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Preventing quality degradation of video streaming using selective redundancy

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ABSTRACT

Video streaming has already become the main portion of data traffic over the Internet. However, it is hard to guarantee the user-perceived video quality in single access networks due to the limitation in available bandwidth. Multihomed systems enable concurrent multipath data transfer over multiple network paths. Path control can be implemented at the transport layer such as in Stream Control Transport Protocol (SCTP). SCTP has built-in support for multihoming, but offers only a limited standard path selection mechanism. In this paper a new method for video streaming over SCTP is proposed. Secondary paths are used to send redundant information to prevent degradation of viewer's perceived quality. The efficiency of the proposed method is demonstrated by computer simulations using video coded with H.264/MPEG-4 AVC (Advanced Video Coding). The results show that the proposed method can prevent degradation of video quality at the cost of small extra bandwidth usage.

1. Introduction

The development of new technologies and the broadening of Internet access facilities lead to an increasing demand for video streaming. The availability of multiple network interfaces in devices brings new possibilities of communication. Smartphones are one of the most significant examples of such systems. A multihomed system uses more than one network interface, physical or logical, for communication. From the viewpoint of transport layer, each interface is a different way to access the network. The use of multiple paths can improve the resilience and stability of a connection, ensuring service continuity.

One of the key factors affecting the quality of video streaming over IP is the end-to-end delay. The increase in delay may compromise the quality of voice and video streaming [1]. Other important factors that affect viewer's perceived quality are the throughput and packet loss. The throughput currently being offered by services providers seems to be on the increase to meet video transmission requirements [2].

The most widely used transport layer protocols are Transmission Control Protocol (TCP) and User Datagram Protocol (UDP). UDP is applied to video streaming and real-time applications while TCP is applied to regular data transfers. However, neither TCP nor UDP are implemented to exploit the available multiple interfaces.

The Stream Control Transmission Protocol (SCTP) has been standardized by the Internet Engineering Task Force (IETF) in RFC4960 [3]. By default, SCTP uses one path for transmission, called the primary path, and leaves the other in redundancy. RFC4960 specifies that the secondary paths can be used for retransmission or backup in case of failure of the primary path. However, multiple paths also can be exploited to improve the quality of service provided to applications. By default, SCTP exchanges the primary path after a loss of 6 packets. A packet is considered lost if the acknowledgment is not received within a certain time threshold, called retransmission time out (RTO). When a packet is declared lost, the next RTO is doubled. The minimum initial value for RTO is set to 1 s. Thus, by default, the switching of primary path will take at least 63 s. Considering video streaming applications, this value is too large to allow the use of default SCTP to mitigate the effects of higher latency of primary path in viewer's perceived quality. An alternative is the use of the end-to-end delay as a criterion for switching primary path. This alternative, known as delay-centric strategy, has been previously proposed by Kelly et al. [4] and Kashihara et al. [5]. In delay-centric the path with lower end-to-end delay is set as primary. The end-to-end delay is estimated by the round trip time (RTT), which is the time interval between the transmission of the message and the receipt of the corresponding confirmation. This type of mechanism has been applied previously in voice over IP systems [6]. The delay-centric strategy for switching primary path can effectively prevent quality degradation, but its use can lead to

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stability problems due to frequent exchange of paths. Nonetheless, the delay centric strategy still uses only one path for data transmission and leaves others in redundancy, which does not fully exploit the ability of multihoming systems to provide the best possible quality to video streaming user.

Concurrent multipath transfer (CMT) is a proposed SCTP extension that uses all the available paths for simultaneous data communication between end hosts [7]. CMT schedules new data to be sent according the congestion window on corresponding paths. When congestion window space is available simultaneously for two or more destinations, data is sent to these destinations in arbitrary order. Thus, CMT allows simultaneous use of available paths to increase the throughput and network resiliency. Recently, the implementation problems of CMT were addressed by many researchers [8,9].

In this paper a method to mitigate the degradation of perceived video quality using the secondary paths for redundant packet transmission is proposed. The packets to be sent redundantly are selected according to their importance for image reconstruction. The amount of information to be sent redundantly is obtained by observing the RTT of primary path. The method is called Selective-Redundancy Multipath Transfer (SRMT). To assess the effectiveness of the proposed method, comparison with the full redundant strategy (FRMT) and with default SCTP was performed. The codec used in the tests was MPEG-4 AVC/H.264, although the proposed approach can be employed with the newer HEVC or any codec that uses temporal redundancy for video compression. The performance evaluation was done through computer simulations, using data sampled in real access networks, and using synthetic delay traces generated according recommendation ITU G.1050 [10]. The video quality was estimated using peak signal-to-noise ratio (PSNR) and structural similarity index (SSIM).

The remaining part of this paper is organized as follows. Section 2 presents related work and Section 3 introduces SCTP multihoming and MPEG key concepts. Section 4 presents the proposed method and Section 5 shows the performance evaluation. Section 6 proposes an alternative method for improvement by exploiting temporal correlation of delay. Section 7 presents the conclusions.

2. Related work

Kim et al. [11] compare different transport protocols in the application of IPTV systems and show that SCTP performs well in real-time applications.

Okamoto et al. [12] propose to duplicate transmission of SCTP segments containing confirmation intervals (ACK gaps). They show that, by doing so, the loss of these packets in one of the paths does not affect the performance of transmission. They demonstrate the benefits of this strategy in an experiment with wireless links. However, this approach does not provide redundant transmission of packets that may contain sensitive data, such as, I-Frames in a video stream.

Koul and Rao [13] propose a method for redundant transmission of I-Frames. Their goal is to ensure that even after possible loss of an I-Frame, there will still be a copy that can be used for decoding the remaining frames of the group of pictures (GoP) by the receiver. This method does not use the concept of multipath. The video quality metrics used were PSNR and SSIM, and results demonstrate improvement in user perceived quality.

The partial reliability extension of the SCTP, PR-SCTP [14], enables a content-sensitive transport service. Time-sensitive applications, such as real-time video or audio streaming, have been shown to benefit from the PR-SCTP [15], which supports choosing the retransmission policy on a per message basis. An example of such policy is timed reliability that allows an application to specify a lifetime for every message it pushes to the PR-SCTP. Upon expiration of this time, the PR-SCTP does not make any further attempt to retransmit the message [16].

Molteni and Villari [17] explored features of PR-SCTP to improve the quality of MPEG4 video streaming. Their aim was to ensure transmission of I-Frames with priority, in the event of congestion of the primary path, and thus mitigate the effects of the loss of important packets for video playback. Only one path is used, and packets of unconfirmed I-Frames are retransmitted using the secondary paths. The results show improvement in PSNR, in comparison to those obtained by using the UDP protocol.

Using PR-SCTP, Sanson et al. [18] propose a probabilistic model for possible retransmissions on the network, anticipating delivery of frames for secondary paths, thus predicting possible packet loss and reducing the consequences of loss by the primary path. PR-SCTP is also used by Boussen et al. [19] in the IPTV over IP Multimedia Subsystem (IMS), using traffic adaptation of video traffic to the network QoS policies. The results indicate a significant improvement in user-perceived quality.

Tarabuta et al. [20] propose a transmission method with full redundancy, using multiple paths simultaneously over Internet network infrastructure. The method uses at least twice the bandwidth for sending data. The results show improvement in packet loss and delay. The benefits in quality, as perceived by the users, were not evaluated.

Xu et al. [21] analyze the performance of multimedia distribution, using SCTP and CMT. The performance is evaluated by using computer simulations and the PSNR of received video.

Huang and Lin [22] propose a partially reliable concurrent multipath transfer (PR-CMT) protocol for multimedia streaming. The protocol combines concurrent multipath transfer with PR-SCTP's partially reliable and prioritized stream transmission. They analyze the problems involved in evaluating congestion window and receiver buffer blocking and propose a method to solve them. The performance of protocol is evaluated by observing the throughput achieved between client and server by using computer simulations.

Natarajan et al. [23] show that CMT suffers from significant "receive buffer blocking" which degrades performance during both permanent and short-term failure, and introduce a potentially-failed destination state, resulting in a new failure detection and (re)transmission policies. The proposed implementation is called CMT-PF. Performance evaluation using computer simulation show that CMT-PF performs on par or better than original CMT.

The quality-aware adaptive CMT (CMT-QA), proposed by Changqiao et al. [9], includes a series of mechanisms to distribute data chunks over multiple paths in order to mitigate the out-of-order data reception and unnecessary fast retransmissions. The performance evaluation was conducted through computer simulations, with the transmission of files in a multipath topology. Results indicate that the use of CMT-QA leads to a better throughput and delay if compared with CMT and CMT-PF.

The distortion-aware CMT (CMT-DA) include video distortion into the decision process of multipath data transfer, in order to optimize the real-time video quality in integrated heterogeneous wireless networks [24]. Performance evaluation of CMT-DA using computer simulations of H.264 video streaming shows that this strategy outperforms CMT-QA in terms of video PSNR, goodput and inter-packet delay.

The content-aware CMT [25] (CMT-CA) deals with high-definition video streaming over heterogeneous wireless networks. The method was designed to improve the perceived video quality. Unlike previous schemes that ignore the application type, this version of CMT takes into account specific features of the application. The CMT-CA estimates the video content parameters and schedule the video frames using the appropriated path to get optimal video quality. The performance evaluation was done using semi-physical