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A survey of Analyzing performance of TCP, SCTP and UDP-based protocols in IOT Networks

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Abstract

The transmission control protocol (TCP) is a transport layer protocol that is widely used for various Internet applications, e.g., World Wide Web (WWW), email, and file transfer. However, the retransmission scheme in TCP is not appropriate for multimedia streaming applications because it can increase the end-to-end delivery latency. Therefore, user datagram protocol (UDP) is a commonly used transport layer protocol for multimedia streaming applications. UDP does not employ any flow control schemes in response to network congestion, and therefore it can burden other users on the network and, ultimately, lower its service quality . To overcome this limitation, real-time transport protocol (RTP) and real-time control protocol (RTCP) can be adopted on top of UDP in multimedia stream applications[2, 3]. That is, the RTP/RTCP layer supplements the functions of UDP by correcting out-of-order data and controlling the volume of data transmitted by senders for congestion control. However, these actions rely on periodic reports between the sender's and receiver's RTCP, they cannot control packets nor respond actively to network conditions. A new transport layer protocol, stream control transmission protocol (SCTP) has been proposed by Internet Engineering Task Force (IETF SIGTRAN Working Group. Although it was first developed for telephone signaling, it is gradually expanded into a general-purpose transmission layer. Like TCP, SCTP provides reliable service and flow control mechanisms. In addition, similar to UDP, it can support unreliable transmission[4]. SCTP can provide multi-stream and multi-homing services for a single connection. In particular, it can differentiate the level of reliability provided to messages, which is called SCTP partial reliability (PR-SCTP)[5]. PR-SCTP has the function of setting the reliability level for specific messages. The preset reliability level is used to determine the timing when the retransmission of a specific data message is stopped. The function can be effectively applied to traffic containing different types of data, such as I, P, B frames in MPEG streaming applications. However, PR-SCTP attempts transmission at least once even for messages that do not require any retransmission due to the stringent delay constraint. In addition, if retransmission is given up, it has to send a Forward-TSN Chunk to the receiver. Recently, multimedia streaming protocols are required to control its sending rate in response to the congestion condition of the network[6, 7] .It is because nonresponsive streaming to network congestion, such as UDP, starves TCP flows under congestion conditions. In this paper, we propose to use SCTP's congestion control for multimedia streaming. The performance metrics used are delay, jitter, throughput, and packet loss. These metrics are evaluated at the base station via TCP, SCTP, DCCP, and UDP protocols over the 4G-LTE technology. The obtained results show that the DCCP performs the best in throughput improvement with the minimization of delay and jitter as compared to UDP, TCP, and SCTP. The rest of this paper is organized as follows. Section 2 reviews existing video data transmission protocols. Section3 describes a proposed scheme for streaming multimedia data, called TC-SCTP. Section 4 evaluates the performance of TC-SCTP and Section. Section 5 SSH OVER SCTP and Section 6 concludes this paper.

Keywords: SCTP, UDP-based protocols, RTP/RTCP, technology, transmission control protocol.