Design and preliminary study of the W-PRDR: A new Congestion Control Scheme for Wireless Networks

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ABSTRACT
This work presents the design and preliminary performance evaluation of a new congestion control mechanism for multimedia applications over wireless networks: the Wireless Proportional and Derivative Algorithm (W-PRDR). W-PRDR is based on the exchange of RTCP reports to feed the sources with the supported rate so that they can adapt their transmission rate according to the loss state and the allowed fair share bandwidth in the network.

In order to fit to a wireless environment, we enhanced the original PRDR algorithm with a loss discrimination scheme to distinguish between congestion losses and random errors due to wireless transmission. We show through simulation the capacity of the W-PRDR mechanism to improve the transmission rate under different simulated network topologies.

Categories and Subject Descriptors
C.2.1 [**Network Architecture and Design**]: Wireless communication.

General Terms
Performance.

Keywords
Wireless networks, congestion control, UDP, multimedia traffic.

1. INTRODUCTION
The demand on multimedia services over wireless networks is growing incessantly. To benefit from the best allowed traffic transmission rate, multimedia applications have to be constantly responsive to the observed network conditions and have therefore to adapt dynamically their transmission rate. In most cases, the variables of interest, like loss rate, delay and jitter are estimated with an end-to-end approach. Indeed, a high packet loss rate denotes network congestion, and therefore, a source has to decrease its sending rate responding to such network condition. Unfortunately, end-to-end feedback is not fitted to wireless multi-hop networks since unlike wired ones, wireless networks are characterized by high random losses due to errors at the wireless links, signal fading, shadowing and interferences. Thus, a loss-based policy implies that the sources reduce their sending rate unnecessarily, resulting in network under-utilization and low performance.

There have been several studies [1, 2, and 3] dealing with TCP behavior in a wireless environment which have shown its degraded performance due to mistaking wireless random loss for congestion. For instance, TCP-derived rate-control protocols, like TFRC [4, 5], may show a significant degradation of performance over wireless links.

Many schemes have been proposed to improve TCP performance in hybrid wireless and wired networks [6-12]. The schemes can be classified as four main categories: pure end-to-end, split-connection, explicit notification and link-layer schemes. Among them, end-to-end discrimination schemes maintain the original end-to-end semantics of TCP congestion control, which only requires the minimal modifications at the TCP sender and/or receiver, without any changes from the intermediate nodes. Thus many researchers try to design a robust loss discrimination algorithm to distinguish the congestion losses from wireless random losses. Relative one-way trip times (ROTT) [13], packet inter-arrival times, known as the Spike scheme [14], RTT variation statistics [7, 10, 12] and hybrid schemes [15], the so-called ZigZag scheme, have been used for such purpose. Biaz and al. [9, 10] claimed that RTT-based congestion avoidance discrimination schemes do not work well sometimes.

Parsa and Garcia-Luna-Aceves [16] proposed a variant of TCP that uses a state machine that changes TCP's congestion window size based on RTT variations. Liu and al. [17] proposed a scheme using Loss Pairs and Hidden Markov Modeling techniques. This work also uses the changes on RTT over time to infer the cause of losses. Many researches have also attempted to make improvements on TCP so that it can work better in wireless ad hoc networks [18].

The above type of loss differentiation can be tricky and unreliable over wireless networks, because large delay fluctuations are inherent in such types of networks. Moreover, some of the previously-cited works report inaccuracies in these differentiators [19].

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Since the UDP protocol (User Datagram Protocol) is more suited to carry real-time and multimedia traffic on the Internet, UDP connections should be enhanced with a mechanism to detect or react to network congestion, especially when using UDP over a lossy, congested multi-hop network running the 802.11 MAC [20]. In fact, end-to-end and network feedback techniques should be combined in order to solve congestion problems, achieve high performance and reduce power dissipation.

Besides, such congestion control algorithms should be enhanced with loss discrimination schemes in order to distinguish between congestion and random losses.

In this work we focused on the two most popular loss differentiation schemes: the packet inter-arrival proposed in [9] and Spike proposed in [13] to design the W-PRDR algorithm (Wireless Proportional and Derivative) as a new congestion control scheme for wireless networks with loss discrimination. W-PRDR is a pure end-to-end mechanism, in the sense that it doesn't allow any kind of signaling from the routers back to the traffic sources. W-PRDR is a rate based mechanism built on the RTP/RTCP protocol [21] and built basically on the PRDR, a rate control algorithm proposed in [22] for the mobile Internet. W-PRDR operates as follows: W-PRDR routers calculate a fair share along the path of the connection. This information is fed back to senders in RTCP reports. Each senders has to adapt its transmission rate according to the loss situation observed in the network.

In order to fit this mechanism to adapt the transmission rate of RTP-based multimedia flows to congestion conditions in wireless networks, we enhanced it using a loss discrimination scheme. Therefore, W-PRDR is able to distinguish between losses due to a congested network (congestion loss) and losses occurring due to the temporary interference of channel disturbances (random loss). We investigate the use of the Spike-train scheme [9] for loss differentiation under various network conditions.

The rest of this paper is organized as follows: in Section 2, we recall the context of the PRDR algorithm, formerly proposed for wired networks. In Section 3, we sum up the two loss discrimination schemes introduced above: the inter-arrival and Spike-train schemes. Section 4 details the implementation of the W-PRDR algorithm in a wireless network: Spike-W-PRDR. In Section 5, we detail the W-PRDR preliminary performance evaluation, we introduce so the simulation model along with the configuration parameters, then we discuss and analyze the results. Section 6 summarizes this work and gives some perspectives.

2. Background: The PRDR Algorithm

2.1 Basic Concepts

The PRDR algorithm has been proposed for the flow control in ABR-service in ATM networks [23] and is based on a concept originally introduced for packet-switched networks [24]: a switch computes the local supported bandwidth using a proportional and derivative controller, having in mind the current and beyond occupancy of the buffering queue.

In a previous work we have adapted the PRDR algorithm to RTP flows congestion control in the Internet [22]. We used the RTCP protocol to exchange control messages and feedback data. The comparative performance simulations have shown the good efficiency of the PRDR algorithm as well as its TCP-friendliness and its capacity to guarantee fairness among competing UDP flows while achieving a steady state in the network.

The PRDR operates as follows: a source transmits periodically a RTCP report that indicates its desired transmission rate R_d. Each gateway on the connection path modifies the R_d value according to a locally computed fair share rate Q and forwards the control message to the next network node until it reaches the destination. The destination set up the fair share rate R_f according to the loss situation observed between two consecutive RTCP source reports: if no loss is detected, the transmission rate is increased using the additive increase scheme, otherwise, R_f is set to the bottleneck value Q. The destination in its turn transmits the updated information back to the source in a RTCP receiver report. Upon the receipt of this report, the source adjusts its transmission rate in accordance to the received transmitting rate value R_f. The W-PRDR algorithm uses the RTP/RTCP protocol to convey the control information not to introduce a new control protocol.

2.2 The PRDR controller

Figure 1 below shows the parameters of a PRDR gateway. For a given gateway i in the network, let $x_i(n)$ be the average buffer occupancy of this gateway at time n and let x_i^0 be a fixed buffer threshold.

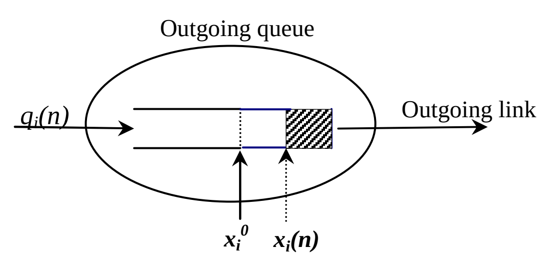

Figure 1. A PRDR controller.

Then, to compute the admission rate at time n, $q_i(n+1)$, the gateway uses a proportional and derivative controller which is defined by the following equation:

$$q_i(n+1) = Sat_{q^0}\Big\{ q_i(n) - \sum_{j=0}^{J} \alpha_j (x_i(n-j) - x_i^0) - \sum_{k=0}^{K} \beta_k q_i(n-k) \Big\} \quad (1)$$

The saturation function is defined as follows:

$$Sat_a(z) = \begin{cases} 0 & if\ z < 0 \\ a & if\ z > a \\ z & \text{otherwise} \end{cases}$$

The saturation rate q^0, is computed using the following formula:

$$q^0 = \frac{\text{Target throughput}}{\text{Number of limited connections}} \quad (2)$$

Detailed studies in [24] have established that the control parameters α_j and β_k have to fulfil the following conditions:

$$\sum_{j=0}^{J} \alpha_j > 0, \ \sum_{k=0}^{K} \beta_k = 0$$

to ensure the system stability.

For the scope of our work, we started with a simplified implementation of the equation (1). Hence, we took into account only the first raw derivative component of the controller defined in this equation which becomes:

$$q_i(n+1) = Sat_{q^0}\{q_i(n) - \alpha_0\,(x_i(n) - x_i^0)\}, i \in \aleph \qquad (3)$$

The rate calculation is performed every time a control packet passes the gateway, and the rate computed at the time n, given by the value of $q_i(n+1)$ is carried in this control message. The minimum value of all the $q_i(n+1)$ on the connection path is considered as the bottleneck throughput and assigned by the destination to the field R_f in the RTP/RTCP message

3. Loss discrimination schemes

Congestion control mechanisms make usually the assumption that packet losses are caused by buffer overflows and try to relax congestion by reducing the source sending rate. Such a behavior works well and fairly in wired networks but this is not the case in wireless networks since losses are often induced by link errors (signal fading, interferences…). A straightforward solution to prevent this misbehaving is to deploy a loss differentiation mechanism.

In this section we summarize the basic loss differentiation schemes: the inter-arrival scheme and the Spike scheme.

3.1 Inter-arrival scheme

The scheme proposed in [9] uses packet inter-arrival time to differentiate error losses from congestion losses. As illustrated in figure 2 (a), this scheme works as follows:

- Let T_{min} be the minimum packet inter-arrival time observed by the receiver during the connection.

- Let P_0 be an out-of-order packet received by the receiver and P_i the last in-sequence packet received before P_0. Let T_g then denote the inter-arrival time between P_i and P_0.

Finally, let m be the number of lost packets between P_i and P_0 (assuming that all packets are of the same size).

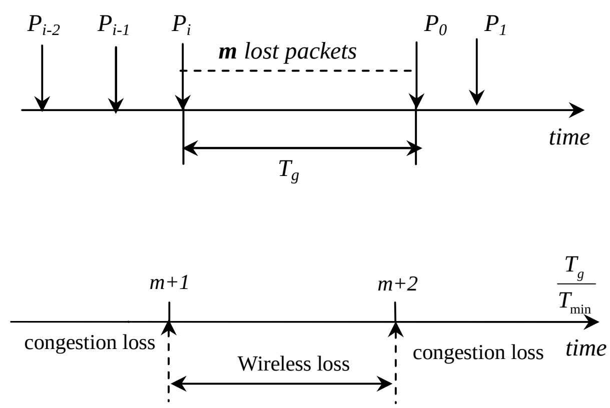

Figure 2. Inter-arrival loss discrimination scheme.

Figure 2 (b) shows that in order to discriminate losses, the inter-arrival scheme depicts that if $(m+1) \le \frac{T_g}{T_{min}} < (m+2)$ then the m missing packets are assumed to be lost due to wireless transmission errors, otherwise, they are assumed to be lost due to congestion

3.2 Spike-train scheme

The work in [13] proposes a loss discrimination algorithm based on the ROTT (*Relative One-way Trip Time*), the time needed by a packet to cross the network from a given sender to a given receiver. The Spike-train scheme set up a two-state Markov chain to differentiate random losses from congestion losses. Hence, if the connection is in a spike state, occurring losses are attributed to congestion, otherwise, they are considered to be random. As plotted in figure 3, the *ROTT* is used to identify the connection state

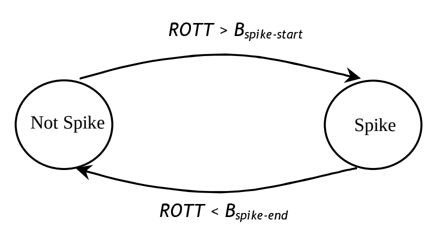

Figure 3. Spike-train loss discrimination scheme.

according to two thresholds $B_{spike\text{-}start}$ and $B_{spike\text{-}end}$.

The threshold values are given in [13] as:

$$B_{spike\text{-}start} = ROTT_{min} + \frac{1}{2}(ROTT_{max} - ROTT_{min})$$

$$B_{spike\text{-}end} = ROTT_{min} + \frac{1}{3}(ROTT_{max} - ROTT_{min})$$

where $ROTT_{min}$ and $ROTT_{max}$ are respectively the minimum and the maximum relative one-way trip time observed so far.

4. W-PRDR: the Wireless PRDR algorithm

The W-PRDR algorithm relies basically on the PRDR mechanism designed for wired networks. As we presented in section 2.1, the routers calculate a fair share along the path to the destination. This information is fed back to the source in the field R_f in the RTCP report along with a loss flag set up by the receiver. If the connection experiences some loss, the source reduces its rate to the received fair share, otherwise, it increments its rate with an additive increase rate A_i. Among the strengths of the W-PRDR mechanism, we highlight the mathematical framework which dictates how to select the values of the control gains α_j and β_k in order to ensure the system stability [25].

In this work we aimed to adapt the PRDR mechanism to UDP traffic control in wireless networks. In order to reach our purpose, we have to endow the end system procedure with a loss differentiation scheme to make it react only to congestion loss and ignore random ones. We implemented first the inter-arrival W-PRDR, then the Spike-train W-PRDR. Since the former didn't exhibit any significant result, we are presenting here the main results when the Spike-train W-PRDR is used as multimedia traffic control mechanism.

4.1 The Spike-train W-PRDR

Using the timestamp in the RTP packet, the receiver calculates the relative one-way trip time ROTT and determines whether it is or not in a Spike state as follows:

```
ROTT = now – t timestamp
//Determine SPIKE state
     if ((ROTT >= B_spike_start) &&
(!in_spike))
                in_spike = true;
         if ((ROTT <= B_spike_end) &&
(in_spike))
                in_spike = false;
```

When a gap between RTP sequence numbers is observed, the receiver sets its loss flag in the RTCP report only if it is in a Spike state, which means in a case of a congestion loss.

```
if (in_spike){
// congestion_LOSS;
             loss_flag = 1;}:
```

5. W-PRDR Preliminary Performance Study

5.1 Simulation model and parameters

To evaluate the efficiency of the W-PRDR algorithm in a wireless environment, the simulation tool network simulator (ns) version ns-2.31 [26] was used to implement a dumb-bell heterogeneous topology illustrated in figure 4.

The W-PRDR proportional and derivative controller was implemented both at the wired routers and the base station. The controller parameters (cf. equation (3)) were set to:

- x_0 = 15 packets
- α_0 = 0.8

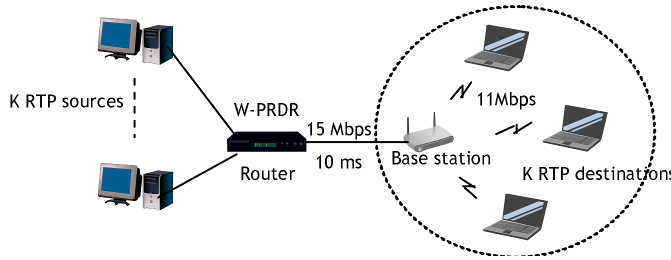

Figure 4. Heterogeneous Network topology.

The MAC layer implements the IEEE802.11 standard, giving a nominal bandwidth of 11Mbps. All the RTP sources share a wired bottleneck link L with a constant delay of 100 ms and a capacity C of 15 Mbps. The RTP packet size was set to 1kb, which is a size often used in video conferencing applications, and the simulation time was set to 400 s.

We used the DSDV (Destination Sequence Distance Vector) routing protocol for all the simulations.

5.2 W-PRDR scalability in a wireless environment

Through these tests we are interested to study the convergence of the W-PRDR algorithm and its ability to regularize the competing traffic flows on the shared wireless channel.

We conducted a series of simulations varying the number K of competing ingoing RTP flows and their initial rate R_0, in order to settle different congestion intensities (see table 1 below).

Table 1. Test configuration

K	R0	Configuration nb
10	400 kbps	1
20	200 kbps	2

Figure 5 shows the unfair allocation of the 802.11 wireless channel in presence of 10 competing RTP flows.

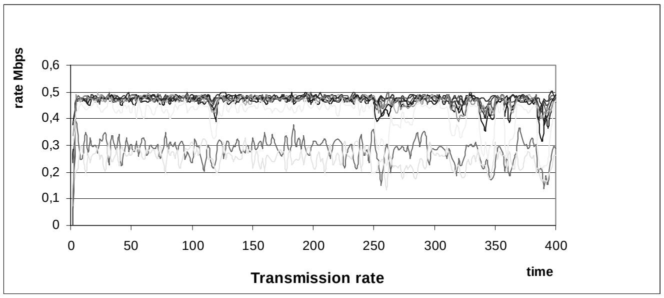

Figure 5. Channel allocation in presence of 10 flows.

When we introduced the W-PRDR controller at the base station's buffering queue, the channel became fairly shared between the 10 flows. Figures 6 and 7 depict respectively the channel allocation using the W-PRDR controller and the standard deviation measured between the 10 flows.

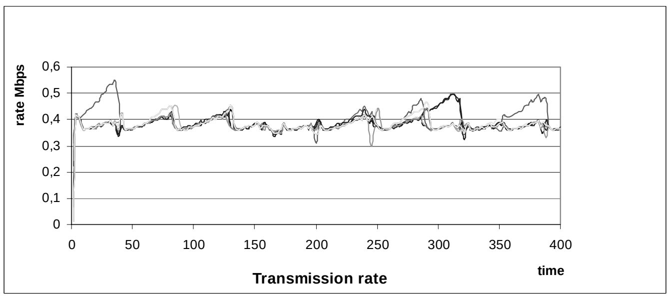

Figure 6. W-PRDR channel allocation.

The W-PRDR algorithm guaranties a mean standard deviation of 13 kbps, where it is of 88 kbps in absence of W-PRDR control.

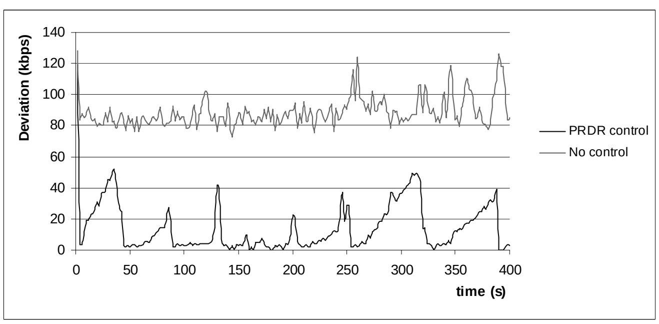

Figure 7. Instantaneous standard deviation of 10 flows.

In configuration 2, a similar result is obtained; figures 8 and 9 show the channel allocation respectively without/with use of the W-PRDR controller.

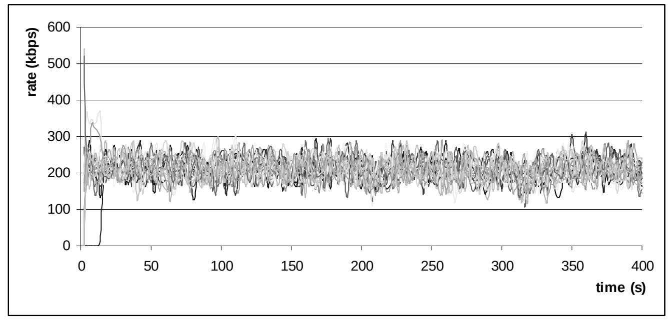

Figure 8. Channel allocation in presence of 20 flows.

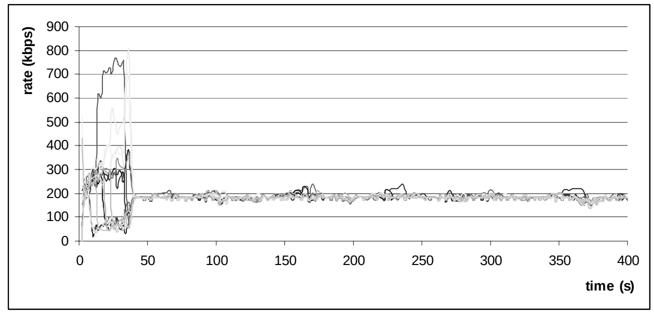

Figure 9. PRDR channel allocation.

As illustrated in figure 10, a mean deviation of 30 kbps is achieved in absence of control, and 6 kbps when using the W-PRDR controller.

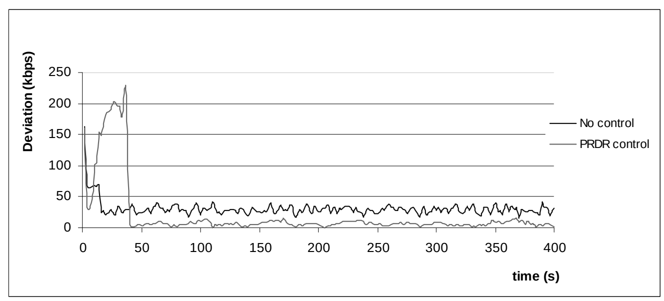

Figure 10. Instantaneous standard deviation of 20 flows.

We can conclude that the proposed algorithm exhibits an ability to solve the problem of unfairness that appears when sharing the wireless channel between a high number of competing flows (more than 7) while managing to regularize fairly the channel allocation among the competing RTP flows.

5.3 W-PRDR in presence of wireless loss

In this section, we present the result obtained from the series of simulations we carried to evaluate the ability of the W-PRDR algorithm to distinguish the cause of packet loss: error-due or congestion-due loss.

We kept the same network topology and traffic parameters as described in section 5.1 and we introduced an error model at the outcoming wireless interface of the base station so that the same error ratio is perceived by all the nodes in the wireless channel.

5.3.1 *Wireless error model*

The base station induces in the wireless channel random errors according to the time-based two state Markov chain model introduced by [27] and based on collected Wave-LAN error traces. This two state error model implements time-based error state transitions. Transitions to the next error state occur at the end of the duration of the current state. The next error state is then selected using the transition state matrix. In our simulations, we used a model where the error-free state has no errors and the error state has an error probability of P_e.

Figure 11 represents the transition probabilities between the model states. The error unit is the packet, the duration of the error-free and the error states are 5 and 10 respectively.

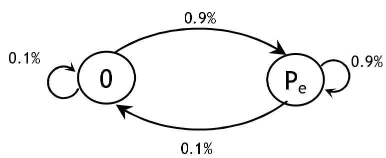

Figure 11. The two state error model.

To evaluate different scenarios, the error probability Pe in the channel is changed between 0.5% and 5%. These parameters are presented as low and high channel error rates (corresponding to packet loss rates of 0.5% and 5% respectively) [27]

The first 100 seconds were considered as an observation period. We used various configurations by tuning the number K of RTP receivers, the RTP initial rate R_0, and the loss probability P_e.

5.3.2 *Simulation results*

Configuration1: K=1, R_0= 4,5 Mbps, P_e =3%

In this case, when errors are perceived, the sender reduces its transmission rate to the fair share bandwidth which is 3.6 Mbps.

Figure 12 shows the sending rate of one RTP flow with and without use of Spike-train loss discrimination scheme in presence of random errors. As we can see from the plot, the sender adapts its transmission rate only in congested states, loss due to wireless errors have no effect. It is important to notice that the W-PRDR achieves a rate reaching the 4 Mbps. However, using PRDR without loss discrimination scheme the rate is close to 3.6 Mbps, i.e. W-PRDR achieves a rate increase of 11%.

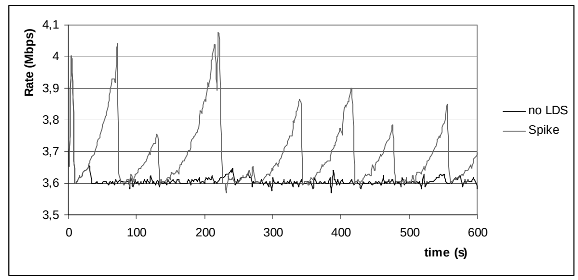

Figure 12. Configuration 1: 1 flow, P_e =3%.

Configuration2: K=4, R_0= 1 Mbps, P_e =5%

In this configuration, 4 RTP senders share the channel with an initial rate each of 1 Mbps, the fair share rate is 0.9 Mbps.

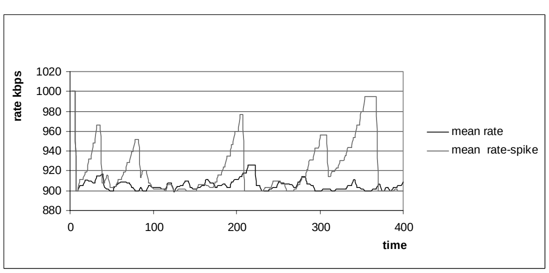

Figure 13. Configuration 2: 4 flows, Pe =5%.

Figure 13 illustrates the mean RTP rate at the receivers. The total rate perceives an upgrade around 7% when applying the Spike-WPRDR scheme.

Configuration3: K=20, R_0= 190 kbps, P_e =0.5%

Here, the number of RTP senders is set to 20. Figure 14 depicts the mean RTP rate at the receivers. Applying Spike-

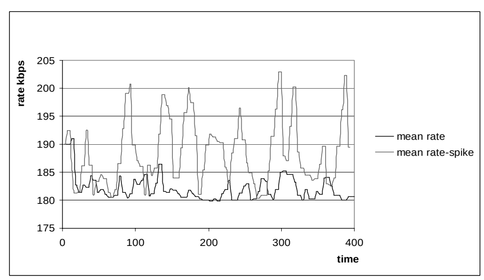

Figure 14.Configuration 3: 20 flows, Pe =0.5%.

W-PRDR, the transmission rate can be improved up to 195 kbps, considering that with only PRDR, the transmission rate is close to the fair share rate of 180 kbps.

6. Summary and future work

The simulation results under different network topologies show a clear improvement in the performance of the W-PRDR congestion control algorithm with use of a loss discrimination scheme. In a wireless topology, the use of Spike-train differentiation scheme ameliorates noticeably the sender transmission rate in presence of competing traffic.

Nevertheless, we have to evaluate the performances and accuracy of the W-PRDR congestion algorithm on more complete and realistic network conditions i.e. in presence of high error rate and intensive competing traffic. Moreover, more in-depth studies have to be done on wireless backbone topologies and purely ad hoc networks.

Besides, comparative evaluation is to be carried to other loss discrimination algorithms such as WLDA+ [28]. Finally since the work in [15] shows that hybrid algorithms such as Zig-Zag and ZBS have excellent results in discriminating loss causes under various topologies and competing flows. Therefore, combination of the W-PRDR with such loss differentiation mechanism is to be investigated

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