A Performance Study of the Media Streaming Protocols

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ABSTRACT

Nowadays Media Streaming Service is one of most Internet's bandwidth usage service so it was developed for the better performance. In the past the media streaming technology works with standard protocols as HTTP, TCP and UDP but now, the new protocol such as RTSP and SCTP was developed and focus at smooth transmission with higher level of data's correction. This research is to study and compare the transmitting performance of four protocols in different conditions to find the best fitting protocol for each condition. From the experiment, I found that TCP is appropriate for the networks that have lots of bandwidth and need the data correction, UDP is appropriate for streaming that need a quick and smooth transmitting by ignore some little mistake like packet lost or request time out, RTSP/UDP is appropriated for continuous streaming in large networks, SCTP is appropriated for streaming in small network that need fast and continuous flow transmission with error correction. This result shows that each protocol are suitable for difference conditions, so users should know the condition and purpose of their streaming before select a protocol for the best streaming performance.

Keyword: Media Streaming, RTSP, SCTP, HTTP

1. Introduction

The Streaming Media Technology is increasingly important in the daily lives of worldwide Internet users. This technology was developed to meet the increasing demand and offer better technology to use. Data Streaming is one type of data communication then it is necessary to have a streaming protocol which is a specification for data transmission characteristics. The quality of data transmission is based on the protocol and transmission condition such as the number of recipients, the bandwidth and the traffic in the network during that period. In the beginning of the streaming, TCP and UDP protocol are adapted to make streaming. But these two protocols were limited used then new protocols were developed for streaming especially such as RTSP (Real Time Streaming Protocol) [1] and SCTP (Stream

Control Transmission Protocol) [2]

To study the data streaming performance of each protocol, I have conducted experiments and compare the result among four protocols under controlled conditions to compare the performance of these four protocols.

2. Background

2.1 Streaming Media Technology

Millions of people use their personal computers and streaming media technology to send/receive audio and video over the Internet. With broadband and video/audio compression technologies, streaming media video is rapidly improved in quality. Moreover, Lots of user terminals can now be used, such as office desktops, personal digital assistants and mobile phones [7].

2.2 TCP (Transmission Control Protocol)

TCP [3] is a transport layer and connection oriented protocol which has to established the connection between sender and receiver before transmitting the data. TCP has advantages in data reliability and ordering but if obtained data loss or delay, TCP will not send data to the application and will request the lost or delayed data again until it has reached the destination properly. This problem of TCP is called Head-of-Line Blocking (HOL) which causes TCP is not appropriate to be used for Media Streaming. In addition, another important problem of TCP is that TCP creates a single communication channel. If the HOL problem occurs in a communication channel, data cannot communicate for a long period of time.

2.3 UDP (User Datagram Protocol)

UDP [4] is a transport layer protocol which transmits small packet, called datagram, without receiver's acknowledgement required. UDP can be use for multimedia transmission such as Video On-Demand by both Broadcast and Multicast. The advantages of using UDP is fast and efficient but the disadvantage is the quality of the data that if some data loss during the transmission, no retransmitting this data because the receiver cannot know about the data loss.

2.4 HTTP (HyperText Transfer Protocol)

HTTP [4] [13] is an application layer protocol which is known as a protocol that effectively carries HTML pages and allows the hyperlinks to transfer to another document or web site. HTTP server and HTTP client computers have a two-way connection, that feedback from the client computer can be send to the server. Then, lost or damaged packets can always be retransmitted and the received file can be fully restored. HTTP can also be used for media download, especially if the files are small and the number of concurrent users is limited. If the connection speed is lower than the media data rate, the media still gets through but it may not play smoothly. The transfer time of file download depends on the size of the file and the speed of the connection.

2.5 RTSP (Real Time Streaming Protocol)

RTSP [5] is an application layer protocol which can be used over both TCP and UDP protocol. RTSP is used for continuous media transmission between server and client via Internet and the RTSP server can send the data to the client by both unicast and multicast method. This protocol uses buffer to store data then it can work continuously and suitable for On-Demand service.

2.6 SCTP (Stream Control Transmission Protocol)

SCTP [6] is a transport layer protocol and was developed to support data transmission which is not suitable to TCP. It is message-oriented like UDP and ensures reliable, in-sequence transport of messages with congestion control like TCP. SCTP provides many features that are similar to TCP but has added several features to solve problems found in TCP such as multi-streaming, multi-homing, 4-way handshake and 32-bit receive windows size. SCTP offers the following services to its users:

1) Acknowledged error-free non-duplicated transfer of user data.
2) Data fragmentation to conform to discovered path MTU size.
3) Sequenced delivery of user messages within multiple streams.
4) Optional bundling of multiple user messages into a single SCTP packet.
5) Network-level fault tolerance through supporting of multi-homing.

2.7 Past Experiments

In the past, there were many researches in comparing the performance of protocols for media streaming. Brennan and Curran [8] and Caro et al. [9] tested the congestion control of SCTP in both of the network with high traffic and network with high data loss with the NS-2 system. They set SCTP to work only a single stream and adjusting working patterns to work like TCP and found that when there are lots of lost data, SCTP seem to be high-performance over TCP for about 10 to 30% due to the ability to make fast recovery of SCTP is better than TCP. Natarajan et al.[10] did an experiment by simulating web server running on HTTP protocol and used multi-streaming on SCTP to solve HOL problem. The results indicated that they can eliminates HOL found in TCP and separate transmitting each object in a stream decreased the transmitting time by about 10 to 40%. Rajamani et al. [11] and

Osterdahl [12] have experimented with transmitting SCTP multi-streaming data on HTTP. The results showed that TCP can transmit data faster than SCTP when there is no lost data because SCTP has higher overhead. But when there is some lost data, SCTP can send data faster than TCP for about 15 to 20% and a faster delivery rate of SCTP increases when data lost rate increases. Nagamalai and Lee [14] found that SCTP congestion control mechanisms are based on TCP congestion principals. Then, SCTP congestion control can perform badly in high-speed wide area networks because of its slow response with large congestion window.

3. Experiment

I experimented in a Wide Area Network (WAN) with a Cisco-1921 router interconnected two Local Area Networks (LAN) with different network address. In each LAN, there was a Cisco-2960 switch which interconnected between client or server machines and connected to the router as shown in figure 1.

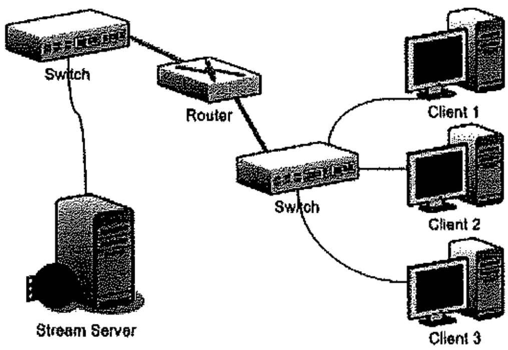

Fig.1 Diagram of Expermental Network.

By streaming video file to the target machines for 60 seconds and check the bandwidth usage in the network systems, I've determined the transmission performance in term of bandwidth usage (kilobyte per second). The bandwidth usage was monitored in every second that can show me the variation of the bandwidth usage. The experimental procedure is as follows:

3.1 Video streaming

The video stream was sent to the client machine and the bandwidth usage in the network was monitored under various scenarios:

- The video was streamed by HTTP, SCTP, RTSP over TCP and Multicast RTSP over UDP, respectively.
- The video was to 1, 2 and 3 clients, respectively.

Each experiment was repeated 3 times.

3.2 Data analysis

The experiment results were averaged and compared as follows:

- Three results of each condition were averaged.
- The peak and average bandwidth usage of whole transmission period of each scenario were determine from averaged data as the result shown in table 1 - 4.

TABLE 1. BANDWIDTH USAGE OF HTTP PROTOCOL

# Client	Bandwidth Usage		
	Peak (KB/s)	*Average (KB/s)*	*Standard Deviation*
1 client	138	66	21
2 clients	222	115	42
3 clients	419	167	76

TABLE 2. BANDWIDTH USAGE OF SCTP PROTOCOL

# Client	Bandwidth Usage		
	Peak (KB/s)	*Average (KB/s)*	*Standard Deviation*
1 client	43	18	8
2 clients	74	38	14
3 clients	220	92	46

TABLE 3. BANDWIDTH USAGE OF RTSP OVER TCP PROTOCOL

# Client	Bandwidth Usage		
	Peak (KB/s)	*Average (KB/s)*	*Standard Deviation*
1 client	188	179	4
2 clients	375	359	8
3 clients	558	538	9

TABLE 4. BANDWIDTH USAGE OF RTSP OVER UDP PROTOCOL

# Client	Bandwidth Usage		
	Peak (KB/s)	*Average (KB/s)*	*Standard Deviation*
1 client	199	192	3
2 clients	197	192	2
3 clients	196	191	4

From table 1 - 4, we can see peak and average bandwidth usage of video streaming for 1, 2 and 3 clients with HTTP, SCTP, RTSP/TCP and RTSP/UDP protocol, respectively.

4. Result and Analysis

From the experiment results, I can display the bandwidth used for 1 client of each protocol in line charts as shown in figure 2. I found that SCTP used lowest bandwidth (18 KByte/sec in average) and relatively steady with a standard deviation equal to 8. HTTP Bandwidth is the second most used (66 kilobyte/sec in average) but unsteady usage with a standard deviation equal to 21. Both RTSP over UDP and TCP protocol are the most bandwidth used protocols (199 and 188 KByte/sec in average) with smooth bandwidth usage.

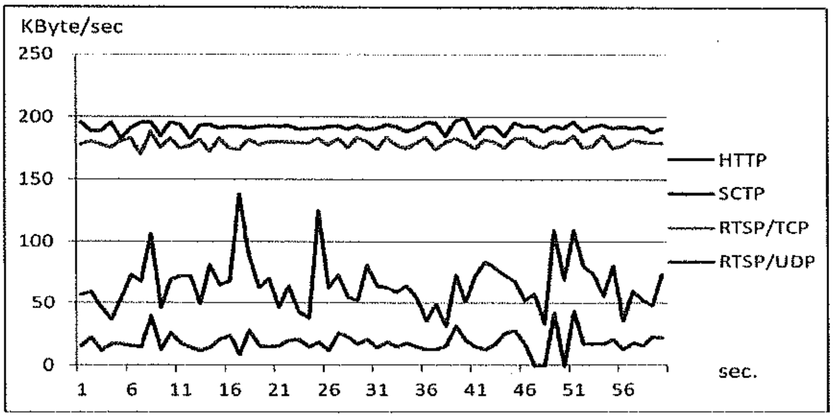

Fig.2 Bandwidth Usage for 1 Client of all Protocols

As a result in the previous section, I found that RTSP over TCP and RTSP over UDP make the similar worst result but RTSP over UDP has the ability of multicast data transfer. From bandwidth usage of HTTP, SCTP and RTSP over TCP which are unicast protocols in figure 3, 4 and 5, we can see that when the client increases the amount of bandwidth usage increases with the number of client. But in the figure 6, bandwidth usage of RTSP over UDP protocol is almost constant because the function of multicast makes the constant bandwidth despite the number of client increase.

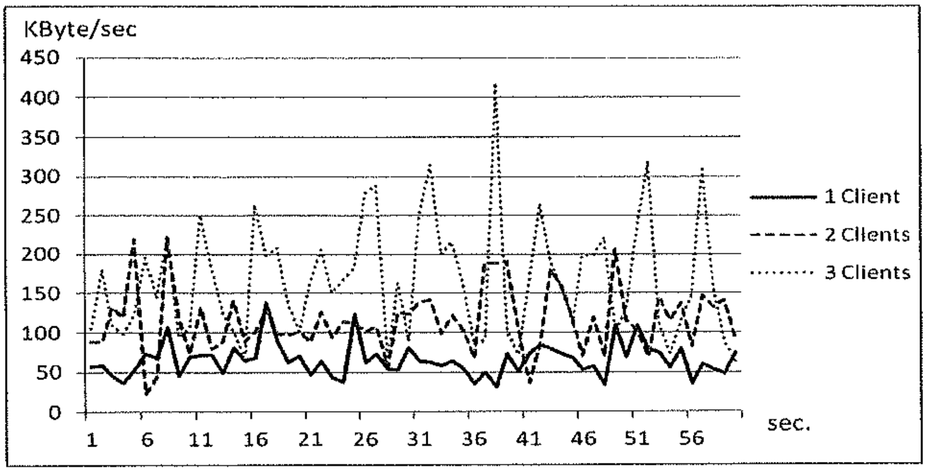

Fig.3 Bandwidth Usage of HTTP Protocol

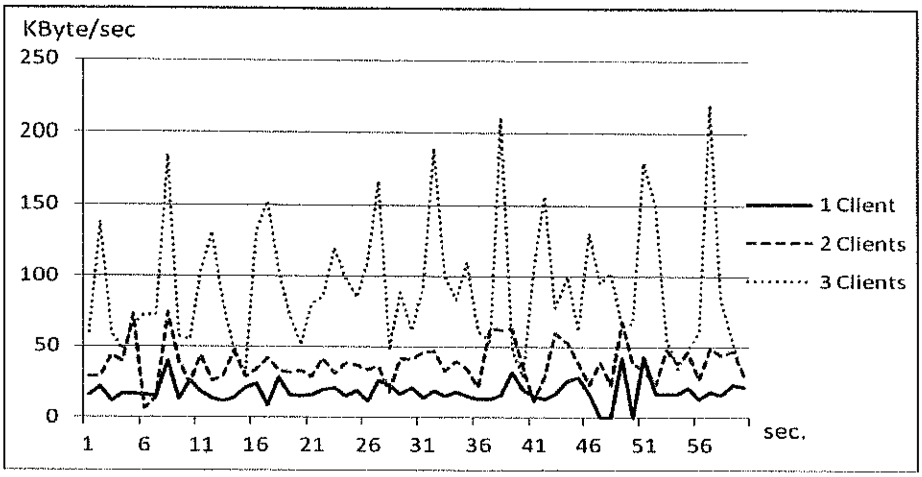

Fig.4 Bandwidth Usage of SCTP Protocol

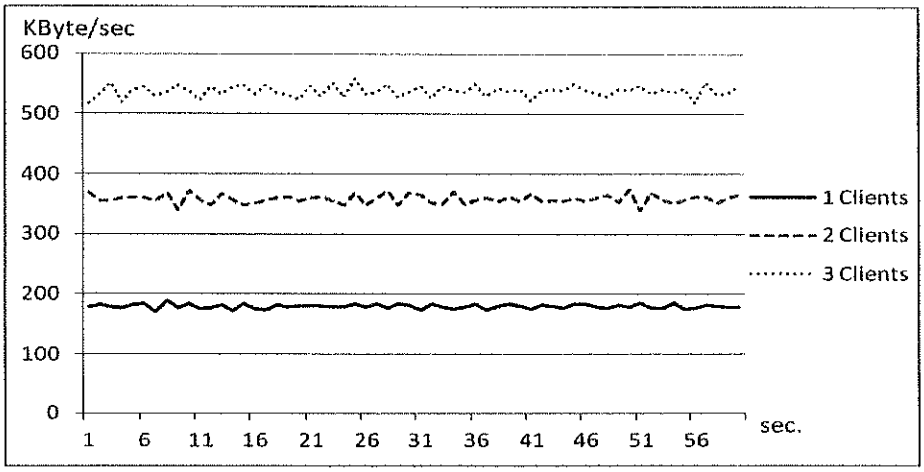

Fig.5 Bandwidth Usage of Unicast RTSP Protocol over TCP

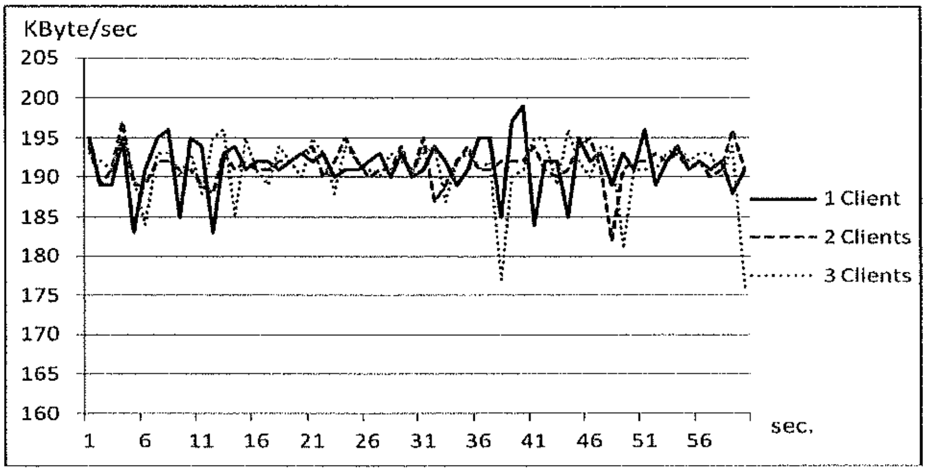

Fig.6 Bandwidth Usage of Multicast RTSP Protocol over UDP

In comparison among different protocols on number of the client, I found that more number of client in the system make more bandwidth usage for the unicast protocols including SCTP and HTTP as the results shown in figure 7 and 8 because unicast data transmission is a communication between a single sender and a single receiver over a network. The term exists in contradistinction to multicast which is a communication between a single sender and multiple receivers. Then, it can be concluded that Multicast RTSP over UDP Streaming System is ideal for a streaming network with large number of client.

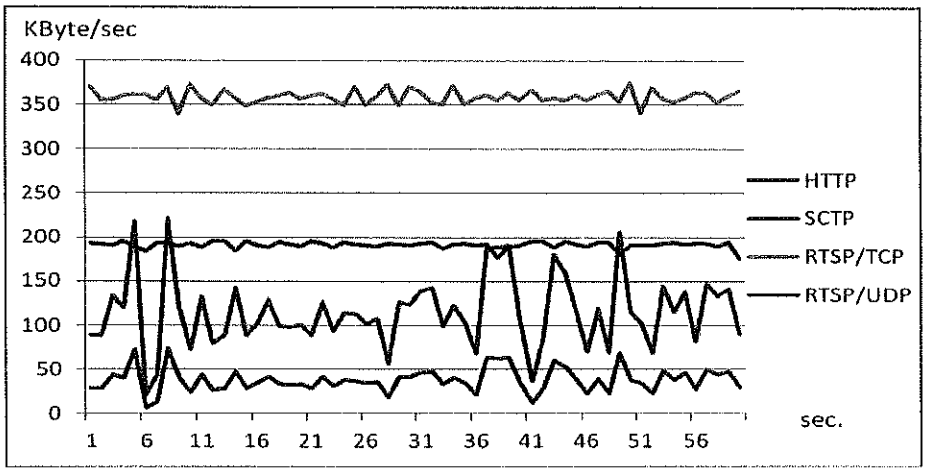

Fig.7 Bandwidth Usage for 2 Clients of all Protocols

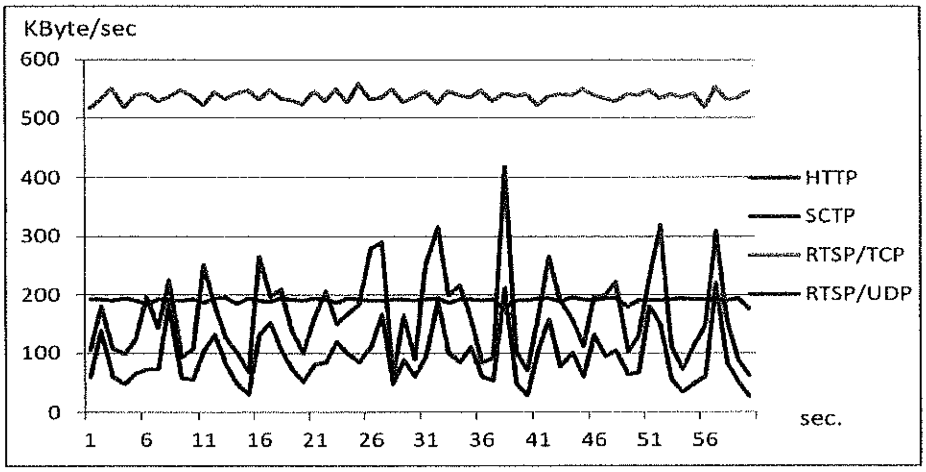

Fig.8 Bandwidth Usage for 3 Clients of all Protocols

5. Conclusion

SCTP presents the best bandwidth usage protocol for single client and small number of client in the system. Beside SCTP is better than HTTP, RTSP/TCP and RSTP over UDP, it has advantages in data ordering and congestion control as TCP with the better performance than TCP. Both RTSP over TCP and UDP give a very smooth bandwidth usage but in the large number of client environment, only RTSP over UDP will be better than SCTP and HTTP because RTSP over UDP have the multicast function. On the other hand, if we use RTSP over TCP, we will get the better reliability than RTSP over UDP.

6. Acknowledgment

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REFERENCES

[1] H. Schulzrinne, A. Rao, and R. Lanphier, "Real Time Streaming Protocol (RTSP)", *IETF RFC 2326 (proposed standard)*, Apr. 1998. [Online]. Available: http://www.ietf.org/rfc/rfc2326.txt

[2] R. Steward, Q. Xie, Stream Control Transmission Protocol (SCTP): *A Reference Guide*, NY: Addision Wessley, 2001.

[3] M. Scharf, S. Kiesel, "Head-of-line Blocking in TCP and SCTP: Analysis and Measurements", *In proc. of Global Telecommunications Conference*, 2006. (GLOBECOM '06), pp. 1 - 5.

[4] D. Austerberry, The Technology of Video and Audio Streaming (2nd Ed.), Taylor & Francis, 2005.

[5] H. Schulzrinne, A. Rao, R. Lanphier, M. Stiemerling, Real Time Streaming Protocol 2.0 (RTSP), *Technical Report*, 2010.

[6] T. Dreibholz, E. Rathgeb, I. Rüngeler, R. Seggelmann, M. Tüxen and R. Stewart,"Stream Control Transmission Protocol: Past, Current, and Future Standardization Activities", *IEEE Communications Magazine*, April, 2011, pp.82-88.

[7] F. Kozamernik, "Media Streaming over the Internet - an overview of delivery technologies", *EBU Technical review*, October 2002.

[8] R. Brennan and T. Curran, "SCTP Congestion Control:Initial Simulation Studies", *Proc. 17th Int'l Teletraffic Congress*, Elsevier Science, 2001.

[9] A. Caro, K. Shah, J. Iyengar, P. Amer, R. Stewart, "SCTP and TCP Variants Congestion Control Under Multiple Losses", *Technical Report TR2003-04*, Computer and Information Sciences Department, University of Delaware, February, 2003.

[10] P. Natarajan, JR. Iyengar, PD. Amer, R. Stewart, "SCTP An Innovative Transport Layer Protocol for the Web", *Proceedings of the 15th international conference on World Wide Web*, 2006.

[11] R. Rajamani, S. Kumar, N. Gupta, "SCTP versus TCP Comparing the Performance of Transport Protocols for Web Traffic", University of Wisconsin-Madison, 2002.

[12] H. Osterdahl, "A comparison of TCP and SCTP performance using the HTTP protocol", 2005, *[Online]*. Available: http://www.it.kth.se/courses/2G1305/Sample-papers/Henrik_Oesterdahl-sctp-20050525.pdf.

[13] R. Fielding, UC Irvine, J. Gettys, J. Mogul, H. Frystyk, T. Berners-Lee, "Hypertext Transfer Protocol (HTTP)", *RFC 2068*, Jan. 1997. [Online]. Available: http://www.ietf.org/rfc/rfc2068.txt

[14] D. Nagamalai, J. Lee, "Performance of SCTP over high speed Wide Area Networks", *Proceedings of the 2004 IEEE Conference on Cybernetics and Intelligent Systems Singapore*, 2004.

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On conclusion of double-blind review results, we are pleased to inform you that your paper above is accepted for **Oral** presentation at 2016 Annual Conference on Engineering and Information Technology (ACEAIT), held from March 29-31, 2016 in Kyoto, Japan We sincerely invite you to present your research in ACEAIT 2016.

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