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Bounds on end-to-end statistical delay and jitter in multiple multicast coded packet networks

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A B S T R A C T

Network coding is a paradigm in data transport that allows a network node to code information flows before forwarding them. While it has been theoretically proven that network coding saves bandwidth and increases throughput of multicast communication, it does not directly consider the quality of service (QoS) requirements of multicast routing. In this paper, we study the problem of establishing minimum-cost, multiple multicast sessions over coded packet networks to meet the statistical QoS target considering bounded end-to-end statistical delay and jitter from source to each destination. For this purpose, we first propose a path-based formulation for the problem and prove that this problem is NP-hard. Then, in order to obtain an exact solution, we present an effective and efficient decomposition approach in which the problem is decomposed into master problem and subproblems, and the solution is built up successively by feasible path generation. Computational results on random networks show that the proposed method provides an efficient way for solving the problem, even for relatively large networks.

1. Introduction

Network coding is a promising generalization of routing that allows a network node to code the information flows before forwarding them. In the traditional routing paradigm, each relaying node can only forward or replicate data without changing its data content. However, when we employ network coding as the data delivery mechanism, nodes are allowed to generate new packets and perform algebraic operations on packets received over incoming links. The idea of network coding was introduced by Ahlswede et al. (2000), where it was shown that network coding is able to achieve the maximum-flow/minimum-cut bound on the multicast capacity. Li et al. (2003) proved that linear coding achieves an optimal throughput of a multicast capacity, and later, Jaggi et al. (2005) presented the algorithm for constructing such linear codes in polynomial-time. Lun et al. (2006) presented the problem of finding a minimum-cost multicast over coded packet networks. It was shown that the solution to the problem can be computed using a linear optimization formulation and they proposed a distributed algorithm to solve it. The analogous problem for the traditional routing is called the Steiner tree problem, which is NP-complete (Resende and Pardalos, 2006; Oliveira and Pardalos, 2011). Chi et al. (2008) addressed the topology design problem of networks based on the characteristics of multicast and network coding. They proposed two heuristic algorithms using link deletion and addition to obtain an optimal topology design of network-coding-based multicast networks. Xi and Yeh (2010) provided an analytical framework as well as a set of distributed solutions for optimizing the configuration of network coding in both wireline and wireless networks. Raayatpanah et al. (2012) considered multiple multicast sessions, where rates over all links are integer multiples of a basic rate. They formulated the problem as a mixed integer linear programming problem and developed an algorithm based on Benders decomposition. In recent years, QoS requirements have emerged due to the development of communication networks and insufficient capacity for real-time and multimedia applications. In order to support real-time multimedia communication, Peng et al. (2012) proposed a cross-layer QoS-aware routing protocol on OLSR (CLQ-OLSR) by efficiently exploiting multi-radio and multi-channel methods. The proposed CLQ-OLSR is designed to estimate the available bandwidth on a channel based on a distributed bandwidth estimation scheme. Yin et al. (2013) considered the minimum-cost multicast tree with the delay-and-bandwidth constraint problem in QoS traditional multicast routing. They proposed an ant colony optimization with colony guides algorithm to solve this problem. Tsao et al. (2013) studied an end-to-end channel allocation scheme that extends the radio-frequency-slot

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method in order to minimize co-channel interference. This scheme first separates the transmission and reception of packets into two channels and then classifies the state of each radio-frequency-slot as transmitting, receiving, interfered, free, or parity. Chang and Mai (2013) considered an end-to-end QoS guarantee framework and its handoff flows for interworking WiMAX PMP (Point-to-Multipoint) and mesh networks with Internet. They proposed an end-to-end QoS adjustment algorithm and a QoS adaptation algorithm to tackle this issue. In order to guarantee both fair resource allocation and high system throughput, Lee et al. (2013) proposed an opportunistic resource allocation approach under combinations of QoS and non-QoS connections in OFDMA networks.

Due to inherent benefits of network coding over routing, such as higher throughput (Ahlswede et al., 2000), higher reliability (Lun et al., 2008; Yang et al., 2010), higher security (Cai and Yeung, 2002), cheaper routing costs networks (Lun et al., 2004), and lower delays (Yeung et al., 2005), research efforts have been developed to enhance conventional IP-based architectures and protocols with QoS support using concept of network coding.

Salavati et al. (2008) proposed an approach to provide QoS using network coding. They proposed an algorithm based on primal–dual decomposition methods to guarantee QoS requirements. Ghasvari et al. (2011) considered the problem of finding a minimum cost multicast based on network coding, where delay values associated with each link, limited buffer-size of intermediate nodes, and link capacity variations over time were taken into account. They presented a decentralized algorithm based on primal and dual decompositions. Mahmino et al. (2006) introduced a coding strategy to obtain minimal upper bounds on the rate of the output flow without excessive buffering and delays. They presented a method to obtain global service curves of the network based on a transfer matrix. Xuan and Lea (2011) considered the admission control problem aimed at preventing network congestion and guaranteeing QoS with network coding. They applied QoS architecture as described in Chu and Lea (2009) that requires no admission control inside the network and can guarantee the congestion-free property.

However, with the development of applications network coding in computer networks, they can be able to support multimedia applications like audio, video conferencing, FTP, HTTP service, and network video meetings, using network coding (Magli and Frossard, 2009). Since real-time transactions are sensitive to network characteristics, such as delay, delay variation, bandwidth, and cost, such applications require the network to provide QoS guarantees. In order to avoid breaks in continuity of audio and video playback, it is necessary to provide bounded delays and bounded delay variation among the source and all destinations while keeping the overall cost of the solution low. In this paper, we consider the problem of choosing an appropriate coding subgraph over coded packet networks under end-to-end QoS guarantees. The selected subgraph provides guarantees in terms of end-to-end delay and delay variation with minimum-cost subgraph while satisfying bandwidth constraints. A similar problem for traditionally routed packet networks was studied in several works (Lee et al., 1995; Wang et al., 2004, 2009; Huang et al., 2011). In fact, delays on a network link occur due to propagation, transmission, and queuing. The delay time in a single multicast session on each link usually assumes a fixed deterministic value. This is due to the fact that there is practically zero queuing delay at each link, propagation and transmission delays will be constant. However, this assumption is not very realistic in multiple multicast sessions. Consequently, in order to provide a more realistic and accurate model, multiple multicast sessions with intra-session network coding are taken into account. In multiple multicast sessions, we assume that instead of one source process at a single node, s, there are M single-rate multicast source processes at nodes $s_1, s_2, \ldots, s_M$. Hence, in a network with limited-bandwidth links, the problem of specifying whether or not a set of multicast connections is feasible becomes more difficult than an equivalent problem with only a single multicast connection. Since multiple multicast sessions share a common link capacity, single multicast session problems are not independent. Consequently, in order to find an optimal flow, we need to solve the problems in a joint manner. Moreover, in this case, the delay times are usually considered as random variables. It is also assumed that the delay distribution for a given link has an unknown distribution and only the mean and variance of the delay distribution are empirically computed. Consequently, we propose the statistical QoS guarantees by defining constraints in terms of expected end-to-end delay and the corresponding violation probability. Furthermore, we consider the constraint on the maximum difference between real-time packet delay and mean delay, known as jitter, along individual paths from source to each destination node. In order to obtain an exact mathematical model for the problem, we first introduce a path-based mixed integer linear programming (MILP) model, which involves a polynomial number of constraints and an exponential number of variables. Subsequently, we show that the problem is NP-hard. Since the problem formulation contains a large number of variables, an iterative algorithm is developed based on decomposition methods. Each iteration of the algorithm considers a primal decomposition method to deal with the path-based MILP model and obtains an upper bound on the optimal objective value. This approach first partitions the original problem into a master linear programming problem and several integer programming subproblems, instead of considering all variables and constraints of a large-scale problem simultaneously. Since the computational complexity of a path-based MILP optimization problem significantly increases as the number of variables and constraints is increased, solving smaller problems with fewer variables and constraints iteratively can be more efficient than solving a single large problem. Moreover, at each iteration of the algorithm, we describe a procedure based on Lagrangian dual decomposition to obtain a lower bound on the optimal objective value of the problem in a reasonable amount of computational time. Since the algorithm, at all iterations, provides both upper and lower bounds on the optimal objective value, a decision maker can terminate the procedure when the two bounds become sufficiently close.

The ability of the proposed approach in reaching optimal solutions are demonstrated through simulation results on random graphs.

The rest of the paper is organized as follows: Section 2 presents a path-based formulation of the proposed problem and its complexity. In Section 3, the proposed problem is solved using the decomposition algorithm. Computational results and the conclusion are presented in Sections 4 and 5, respectively.

2. Problem statement and formulation

2.1. Network model

A communication network is represented by a directed graph $G=(V,A)$, where V is a set of nodes and A is a set of links. We denote a link either by a single index e or by the directed pair (i,j) of nodes, which represents a lossless point-to-point link from node i to node j. Each link, e, is associated with three parameters: a nonnegative cost, c_e, denoting the cost per unit rate of sending coded packets over link $e \in A$; a nonnegative integer capacity, u_e, which denotes the number of packets that can be sent over link e in one time unit; and a random variable, d_e, denoting the packets delay on the link e. It should be noted that in this study one time unit is equal to milliseconds. But, in the real time protocol, one