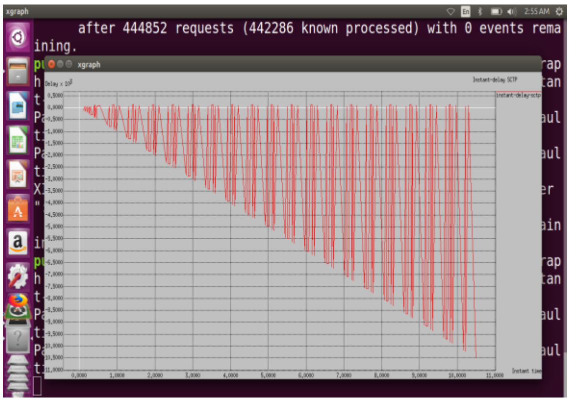

Figure 6: Instant delay Vs Variant time for SCTP

Figure 7: Instant delay Vs Variant time for DCCP

Instant delay: instant delay for DCCP is highly variable with respect to variant time. It is high at initial timings, but suddenly goes down and up. Again at higher stages of variant timings, it goes down and again up. For low to high variant timings, instant delay is at highest position. In case of SCTP network protocol, instant delay fluctuates up and down with respect to variant timings. Instant delay is increased with respect to variant timings. Fluctuations of instant delay are highly increased as the

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value of variant time is increased. For TCP network protocol, instant delay is low to high at lower variant timings. But, it is approximately remains same at specific network conditions at higher variant time. Instant delay for TCP is totally different as compared to SCTP, UDP, and TFRC network protocols **(figure 5-7).**

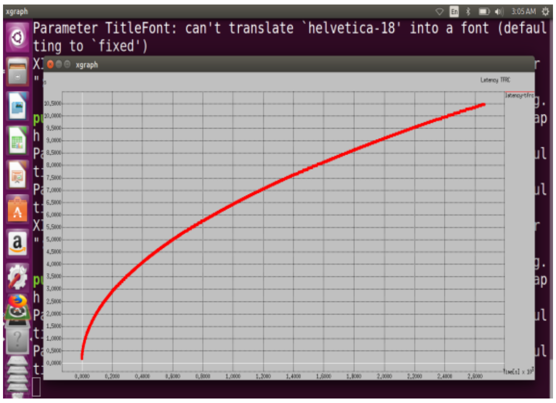

Figure 8: Latency for TFRC

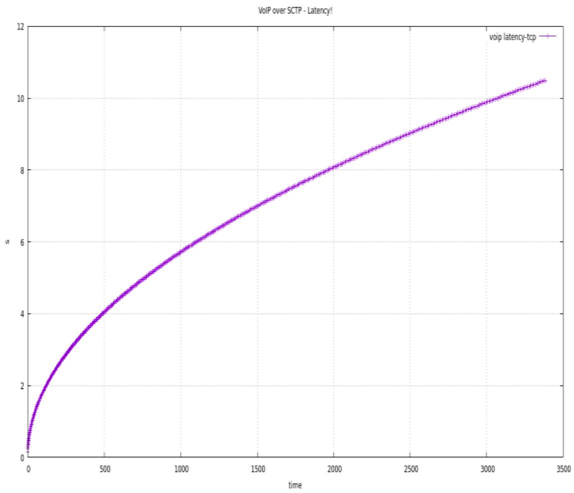

Figure 9: Latency for TCP

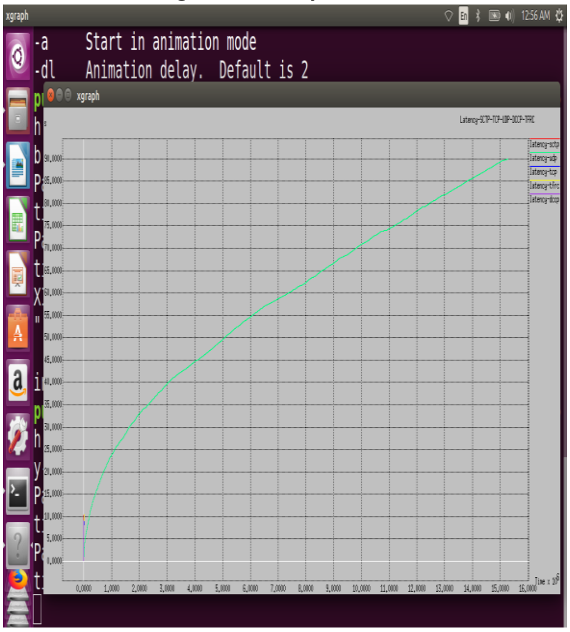

Figure 10: Latency comparison

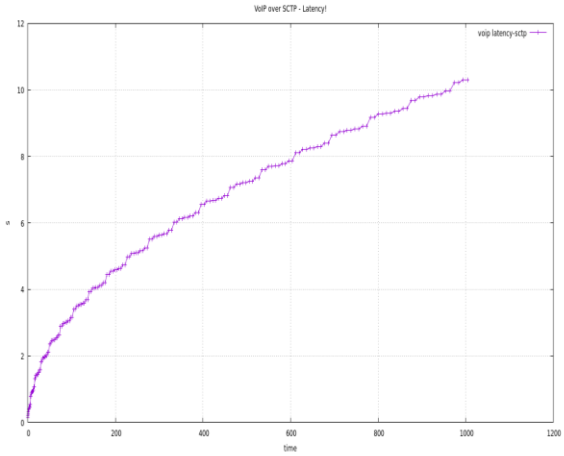

Figure 11: Latency for SCTP

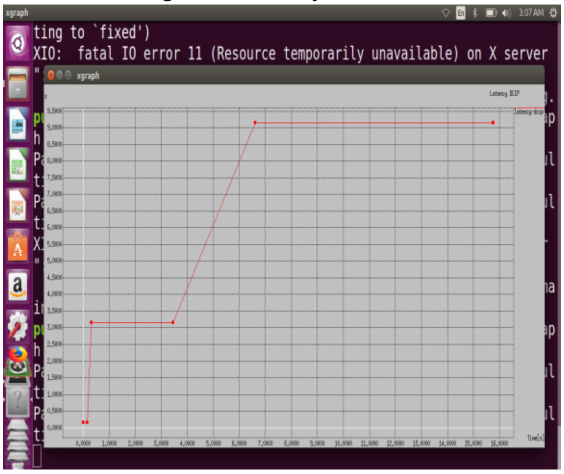

Figure 12: Latency for DCCP

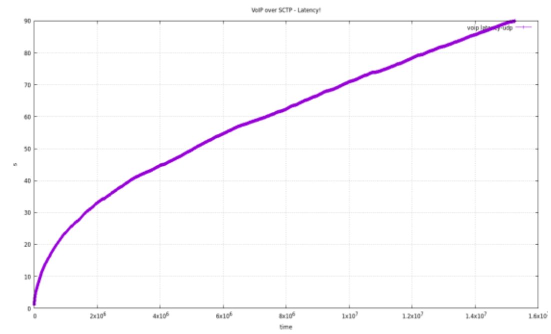

Figure 13: Latency for UDP

Latency: latency for SCTP is increased as simulation time is increased with slightly fluctuations (**figure 8-13**). For UDP, latency remains constant from simulation time 0.3 to 3.43. But, it increased up to 6.5s and remains same up to the maximum value of simulation time. Overall, in case of UDP, latency increased with higher-highest simulation time. In case of TFRC, TCP, SCTP, latency is increased with respect to simulation time.

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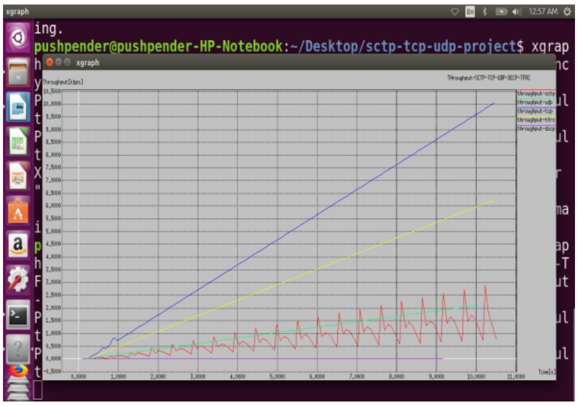

Figure 14: Comparison for Throughput Vs Simulation time

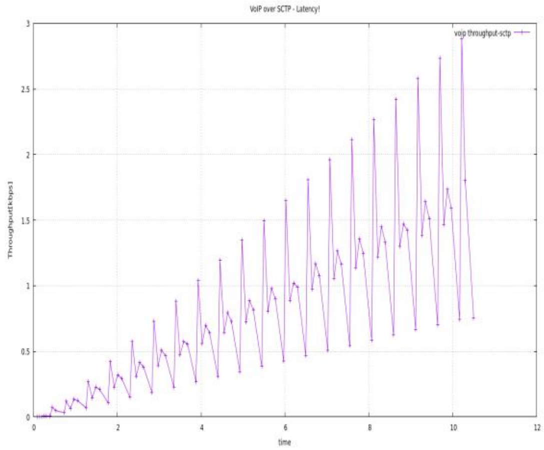

Figure 15: Throughput Vs Simulation time for SCTP

Throughput: throughput for UDP is slightly increased with respect to simulation time (**figure 14-15**). Throughput for TCP is highest as compared to SCTP and UDP. As simulation time is increased, the throughput fluctuation rate for SCTP is increased. SCTP outperforms in respect of throughput as compared to UDP and DCCP network protocols. DCCP have the lowest throughput rate as compared to SCTP, TCP, DCCP, and TFRC protocols. Throughput for DCCP remains same at all simulation timings. As the simulation time is increased, throughput for TCP, SCTP, UDP, and TFRC is increased. There is small throughput fluctuation in case of TCP at lower simulation time. But there is continuous throughput fluctuation in case of SCTP network protocol. It increases with respect to simulation time.

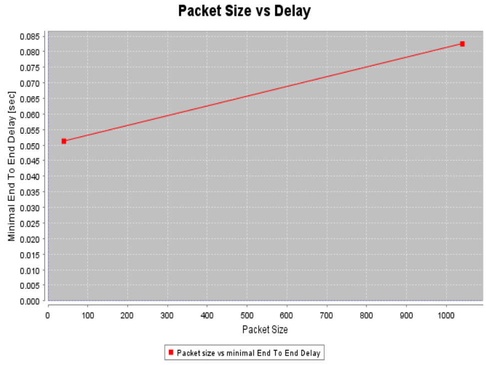

Figure 16: Packet size Vs minimal End To End Delay (TCP)

Figure 16 shows the packet size w.r.t. End to End delay for TCP network protocol. Here, it is clearly illustrated that when packet size will increase, the minimal End2End delay will also increase. No any fluctuation for minimal End2End delay is presented w.r.t. the packet size. When packet size is reduced, the minimal End2End delay will also goes down.

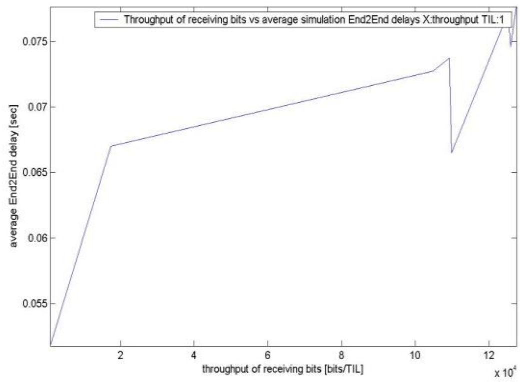

Figure 17: Throughput of receiving bits Vs average End2End delay (SCTP)

Figure 17 illustrates the throughput of receiving bits w.r.t. average End2End delay in SCTP protocol. When throughput of receiving bits in SCTP is increased, the average End2End delay is also increased. When this throughput reaches a stage at 10×10^4 bits/s, average End2End delay suddenly reduces to some position down and again increases; whenhen throughput of received bits is increased. At the next higher stage of throughput, End2End delay fluctuates.

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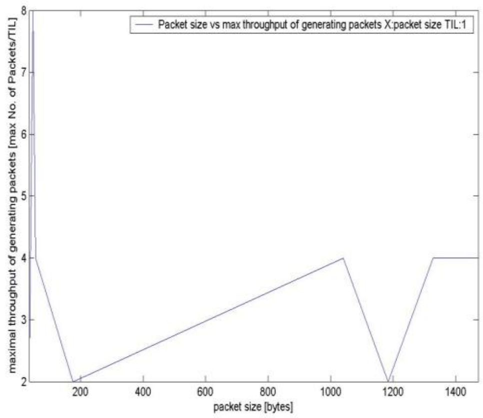

Figure 18: Packet size Vs Maximal throughput of generating packets (SCTP)

Figure 18 illustrates the packet size w.r.t. maximum throughput of total generated packets for SCTP protocol. Here, when packet size is at its lowest size, the maximal throughput of generated packets is very high. But when packet size increases up to 200 bytes, the maximal throughput is at its lowest stage. Suddenly when packet size increases from 200 bytes to 1000 bytes, the maximal throughput is also increased. But, at the next stage of packet size (i.e. 1000 bytes to 1200 bytes), the maximal throughput of generated packets is again goes to its down position. The maximal throughput from packet size 1200 bytes to 1300 bytes is again increased and remains constant up to 1400 bytes packet size.

V. CONCLUSION

We have simulated five transport protocols in this work. UDP and DCCP are mostly unreliable protocols. SCTP produces best results in respect of overall delay. SCTP supports multi-homing and multi-streaming characteristics for data transmission over transport layer. In some cases TCP outperforms as compare to SCTP, UDP, DCCP, and TFRC transport protocols. DCCP is not a complete protocol, just it support for congestion control and more compatible with TCP protocol. UDP produces better throughput results; but only when data transmission is only for short time period. But, when transmission time is high, TCP and SCTP outperform as compare to UDP. Throughput for UDP remains mostly constant at all simulation times; while it fluctuates for SCTP. Overall performance for SCTP is best for all network scenarios i.e. less packet lost, reliable service, good acknowledgements, data transmission in the form of streams. Best features of TCP and UDP have been including in SCTP transport protocol. DCCP protocol is helpful for TCP in respect of congestion control over the network. Average delay for SCTP is higher than UDP, but as data transmission time is increased, UDP produces poor performance in respect of packet lost and throughput.

REFERENCES

1. Ali Hussein Wheeb, “Performance Evaluation of UDP, DCCP, SCTP and TFRC for Different Traffic Flow in Wired Networks”, International Journal of Electrical and Computer Engineering (IJECE) Vol. 7, No. 6, December 2017, pp. 3552~3557 ISSN: 2088-8708, DOI: 10.11591/ijece.v7i6.pp3552-3557.
2. Pallavi Gangurde, Sanjay Waware, Dr. Nisha Sarwade, “Simulation of TCP, UDP and SCTP with constant traffic for VOIP Services”, International Journal of Engineering Research and Applications (IJERA) ISSN: 2248-9622, Vol. 2, Issue 3, May-Jun 2012, pp.1245-1248.
3. Ravonimanantsoa Ndaohialy Manda-Vy, Ratiarison Adolphe, “Performance Analysis Of Voip Using Udp And Tcp Protocol”, International Journal of Engineering and Advanced Research Technology (IJEART) , ISSN: 2454-9290, Volume-2, Issue-2, February 2016, pp. 10-15.
4. SielayGebru, PrachiKadam, “Simulation Based performance evaluation and comparison of wired VoIP services over UDP and SCTP protocol”, International Journal of Engineering Trends and Technology (IJETT) – Volume 33 Number 9- March 2016, pp. 462-467.
5. Sharat Kaushik, Poonam, Anita Tomar, “A Comparative Analysis of Transport Layer Protocols”, International Journal of Information & Computation Technology. ISSN 0974-2239 Volume 4, Number 14 (2014), pp. 1403-1410.
6. M.V.Sreeraj, T.Satya Savitri, “SCTP and FEC based Loss Recovery Technique for VoIP”, International Journal of Innovative Research in Electrical, Electronics, Instrumentation and Control Engineering, Vol. 2, Issue 1, January 2014, pp. 569- 575.
7. Shiferaw Hermela Shimelash, Li Yao Hui,Fares Mohmmed Mehdi Hassan, “Study on Performance of simulation analysis on Multimedia Network”, International Research Journal of Engineering and Technology (IRJET) e-ISSN: 2395 -0056, Volume: 04 Issue: 03| March -2017, pp. 1726- 1731.
8. G. N. Vivekananda and P. Chenna Reddy, “Performance Evaluation of TCP, UDP, and SCTP in MANET”, ARPN Journal of Engineering and Applied Sciences, VOL. 13, NO. 9, MAY 2018, pp. 3087-3092.
9. Jiawei Chen, “Congestion Control Mechanisms of Transport Protocols”, TKK T-110.5190 Seminar on Internetworking, April, 2009, 31(4):360–404.
10. Adnan ghani, Alaa almagrabi, Shakeel ahmad, Sher afzal khan1,, “Performance Evaluation of TCP and DCCP Protocols in Mobile Ad-hoc Networks (MANETS)”, VFAST Transactions on Software Engineering, Volume 10, Number 1, May-June , 2016, pp. 1-12.
11. G. Fairhurst, B. Trammell, M. Kuehlewind, Services Provided by IETF Transport Protocols and Congestion Control Mechanisms, RFC 8095.
12. S. Floyd, M. Handley, J. Padhye, J. Widmer, TCP Friendly Rate Control (TFRC): Protocol Specification, September 2008, RFC 5348.
13. Bhrouz A Forouzan, Data Communications and Ntworking, Tata McGraw Hill Education Private Limited, New Delhi, Fourth Edition, 2006.
14. https://www.isi.edu/nsnam/ns/

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