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Delay optimization using Knapsack algorithm for multimedia traffic over MANETs

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ABSTRACT

Multimedia transmission over Mobile Ad-hoc Networks (MANETs) is crucial to many applications. However, MANETs possess several challenges including transmitting large size packets, minimizing delay, loss-tolerant and buffer size estimation. For effective multimedia transmission, delay should be minimized and packets should be received in the defined order. The existing standards such as 802.11b and 802.11e perform well in wireless networks, but exhibit poor response in MANETs for multimedia traffic, especially in multi-hop networks. In this paper, we first establish the dependency of delay on buffer size and packet size, and then present a delay optimization approach for multimedia traffic in MANETs. We use Knapsack algorithm for buffer management to maximize the in-order packets and minimize the out-of-order packets simultaneously. Our approach exploits the buffer internals and dynamically adjusts the buffer usage so that a node transmits the packets in the desired order to its successive nodes. Careful estimation of packet size and buffer size helps in minimizing the delay, improving the capability of receiving packets in the correct order and reducing out-of-order packets in the buffer at intermediate nodes. Our approach also controls the loss of multimedia data packets during transmission. We validate our approach with real-world examples using network simulator.

1. Introduction

Mobile Ad-hoc Network (MANET) is a network of mobile nodes without any predefined infrastructure. If the two communicating nodes lie within the sensing range they can directly communicate to each other, otherwise they can communicate through multiple hops where nodes act as intermediate routers. The queuing delay is reduced when the nodes are one hop away from each other in the network and lies within the transmission range. Since the communication through intermediate nodes in multi-hop networks increases delay and thereby affect the multimedia transmission. MANETs have limited resource availability and thus need additional support for applications such as video conference and video on demand, which generate huge real-time traffic. Such applications require guaranteed Quality of Service (QoS) of the network connection. The two most important parameters for multimedia traffic are (i) the delay and (ii) order of packets received. Hence, for multimedia traffic each packet should reach the destination within the specified time (before a deadline.), otherwise it becomes ineffective as the delay of the packet receipt increases resulting in either packet loss or packets receive out-of-order.

1.1. Background and motivation

Usually Forward Error Correction (FEC) or Checksum techniques are used to detect errors. Several studies have been proposed in the literature to determine the optimum pre-defined packet lengths for various network design and medium conditions (Sarraf, 1989; Siew & Goodman, 1989). However, small packet size results in higher protocol header overhead. In Przybylski, Belter, and Binczewsk (2005), the authors showed that 70% of packets will receive in-order. On the other hand, in Jaiswal, Lannaccone, Diot, Kurose, and Towsley (2007), the authors showed that 40% of the links in the database would re-order the packets and out of sequence packets will range from 3% to 5%. Boyce and Gaglianello (1998) presented an approach (referred to as Boyce approach in this paper) for streaming MPEG video compressed over

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the internet. They showed that out-of-order packets are 6%, in-order packets are 85% and loss of packets is 7%, which shows that 1.75–2% out-of-order packets will become in-order within the buffer in wireless networks, In Liu, Jia, Wan, Liu, and Yao (2007), a flooding algorithm in MANETs is presented. However, both these approaches did not arrive at optimal solution with respect to order of packets. In Wang and Zhang (2002), out-of-order packets and their response to improve the performance of MANETs is described; however, the performance degrades if there is a congestion.

Multimedia traffic needs more bandwidth to transmit its data packets, less delay to avoid loss and maintain the sequence of packets to synchronize with the receiver. Bandwidth requirement could be overcome to some extent by the use of compression techniques. Minimization of delay could be controlled by choosing a less delay path from source to destination and sequence of packets can be maintained by the use of virtual circuit switching. However, for multi hop networks, this type of switching will increase the overhead because of unique route. This results in loss of packets and degradation of packet delivery ratio, which in-turn increases the delay. To reduce the overhead, multicasting principle is commonly used to transmit the data packets from source to destination for multimedia data transmission in MANETs. So, maintaining the sequence of data traffic and reducing the delay are crucial. That is, packets can adopt multiple paths for transmission towards destination. Hence, there is a probability that packets may come randomly into the buffer which in turn causes delay. Further, packets may not reach the destination within a given amount of time (referred to as *time to live* (TTL)) which causes loss of packets. Since multimedia traffic is a loss tolerant, it needs streamlined data to avoid delay. The existing approaches either reduce the packet size or increase the streaming compression to reduce the loss (e.g., Tolani & Mishra, 2012; Boyce & Gaglianello, 1998). However, reducing the packet size increases the congestion whereas streaming compression only optimizes the bandwidth. These two directions do not modify the order of packets which is an important criterion for multimedia traffic. On the other hand, the standard graph based algorithms (e.g., Kruskal, Prime, Dijkstra, Greedy, Bellman-Ford, etc.) are used to find shortest path and optimize the delay in networks. These algorithms consider the basic network parameters such as queuing delay, propagation delay, transmission delay, processing delay and bandwidth. These parameters are mainly affected by the external to the node buffers. Since packets are transmitted through node buffers to reach the destination, the packets which are not coming in a defined order are stored within the buffer itself. After elapsing some time, these packets fill the buffer space and thereby dropping the incoming packets. Our work investigates the effective usage of the node buffer, which is treated as a leaky bucket, for transmitting the packets in in-order. We use Knapsack algorithm for optimizing the buffer usage. This is because, unlike other optimization algorithms, Knapsack possesses a unique property to maximize one parameter and minimize other parameter simultaneously.

1.2. Contributions and organization of the paper

In this paper, we propose a delay optimization approach for multimedia traffic in MANETs. We do this by (i) developing a mathematical model for delay that establishes the dependency between the buffer size and packet size, and (ii) extend the Knapsack algorithm to suit the nature of routing buffer (i.e., leaky bucket). Here, we increase the in-order property of packets and at the same time decrease the out-of-order packets in the buffer. When the buffer size changes, the number of in-order and out-of-order packets in the buffer changes and thereby overall delay also changes. This is because delay is inversely proportional to the size of the buffer. As the order of packet transmission movements inside the buffer changes dynamically, there are chances for conversion of out-of-order packets into in-order within the buffer. Our optimization approach maximizes the number of in-order packets that are supposed to be received in the out-of-order form within the buffer. This result in minimization of delay as the in-order packet transmission towards destination will increase.

The rest of the paper is organized as follows. Section 2 presents the related work. In Section 3, we describe our mathematical model for delay optimization and in Section 4 we show the viability of our approach with real-world examples. In Section 5, we present discussion and we conclude the paper with Section 6.

2. Related work

In the field of communication and networking, researchers studied the optimum pre-defined size of the packet under certain channel conditions and network design (Sarraf, 1989; Siew & Goodman, 1989). Also existing works have addressed the network condition by considering the variations in the network design. Analysis and calculations for adaptive packet size have been proposed in Lettieri and Srivastava (1998), Modiano (1999), Sheu, Lee, Chen, Yu, and Huang (2000), Yin, Wang, and Agrawal (2004). However, these approaches focus on the design pertaining to medium access control (MAC) or link layer. According to the network designers' view, it is necessary to operate in the MAC layer in case of issues related to the physical transport channel. Performance of data packet transmission can be improved if an approach of fragmenting data units is being optimized for specific applications. However, the fragmentation process increases the bandwidth usage and the delay, and hence not suitable for multimedia applications.

A well-known problem that exists in the wireless communication is the differentiation between packet losses due to congestion and errors in the bit stream. This information may be useful for streaming the packets, but not for finding the loss of packets because of congestion. Other works for packet loss differentiation have been proposed with a focus on either analyzing end-to-end delay or bit errors existing in the frame/packet, which helps in finding the changes in the transport layer protocol (Biaz & Vaidya, 1999; Cen, Cosman, & Voelker, 2003; Garcia & Brunstrom, 2002; Tobe, Tamura, Molano, Ghosh, & Tokuda, 2000).

Nisar, Amphawan, Hassan, and Sarkar (2013) presented a survey which indicates the importance of scheduling techniques in Wireless LAN for real-time applications. Kohlar, Handley, and Floyed (2006) proposed a Datagram Congestion Control Protocol (DCCP) to maximize the performance of multimedia applications. This protocol has poor response for in-order packet delivery. Sarwer, Lochin, and Boreli (2011) described the impact of packet re-ordering to maximize performance of multimedia applications based on the buffer size in order to get more in-order packets in a given time for better QoS during multimedia transmission. However, this approach shows reduced response at the receiver side, as receiver buffer would increase delay when more number of packets received is in out-of-order and thereby resulting in loss of packets. Liao and Chen (2011) described an approach for video quality monitoring and designed a model to improve video performance. However, this approach has much error resilience which degrades video quality. Unlike existing works, our work focuses on improving the in-order packet transmission to improve the QoS for multimedia traffic. Singh, Kumar, and Verma (2012) proposed optimal routing solution from source to destination using ant colony algorithm. However, their approach does not holds good in multi-hop networks for multimedia transmission in MANETs as delay is increased when hops are more between sources to