Automatica 49 (2013) 1310–1317

Contents lists available at SciVerse ScienceDirect

Automatica

journal homepage: www.elsevier.com/locate/automatica

Brief paper

On a rate control protocol for networked estimation[☆]

Vaibhav Katewa[1], Vijay Gupta

Department of Electrical Engineering, University of Notre Dame, Notre Dame, IN-46556, USA

ARTICLE INFO

Article history:
Received 8 November 2011
Received in revised form
21 November 2012
Accepted 29 November 2012
Available online 28 February 2013

Keywords:
Networked estimation
Congestion control
Rate control
Network utility maximization
Stability with delays

ABSTRACT

We study the problem of congestion control in a communication network that is supporting remote estimation of multiple processes. A stochastic rate control protocol is developed using the network utility maximization (NUM) framework. This decentralized protocol avoids congestion by regulating the transmission probabilities of the sources. The presence of estimation costs poses new challenges; however, for low congestion levels, the form of rate controller resembles that of the standard TCP rate controller. Stability of the protocol is analyzed in the presence of fixed network delays.

1. Introduction

The architecture and protocols in a communication network should ideally depend on the objectives of the end users. Traditionally, such networks were used with the sole goal of reliable data transfer. More recently, such networks have been proposed to be used in control and estimation applications in the so-called Networked Control Systems (see, e.g., the special issue Antsaklis & Baillieul, 2007 and the references therein). In such applications, the performance metric is a complicated function of delay, throughput, and reliability; hence, traditional network protocols may not be suitable. For both the cases when the communication network is designed specifically for estimation or control, and when the communication network is shared with data unrelated to such applications, it is of interest to design network protocols that optimize the performance relevant to these applications.

However, most of the research in Networked Control Systems so far has focused on analyzing and designing a single networked control system in isolation. While this has led to important foundational results, it has ignored the new problems that may arise when multiple such systems operate over a common communication network. As an example, networked communication may give rise to congestion or MAC delays. Such effects will impact the performance of every networked control system and in fact, will couple their performance even though the systems may not be dynamically coupled. It is, thus, of interest to study the impact of communication network protocols on the performance of multiple control systems sharing a common network, and further, design network protocols more suitable for estimation and control (Garone et al., 2007; Schenato et al., 2007).

In this paper, we focus on a rate control protocol suitable for an estimation oriented cost function. We consider multiple systems, each of which consists of an estimator that remotely estimates the state of an associated process. A sensor collocated with each process transmits information over a shared communication network to the estimator. The network has capacity constraints for every link. Such a capacity constrained network may result in congestion when the network load increases. Congestion results in packet losses and delays, which adversely affect the estimation performance. We show that traditional rate control protocols such as TCP may not be suitable for optimizing estimation performance, and propose a new distributed rate control protocol that can co-exist with existing rate control protocols.

The problem of congestion control has been well studied for communication networks (see, e.g., Jacobson, 1988). TCP (RFC, 1981) is the most widely used congestion control protocol on the Internet. While originally an engineering heuristic, TCP has now been reverse engineered to show that it is a distributed solution that optimizes a particular utility function (Kelly, 2001). The chief tool in this regard is the Network Utility Maximization (NUM) framework (Kelly et al., 1998) which transforms the end objective to an optimization problem with constraints. The communication protocols are the distributed solutions to these optimization problems (Chiang et al., 2007).

☆ Research was funded in part by NSF awards 0834771 and 0846631. The American Control Conference (ACC2011), June 29–July 1, 2011, San Francisco, California, USA. This paper was recommended for publication in revised form by Associate Editor Huijun Gao under the direction of Editor Ian R. Petersen.

E-mail addresses: vkatewa@nd.edu (V. Katewa), vgupta2@nd.edu (V. Gupta).

[1] Tel.: +1 5746311136; fax: +1 5746314393.

0005-1098/$ – see front matter © 2013 Elsevier Ltd. All rights reserved.
doi:10.1016/j.automatica.2013.01.050

The primary aim of traditional TCP is reliable transfer of data, even at the expense of delays. For estimation and control, it may be more useful to have a lower reliability, but a higher throughput. Moreover, not all processes need to transmit data at the same rate to achieve the same estimation error covariance. Thus, issues such as fairness relevant to traditional TCP may not be applicable. In fact, using TCP for estimation purposes may result in instability of the estimation error covariance. Because of these reasons, designing an estimation oriented rate control protocol is not simply a matter of substituting the estimation error covariance as a cost function instead of the throughput. Our proposed protocol, while sharing the formal structure of TCP protocols, considers these issues directly. The proposed protocol is implemented at the transport layer of the standard OSI layer stack, and thus, preserves the layered structure of the network.

To ensure that the proposed protocol can coexist with the standard TCP, we use a cost minimization framework that is analogous to the standard NUM framework. The total cost that the rate control protocol aims to minimize includes both an estimation performance cost and a congestion cost. The work closest to ours is that of Al-Hammouri et al. (2006) which presents a bandwidth allocation scheme by using a dual form of NUM problem. However, our solution is in the primal form and is similar to the structure of the standard TCP protocol. Moreover, we present a stochastic transmission scheme as opposed to the deterministic transmission scheme in Al-Hammouri et al. (2006).

We also come up with conditions on network delay and system parameters for which the original protocol remains stable. The delays can be time varying in realistic networks. However, we analyze the stability of the system with fixed delays for tractability. Although it is a special case, fixed delay analysis is important and has a rich history for standard TCP (Chiang et al., 2007; Johari & Tan, 2001; Low & Lapsley, 1999; Vinnicombe, 2002).

The main contributions of the paper are as follows

- We propose a probabilistic rate control strategy and evaluate an estimation error measure.
- Using the NUM framework, we obtain a scalable rate control protocol that allocates rates optimally such that an estimation error metric is minimized.
- The protocol is developed in primal form and we show that under low network congestion, it resembles the structure of the standard TCP protocol.

The rest of the paper is organized as follows. In the next section, we describe the problem setting with random delays and formulate an optimization problem. In Section 3, we propose a distributed solution to the problem using the NUM framework and present our analysis results. In Section 4, we obtain conditions under which the network is stable for fixed delays and present simulation results. We conclude in Section 5.

2. Problem formulation

Network and process setting: Consider the problem set up shown in Fig. 1. Let all the sources form the source set $\mathcal{S}$. With every source $s \in \mathcal{S}$, associate a unique destination d and denote the destination set by $\mathcal{D}$. Let every source be connected to its corresponding destination through a shared capacity constrained network $\mathcal{N}$. We model the network as a graph, wherein the end-nodes are the sources and the destinations, the intermediate nodes are routers that forward packets and the edges correspond to the communication channels in the network. Let $\mathcal{L}$ be the set of links in the network and $L(s)$ be the set of links that are used by source s to communicate with its corresponding destination d. Further, denote the route between source s and destination d by R_s. Each link $l \in \mathcal{L}$ has a limited capacity c_l in terms of "packets per time slot" on average. Any individual link may be shared by one or more sources.

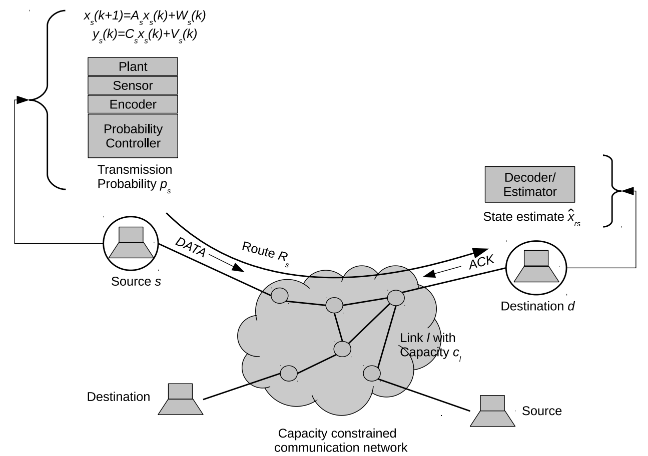

Fig. 1. The problem setup considered. Multiple processes are remotely estimated across a shared communication network.

Each source s comprises of a process P_s, a sensor SR_s, an encoder ENC_s, and a rate controller PC_s. The process P_s evolves according to the discrete-time linear model

$$P_s : x_s(k+1) = A_s x_s(k) + W_s(k), \quad k \geq 0 \tag{1}$$

where $x_s(k) \in \mathbb{R}^{n_s}$ ($n_s \in \mathbb{N}_+$) is the process state and $W_s(k)$ is the process noise. The initial condition $x_s(0)$ and the white process noise $W_s(k)$ are assumed to be Gaussian with zero mean and variance $X_s > 0$ and $Q_s > 0$, respectively. The output of the process P_s is sensed by the sensor SR_s which generates noisy measurements according to the relation

$$SR_s : y_s(k) = C_s x_s(k) + V_s(k), \quad k \geq 0 \tag{2}$$

where $y_s(k) \in \mathbb{R}^{m_s}$ ($m_s \in \mathbb{N}_+$) is the process output, $V_s(k)$ is the measurement noise that is assumed to be white, Gaussian with zero mean and variance $\Sigma_s > 0$. The initial state and the noises $\{x_s(0), W_s(k), V_s(k)\}$ are assumed to be mutually independent $\forall s \in \mathcal{S}$ and $\forall k$. Further, these random variables are assumed to be mutually independent among all sources. Finally, we assume that each pair (A_s, C_s) is observable.

The encoder ENC_s uses the noisy measurements to generate transmission data and sends it to its corresponding destination using constant size packets. The packet size is assumed to be large enough to represent a real number with negligible quantization error. The data from ENC_s is received at the corresponding destination possibly with a stochastic delay τ_{sd} which models the transmission delay. Each destination comprises of a decoder DEC_d, that uses the received data to generate a state estimate that is optimal in the minimum mean squared error (MMSE) sense. We ignore any queuing delays in the network and assume the existence of a time stamp for every transmitted packet. When a destination receives a packet, it sends back an acknowledgment (*ACK*) to the corresponding source. We assume that *ACKs* are never lost in the network.

We employ the encoder and decoder scheme described in Gupta et al. (2009). At source s, denote the local estimate of state $x_s(k)$ given the measurements $\{y_s(j)\}_{j=1}^{k}$ by $\hat{x}_s(k)$. Further, denote the remote state estimate, produced by DEC_d at the corresponding destination d, by $\hat{x}_{rs}(k)$. The encoder and the decoder are given by

- ENC_s:
  - At each time slot k, calculate $\hat{x}_s(k)$ using (say) a Kalman Filter.
  - Transmit $\hat{x}_s(k)$ along with the time stamp k.
- DEC_d:
  - If $k = 0$, set the stored time stamp $t_d = -1$.
  - If DEC_d receives a packet in time slot k, extract the time stamp k' from the packet.