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Congestion Control in Communication Networks with Variable Discretization Period*

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Abstract. *In this paper, discrete flow control mechanism for connection-oriented telecommunication networks is proposed. In the considered system the feedback information about the current network state is delivered to the sources for the purpose of rate adaptation in discrete moments of time, however, the sampling period does not remain constant, but may change during the execution of the control process. The applied nonlinear algorithm allows eliminating data losses and ensures full usage of the available bandwidth. Since the rates generated by the controller are always nonnegative and bounded, the described strategy can be directly applied in real telecommunication systems.*

1. Introduction

Dynamically growing demand for fast data transfer in telecommunication channels, increasing number of users of wide area networks, and their stringent and each time more sophisticated requirements create the need to search for new technological solutions and upgrades to the existing concepts of data transmission. That is why in recent years much stress has been laid upon the improvements of mechanisms used for data transfer in connection-oriented communication networks.

The operation of networks of this type is based on the techniques of circuit and packet switching and, thanks to combining the advantages of both data transmission concepts, it has become a very attractive method of information

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interchange. Therefore, in the further part of the paper, the problem of flow control in the connection-oriented telecommunication networks will be addressed. The main focus of the presented analysis will be placed on the systems working similarly as the Available Bit Rate (ABR) service category in the Asynchronous Transfer Mode (ATM) technology, which allow exploitation of the feedback mechanism to regulate the transfer speed of the sources. The considered control scheme could be found useful also for the Multiprotocol Label Switching (MPLS) networks.

In recent years, the congestion control problem in the broadband connection-oriented networks has been the subject of many research projects and publications. The main goal of the presented algorithms and the ones currently developed is increasing the efficiency of data transmission. First such mechanisms used a single bit in the transferred cells to control the input rate into the ATM networks. According to the current system state (the buffer occupancy and the available bandwidth on the data path of the regulated flows) the bit was set either to increase the rate of a source or temporarily turn it off. Later on, if the conditions in the system changed, the input rate was modified accordingly. However, the effectiveness of the congestion avoidance methods based on a single bit only was far from satisfactory. Therefore, soon, a new scheme was proposed. The modified approach employed special control units – Resource Management (RM) cells – to exchange the feedback information between the sources and the network. The RM cells, emitted by each source once every N (usually $N = 32$) data cells, traveled along the established route interleaved with the data cells. The network nodes used the multibit Explicit Rate (ER) field in the RM cells to indicate more accurately the desirable transfer speed of the sources. The earlier congestion control schemes, which made use of this mechanism, such as BECN, DECbit, EFCI, PRCA, EPRCA, OSU, ERICA, were discussed by Jain in [7]. The algorithm developed by that author, namely Explicit Rate Indication for Congestion Avoidance (ERICA) [9], and its improved version ERICA+ [8], after substantial research, resulting in numerous publications was accepted by the ATM Forum (main standardization body for ATM) and incorporated into the official traffic management specification of the ATM technology [18]. The ERICA algorithm calculates the rate values on the basis of the current network load and the number of the established connections. It proves to be straightforward and cost-effective in the implementation, as it does not require complex mathematical computations. Moreover, it ensures good system operational conditions, and, according to the performed simulation tests, it usually (but not always) allows to satisfy *max-min fairness* criterion of the bandwidth allocation. The detailed description of the functional principle and crucial properties of the ERICA algorithm together with refinement and supplement to the earlier analysis was given by Kalyanaraman *et al.* in [10].

Interesting remarks on the unfavorable effect of the source rate synchronization occurring in some algorithms, that examine the queue length in the bottleneck node buffer, were given by Jae Yong Lee *et al.* in [12]. The proposed solution to the synchronization problem, derived on the basis of

Enhanced Proportional Rate Control Algorithm (EPRCA), involved a scheme of probabilistic detection of bottlenecks in the network. A detailed, theoretic approach for the design of data rate controller for ATM networks was presented by Imer *et al.* in [5]. The authors considered various aspects of the congestion control, such as the model of control information distribution, efficiency assessment criteria of the algorithms, the structure and operation of the ATM switch, and available bandwidth allocation among the regulated flows. The new, stochastic algorithms proposed in [5], determine the transfer speed in the process of the objective function minimization. In paper [11] Laberteaux *et al.* applied adaptive methods to control the transfer speed of the sources. The authors demonstrated that their strategy results in better utilization of the network resources and faster convergence to the steady state than other algorithms described earlier in literature. Another interesting approach to control the network congestion in the connection-oriented networks was given by Quet *et al.* in [17]. The flow control problem in the bottleneck node supporting multiple connections was solved by minimizing the H_∞ norm. It was shown that the proposed method of source rate calculation ensures system stability, even if the transfer delays of the considered connections are not known in advance and may change during the control action. Also a neural network concept was exploited to control the transfer speed of the sources [6]. Jagannathan and Talluri demonstrated stable operation of their neural network controller and discussed the possibility to obtain favorable Quality of Service (QoS) in the system. Similarly, in literature, we can find examples of classical PD [13], PID [3] and fuzzy PID [16] controllers in papers published so far.

Due to possibly long propagation delays in the network, the Smith predictor was employed to control the input rate of the system [1, 2, 4, 14, 15]. In paper [14] Mascolo considered the single virtual connection control problem. In the described algorithm, the rate was calculated on the basis of the comparison of the current queue length with its demand value, reduced by the number of the ‘in-flight’ cells (those cells which were delivered to the network in response to the control signal, but not yet reached the bottleneck node due to large propagation delay). The presented strategy eliminates the risk of buffer overflow at the congested switch, guarantees full resource utilization and fast convergence of the queue length to the steady state. The same author extended the idea of the Smith prediction to control the network supporting multiple data flows [15]. Further research on the congestion control in the ABR mode was conducted by Gómez-Stern *et al*. In paper [4] they proposed an algorithm combining the Smith principle with the PI controller. The suggested scheme allows for decreasing of the average queue length and queue length sensitivity to the available bandwidth. On the other hand, in [1], nonlinear algorithms for the flow regulation in the connection-oriented networks were suggested. The presented methods guarantee maximum throughput in the considered system and no data loss even though the propagation delays of the controlled connections are not known exactly, but can be determined by the controller placed at the bottleneck node only with limited degree of accuracy. Furthermore, the discussed nonlinear strategies preserve the advantages of earlier solutions, but impose smaller memory requirements on the nodes than similar algorithms described elsewhere in literature. Since modern congestion control schemes are usually realized as discrete systems, in paper [2], a Smith predictor based strategy was adapted to the network with nonzero period of feedback signal availability. The conditions were formulated, which allow to prevent the buffer overflow and, at the same time, to guarantee full resource utilization at the output connection of the congested node.

The approach presented in this paper applies the Smith predictor combined with the proportional controller with saturation to control the transfer speed of the sources in the network supporting multiple connections. On the contrary to the similar results published earlier [1, 2, 4, 14, 15], the strategy described here gives special consideration to variable period of control information availability in the system. The applied scheme eliminates the threat of data loss resulting from the congestion and ensures full resource usage, despite possible lack of synchronization in the feedback signal distribution. Furthermore, as the rates generated by the controller are always nonnegative and limited, the proposed mechanism can be feasibly incorporated in real telecommunication systems.

The remainder of this paper is organized in the following way. In Section 2, detailed description of the network model is given. Afterwards, in Section 3, the nonlinear control strategy is presented, and the features of the algorithm explicitly taking into account aperiodic nature of feedback distribution are discussed and strictly proved. Later on, the favorable features of the proposed scheme are verified by a simulation example, described in Section 4. The paper concludes with a brief summary of the presented strategy, given in Section 5.

2. Network model

We consider a connection-oriented network, consisting of sources, intermediate nodes and destinations. Every node has a separate buffer allocated for each output interface. The switches operate in the store-and-forward mode without the traffic prioritization. It means that when a data packet is received at the node input link, it is directed to the buffer of the appropriate output interface, where it waits in the First In First Out (FIFO) queue, to be relayed to the next node on the established path or to be sent to the destination (if the considered node is the last on the transfer route). If the switch receives a packet, which should be forwarded by the interface, whose buffer is entirely filled with data, the incoming packet must be discarded, and retransmitted afterwards. As a consequence of the congestion, the part of the available bandwidth is consumed for retransmissions, and efficiency of the resource utilization drops.

In a general case n Virtual Connections (VCs) will pass through the output link of the bottleneck node, each originating at a different source. The decision, whether a new flow is accepted by the network (and whether it can be handled by the switch) is taken by the admission control at the connection set-up phase together with the parameter values negotiation. It is assumed, that only a single node is the bottleneck for the considered set of VCs.

The network delivers the feedback information to the sources in control units (in the ATM technology these are the RM cells). Each node can read from and write the feedback information into the control units. As these units are given priority before the data packets (they are not queued in the buffers, but served immediately by the switches), Round Trip Time (*RTT*) remains constant for the duration of the connection for each flow. For the j-th (j = 1, 2, 3,.., n) virtual circuit we can state

$$RTT_j = T_{fj} + T_{bj} = const \tag{1}$$

where T_{fj} is forward propagation delay (delay from source j to the bottleneck node) and T_{bj} is backward propagation delay (delay from the bottleneck node to the destination and back to source j).

The VCs are numbered in the following way

$$RTT_1 \leq RTT_2 \leq \ldots \leq RTT_{n-1} \leq RTT_n \tag{2}$$

where $n = const$ is the total number of flows participating in the control process.

It is assumed that the control units are generated by each source once every N data packets. As the source adjusts the transmission speed only at the control unit arrival (it is a specific moment, when the source has meaningful information about the current network state), in most cases, the rate is modified at irregular time intervals. However, the source generates the control unit not less frequently than once every T_{rm}, where $T_{rm} < RTT_1$ is a real constant. Moreover, it is assumed that source j emits the control units since $-T_{fj}$ time instant.

Another important issue is the determination of moments of time when the rate is to be generated at the bottleneck node. In the considered model, as it is assumed, that the sources cannot be synchronized, the algorithm calculates the overall transfer speed for all VCs at the t_k time instants (k = 0, 1, 2,..., and $t_0 = 0$), while the time interval between consecutive rate generations may vary.

The queue length in the bottleneck node buffer at the t time instant will be denoted by $x(t)$. Since the buffer is empty before the connection establishment phase, $x(t < 0) = 0$. For any moment of time $t \geq 0$, the length of the queue can be calculated from the following relation

$$x(t) = \sum_{j=1}^{n} \int_0^t a_j(\tau - T_{fj}) d\tau - \int_0^t h(\tau) d\tau \tag{3}$$

The aggregate rate calculated by the controller at the t time instant is denoted by $a(t)$, while the transfer speed of source j at the t instant by $a_j(t)$. The maximum rate of any source cannot exceed $a_{\max} / n$. Moreover, it is assumed that $\forall j\ a_j(t < T_{bj}) = 0$.

The available bandwidth at the output link of the congested node at the t moment of time will be denoted by $d(t)$, and the utilized bandwidth by $h(t)$. If at the t time instant there are packets in the buffer ready to be transferred at the bottleneck link, $h(t) = d(t)$. Otherwise, i.e. when the buffer is empty, the outgoing transmission speed $h(t)$ equals the incoming rate. Thus,

$$0 \le h(t) \le d(t) \le d_{\max} \tag{4}$$

where $d_{\max}$ denotes the maximum available bandwidth.

In the subsequent part of the paper, a new flow control strategy for the network with variable sampling period will be presented and its crucial properties will be proved.

3. Control algorithm

The algorithm proposed in this section adjusts the input load to the network (and the congested switch) by regulating the rate of multiple sources according to the present system condition. The transfer speed is calculated at discrete moments of time t_k (recall $k = 0, 1, 2,...,$ and $t_0 = 0$). Although the time separation between the consecutive time instants t_k does not have to be constant, it is assumed that it cannot exceed $T_{\max}$, i.e.

$$\underset{k=0,1,2,...}{\forall} t_{k+1} - t_k \le T_{\max} \tag{5}$$

where $T_{\max}$ is a positive real value subject to the following constraint

$$\underset{j=1,...,n}{\forall} T_{\max} < RTT_j \tag{6}$$

Let us define a function W

$$W(t_k) = K\left[x_d - x(t_k) - A(t_k)\right] \tag{7}$$

where $K > 0$ is the controller gain and $x_d > 0$ is the demand queue length. The component $A(t_k)$ represents the 'in-flight' data, i.e. the packets, which have not appeared yet at the bottleneck node until t_k time instant, but will arrive due to the command calculated and sent to the sources by the controller within the last RTT. The amount of the 'in-flight' data can be determined from the relation given below

$$A(t) = \sum_{j=1}^{n} \int_{t}^{t+RTT_j} a_j(\tau - T_{fj}) d\tau \qquad (8)$$

As the history of the previously assigned transfer speed values is always accessible at the node, calculation of quantity (8) poses no difficulty. However, it should be pointed out that storing the source rates (allocated within the last *RTT*) together with the moments of control unit arrivals may consume some of the switch operational memory.

The rate $a_j(t)$ is recorded and sent for source j at the instant of the control unit passing through the congested node as the n-th share of the aggregate transfer speed, while the total rate is determined from the following formula

$$\underset{t \in [t_k, t_{k+1})}{\forall} \quad a(t) = a(t_k) = \begin{cases} 0, & \text{if } W(t_k) < 0 \\ W(t_k), & \text{if } 0 \le W(t_k) \le a_{\max} \\ a_{\max}, & \text{if } W(t_k) > a_{\max} \end{cases} \qquad (9)$$

where $a_{\max} > 0$ denotes the upper saturation limit.

Theorem 1

For any time instant $t \geq 0$ the queue length in the bottleneck node buffer is upper bounded by the following inequality

$$x(t) < x_d + a_{\max}\left(T_{\max} + T_{rm}\right) \qquad (10)$$

Proof

Let us consider a time instant $t \geq 0$. Let y be the biggest index, such that $t_y \leq t$, where t_y is the moment of rate calculation by the controller at the congested switch. Then, we can write $t = t_y + \delta$, where δ denotes a real value from the interval $[0, T_{\max})$. As the transfer speed is calculated at discrete moments of time, it is necessary to investigate the values of the function W at the instant t_y. We will consider two cases: first the situation when $W(t_y) < 0$, and afterwards, the circumstances when $W(t_y) \geq 0$.

Case 1: We analyze the situation when $W(t_y) < 0$. Let us determine the biggest index z such that $t_z < t$ and t_z is the (last) moment of time before t, when the function W was positive. Notice that

$$W(0) = Kx_d > 0 \qquad (11)$$

and the t_z instant actually exists. The value of the function W at the t_z time instant can be calculated from the relation given below

$$W(t_z) = K[x_d - x(t_z) - A(t_z)] > 0 \tag{12}$$

Performing algebraic manipulations on inequality (12), we get

$$x(t_z) < x_d - A(t_z) \tag{13}$$

Using relation (13), we may express the queue length at the bottleneck node at the t time instant in the following way

$$\begin{aligned} x(t) &= x(t_z) + B(t_z, t - t_z) - \int_{t_z}^{t} h(\tau)d\tau < \\ &< x_d - A(t_z) + B(t_z, t - t_z) - \int_{t_z}^{t} h(\tau)d\tau \le x_d + B(t_z, t - t_z) - A(t_z) \end{aligned} \tag{14}$$

where $B(\tau, \alpha)$ denotes the amount of the 'in-flight' data, which arrived at the congested node within $[\tau, \tau + \alpha)$ interval. In expression (14), the value of the component

$$B(t_z, t - t_z) - A(t_z) \tag{15}$$

reflects the difference between the data arriving from t_z to t and the 'in-flight' data at the t_z time instant. As the number of the incoming packets in the interval $[t_z, t)$ cannot exceed the sum of the 'in-flight' data at the t_z time instant and the data, for which the rate was assigned in the interval $[t_z, t)$, quantity (15) can be estimated as follows

$$B(t_z, t - t_z) - A(t_z) \le \sum_{j=1}^{n} \int_{t_z}^{t} a_j(\tau + T_{bj})d\tau \tag{16}$$

Notice that the algorithm generates the nonzero rate at the t_z instant for the last time before t. Afterwards, at the $t_{z+1} > t_z$ the transfer speed, according to (9), is equal to zero. Consequently, in the interval $[t_{z+1}, t_z)$ the rate incorporated into the control units is also equal to zero. As the rate calculation by the controller and the rate recording into the control units does not have to occur at the same moment of time, the amount of the 'in-flight' data may increase after t_{z+1} despite the overall rate value $a(t_{z+1})$ remaining at the zero level. Nevertheless, the time period between t_{z+1} time instant and the moment of the arrival of the control unit belonging to the j-th flow is not bigger than T_{rm}. Therefore,

$$B(t_z, t-t_z) - A(t_z) \le \sum_{j=1}^{n} \int_{t_z}^{t} a_j(\tau + T_{bj}) d\tau \le$$
$$\le a_{\max} T_{\max} + \sum_{j=1}^{n} \int_{t_{z+1}}^{t_{z+1}+T_{rm}} a_j(\tau + T_{bj}) d\tau \le a_{\max}\left(T_{\max} + T_{rm}\right) \quad (17)$$

Substituting (17) into (14), we get

$$x(t) < x_d + a_{\max}\left(T_{\max} + T_{rm}\right) \quad (18)$$

This concludes the first part of the proof.

Case 2: Now, we will consider the situation, when $W(t_y) \ge 0$. On the basis of (7), we can write

$$W(t_y) = K\lfloor x_d - x(t_y) - A(t_y) \rfloor \ge 0 \quad (19)$$

Term rearrangement yields

$$x(t_y) \le x_d - A(t_y) \quad (20)$$

Using inequality (20), we can present the queue length at the bottleneck node at the $t = t_y + \delta$ time instant in the following way

$$x(t) = x(t_y) + B(t_y, \delta) - \int_{t_y}^{t} h(\tau) d\tau \le x_d - A(t_y) + B(t_y, \delta) - \int_{t_z}^{t} h(\tau) d\tau \quad (21)$$

Relation (6) implies that for each $j = 1,...,n$ RTT_j is bigger than $T_{\max}$, and hence, is longer than δ. Thus, the amount of data arriving at the congested node in the interval $[t_y, t_y + \delta)$ cannot be greater than the number of the 'in-flight' packets at the t_y time instant. In consequence, $A(t_y) - B(t_y,\delta) > 0$, and we obtain the following evaluation

$$x(t) \le x_d - A(t_y) + B(t_y, \delta) - \int_{t_y}^{t} h(\tau) d\tau < x_d < x_d + a_{\max}\left(T_{\max} + T_{rm}\right) \quad (22)$$

This ends the proof.

Theorem 1 specifies the upper-bound of the queue length in the bottleneck node and indicates the minimum buffer space to be reserved for the data storage, which will ensure zero packet loss. Another desirable property of the considered flow control scheme is the maximum resource exploitation. The theorem presented in the sequel defines the condition, which will guarantee that the available bandwidth at the output link of the congested switch is entirely utilized for the data transfer.

Theorem 2

If the maximum rate $a_{\max} > d_{\max}$ and the demand queue length satisfies the following inequality

$$x_d > a_{\max}\left(\sum_{j=1}^{n}\frac{1}{n}RTT_j + \frac{1}{K}\right) + d_{\max}\left(T_{\max} + T_{rm}\right) \tag{23}$$

then for any $t > RTT_n + \psi$, where

$$\psi = \frac{a_{\max}}{a_{\max} - d_{\max}}T_{rm} \tag{24}$$

the queue length in the bottleneck node buffer is always positive.

Proof

We can represent relation (23) in the following form

$$x_d = \theta + a_{\max}\left(\sum_{j=1}^{n}\frac{1}{n}RTT_j + \frac{1}{K}\right) + d_{\max}\left(T_{\max} + T_{rm}\right) \tag{25}$$

where θ is some positive constant. Let us define additional function of time

$$\varphi(t) = x(t) + A(t) \tag{26}$$

The value of the function φ represents the sum of data waiting in the queue to be transferred at the output connection of the bottleneck node and the amount of the 'in-flight' data at the t time instant. Applying (3) and (8), we can express the function φ in the following way

$$\begin{aligned}\varphi(t) &= x(t) + A(t) = \sum_{j=1}^{n}\int_0^t a_j(\tau - T_{fj})d\tau - \int_0^t h(\tau)d\tau + \\ &+ \sum_{j=1}^{n}\int_t^{t+RTT_j} a_j(\tau - T_{fj})d\tau = \sum_{j=1}^{n}\int_0^{t+RTT_j} a_j(\tau - T_{fj})d\tau - \int_0^t h(\tau)d\tau\end{aligned} \tag{27}$$

For $k = 0$ the value of the function $\varphi(t_k)$ is equal to

$$\varphi(t_k) = \varphi(0) < x_d - a_{\max} / K \tag{28}$$

Let t_v be some moment of time, when the controller generates the overall rate $a(t_v)$. If the inequality $\varphi(t_v) \le x_d - a_{\max} / K$ is fulfilled, i.e. after taking into account (25), if the following relation holds

$$\varphi(t_v) \le \theta + a_{\max} \sum_{j=1}^{n} \frac{1}{n} RTT_j + d_{\max} (T_{\max} + T_{rm}) \tag{29}$$

then from (7) and (26) we obtain

$$W(t_v) = K[x_d - \varphi(t_v)] \ge K\left[\frac{a_{\max}}{K}\right] = a_{\max} \tag{30}$$

Relations (9) and (30) imply that the rate determined by the controller at the t_v time instant is equal to $a_{\max}$. This leads to the conclusion, that whenever the value of the function φ is smaller than or equal to $x_d - a_{\max} / K$, the algorithm generates the maximum transfer speed. On the basis of this observation, we will demonstrate, that the function φ attains the value $a_{\max} \sum_{j=1}^{n} RTT_j / n$ in the interval $(0, RTT_n + \psi]$.

Notice that for any t smaller than RTT_1 the function $h(t) = 0$, for any t from the interval $[RTT_1, RTT_2)$ the function $h(t) \le a_{\max} / n$, for any t from the interval $[RTT_2, RTT_3)$ the function $h(t) \le 2a_{\max} / n$, etc. Finally, for any t from the interval $[RTT_{n-1}, RTT_n)$ the function $h(t) \le (n - 1)a_{\max} / n$.

Therefore, we may conclude that for any t smaller than T_{rm} the value of the function φ does not decrease, for any t from the interval $[T_{rm}, RTT_1)$ the function φ increases at the rate $a_{\max}$, for any t from the interval $[RTT_1, RTT_2)$ the function φ increases at a rate at least equal to $a_{\max}(n - 1) / n$, for any t from the interval $[RTT_2, RTT_3)$ the function φ increases at a rate at least equal to $a_{\max}(n - 2) / n$, and so forth. Finally, for any t from the interval $[RTT_{n-1}, RTT_n)$ the function φ increases at a rate not smaller than $a_{\max} / n$. On the other hand, if for any $t > RTT_n$ the function $\varphi(t) \le x_d - a_{\max} / K$, then it increases at a rate at least equal to $a_{\max} - d_{\max}$. This implies that

$$\varphi(t) \ge f(t) \tag{31}$$

with the function $f(t)$ defined for any $t \in [RTT_m, RTT_{m+1})$, where $m = 0, 1, \ldots, n-1$ and $RTT_0 = 0$, as

$$f(t) = a_{\max} \sum_{j=0}^{m} \frac{1}{n} RTT_j + \frac{n-m}{n} a_{\max} t - a_{\max} T_{rm} \tag{32}$$

For any $t > RTT_n$, in turn,

$$f(t) = a_{\max} \sum_{j=1}^{n} \frac{1}{n} RTT_j + (a_{\max} - d_{\max}) \cdot (t - RTT_n) - a_{\max} T_{rm} \tag{33}$$

From the definition of the function $f(t)$ and inequality (31) we may conclude, that at some time instant t belonging to the interval $(0, RTT_n + \psi]$, the function φ reaches the level of $a_{\max} \sum_{j=1}^{n} RTT_j / n$. Moreover, the value of the function φ at the time instant $t = RTT_n + \psi + \varepsilon$, where $\varepsilon \to 0$, is greater than $a_{\max} \sum_{j=1}^{n} RTT_j / n$.

The presented reasoning implies that, if inequality (29) is fulfilled, then $a(t_v) = a_{\max}$. Hence, the rate $a(t_v)$ is smaller than $a_{\max}$ only if

$$\varphi(t_v) > \theta + a_{\max} \sum_{j=1}^{n} \frac{1}{n} RTT_j + d_{\max} (T_{\max} + T_{rm}) \tag{34}$$

Notice that the function φ cannot decrease faster than at the $d_{\max}$ rate. Moreover, when relation (29) is satisfied, then at some time instant from the interval $[t_v, t_v + T_{\max} + T_{rm})$ the value of φ will begin to grow. This means that the function φ, after exceeding the level of $a_{\max} \sum_{j=1}^{n} RTT_j / n$, will never fall to this value again. So we may write

$$\underset{t > RTT_n + \psi}{\forall} \varphi(t) > a_{\max} \sum_{j=1}^{n} \frac{1}{n} RTT_j \tag{35}$$

Transforming (26) and using (8) and (35), we obtain

$$x(t) = \varphi(t) - A(t) > a_{\max} \sum_{j=1}^{n} \frac{1}{n} RTT_j - A(t) \geq 0 \tag{36}$$

If inequality (23) is fulfilled and the maximum transfer speed $a_{\max} > d_{\max}$, then relation (36) directly implies that the queue length in the bottleneck node buffer for any $t > RTT_n + \psi$ is strictly positive. This ends the proof of Theorem 2.

4. Simulation results

Theoretical considerations concerning the properties of the proposed control strategy were verified by a simulation example performed in Matlab-Simulink® design environment. We considered a system consisting of three connections (n = 3) characterized by round trip times equal respectively to 20 ms, 40 ms and 70 ms. The maximum available bandwidth was set as $d_{\max} = 9000$ packets/s, and the maximum rate as $a_{\max} = 10000$ packets/s. It means that each source had a chance to transmit data at the maximum rate of 3333 packets/s assuming equal resource allocation. The total simulation time was adjusted to 4 s. The bandwidth available for the connections is depicted in Fig. 1.

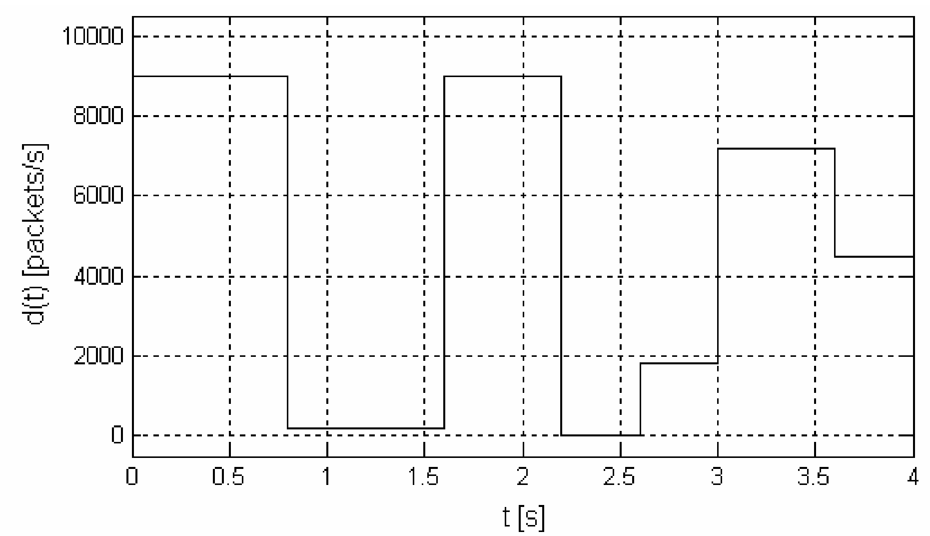

Fig. 1. Available bandwidth at the output link of the bottleneck node.

Moreover, it was assumed that each source probes the network with a control unit once every $N = 32$ equal size data packets, but not less frequently than once every $T_{rm} = 30$ ms. The controller determines the total rate at discrete moments of time with the maximum period between consecutive calculations $T_{\max}$ = 10 ms. The controller gain was adjusted to $K = 60$ s^{-1}. Theorem 2 suggests that in order to obtain full resource exploitation, the demand queue length should be bigger than 960 packets. The value of $x_d = 970$ packets was chosen. The queue length should never exceed the value of 1370 packets, as indicated by Theorem 1. Therefore, in order to eliminate data losses in the considered network, it should suffice to reserve the memory volume for 1370 packets at the congested node.

Figure 2 illustrates the queue length in the congested switch buffer, as obtained after applying the algorithm described in Section 3. The rate generated by the controller is shown in Fig. 3.

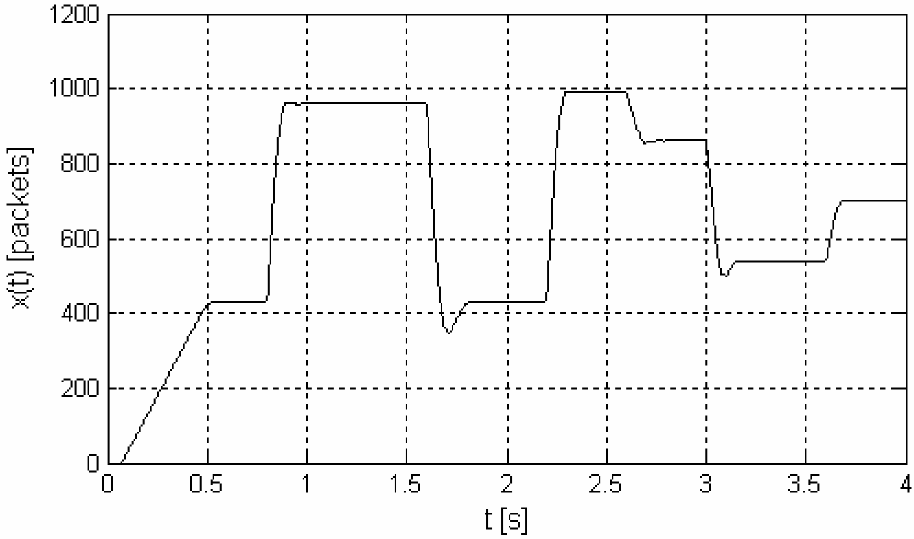

Fig. 2. Queue length in the bottleneck node buffer.

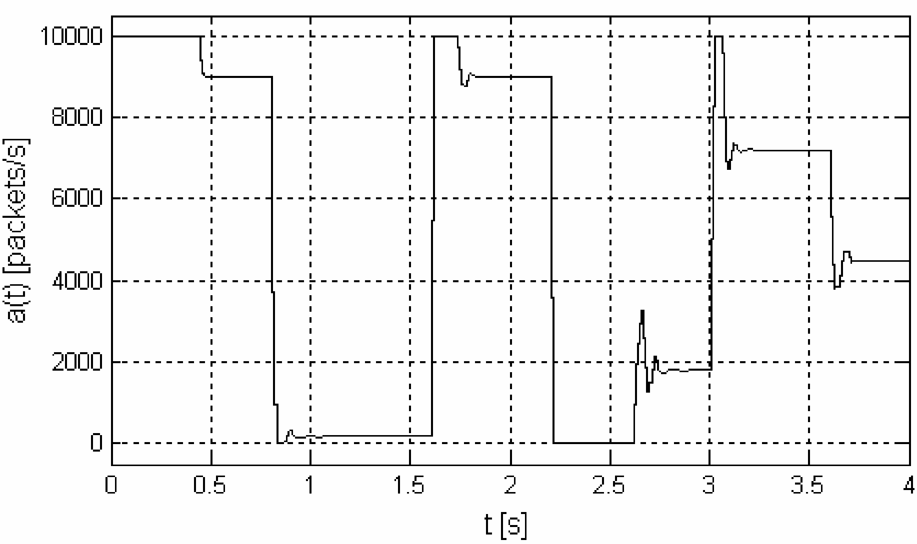

Fig. 3. Total rate generated by the controller.

As we can see, the queue never grows beyond the level of 1370 packets, which ensures no data loss in the network. We can also notice that the queue does not drop to zero. These mean that apart from the buffer space preservation, the designed algorithm ensures the maximum throughput at the output link of the congested node.

5. Conclusions

In this paper a new mechanism of flow control for the connection-oriented telecommunication networks was presented. The proposed strategy combines the Smith predictor with the nonlinear element. With appropriate buffer space allocation (the value is explicitly stated in the article) the described algorithm guarantees efficient usage of the network resources, as all of the available bandwidth at the output connection of the bottleneck node is consumed for the packet transmission. Moreover, the proposed mechanism proves to be especially useful for the critical data transfer (e.g. banking transactions), since it eliminates the risk of losses in the controlled connections. The favorable properties of the considered algorithm are obtained even though the feedback information is accessible at the sources for the purpose of rate adaptation at irregularly spaced time instants and aggregate transfer speed calculation is performed by the controller aperiodically. The analysis of the variable sampling period of the source rate adjustment deserves a special attention, as it closely relates the considered model to the conditions present in real telecommunication systems.

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