Delay Performance Analysis of a Two-Stage Cross-Layer Scheduler for Wireless LANs

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Abstract. The transport of real-time data flows over wireless channels can be optimized by cross-layer schedulers which serve the different users dependent on their traffic load and channel state. This paper investigates the delay performance of a two-stage cross-layer scheduler which has been designed to meet the above requirements. The well-known Weighted Fair Queuing Scheduler can tramsmit packets within a guaranteed delay that can be analytically determined, however the delay cannot be adjusted directly. The latter is possible with the scheduler discussed in this paper, where a transmission is performed in a stochastic way which means a certain percentage of the packet is lost, which is however often tolerable for applications such as video or VoIP. An analytical approach of the delay performance of the scheduler is given and validated by simulation results.

Keywords: Cross-Layer, Scheduling, Wireless LAN, Quality-of-Service.

1 Introduction

The transport of data streams for real-time applications over wireless networks includes challenges. The radio channel is usually time-variant so that prerequisites have to be taken in order to maintain troughput and delay constraints required by the application. A widely used scheduling mechanism for communication networks in general is the Weighted Fair Queueing Scheduler. The "original" version of that scheduler however assumes a transmission media with constant physical bit rate. Given a number of data flows belonging to different users and assuming that the sum of all traffic loads does not exceed the capacity of the line, it guarantees the delivery of each packet within a delay that is dependent on the weights assigned to the different flows. By means of the weighting factors, the scheduler considers the different traffic loads of the flows so that each of them gets a portion of the channel capacity proportional to its traffic load. This however means that there is no direct way to adjust the delay of the packets immediately; the delay can be expressed analytically, it is however a dependent variable of the weighting factors which are dependent on the throughput. The same is true for extensions of the WFQ scheduler for wireless networks, as they for example have been proposed in [1]. In the latter work, the WFQ scheduler

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is extended to cope with varying channel capacities by monitoring if flows are leading or lagging. Similar to the unmodified WFQ scheduler, the article also gives an upper limit for the delay which is again dependent on the weighting factors. However, for the WFQ scheduler and its derivatives, the user can control the throughput for each flow by specifying a weighting factor. Regarding the upper boundary of the delay, a closed expression can be given which is dependent on the weighting factors. That means that the delay is constrained, but it is dependent on the throughput settings.

In this paper, a cross-layer scheduler is analyzed which has been introduced in previous publications such as [4]. In contrast to the WFQ scheduler, the scheduler discussed here allows the immediate setting of the throughput and the delay for time-critical flows, which has been investigated by simulations. In this work, an analytical model of the scheduler is developed which allows to determine if packets can be transmitted within a given time and if this behavior is deterministic – i. e. all packets can be transmitted – or stochastic, which means that there is a certain probability that a packet cannot be scheduled within the given delay constraint and hence has to be discarded. The latter, however, is tolerable for many real-time applications. In a video transmission, a missing frame is hardly noticed by the viewer; in case of VoIP, some missing audio samples do not severely affect the quality of the phone call. The task of this paper is to give a quantitative analysis of the delay characteristics and the packet loss probability dependent on the channel load.

2 Structure of the Scheduler

Investigations on cross-layer scheduling are an important research topic in recent years. For example, the scheduling concept presented in [2,3], which is specially designed for OFDM-TDMA transmissions and integrates the channel state into the MAC layer scheduling. In the approach presented in this paper, TDMA is used as well, however the PHY scheduling is separated from the MAC scheduling; the schedulers communicate through an abstract interface, i. e. providing an importance metric instead of giving detailed information about packet lifetime etc. Two scheduling concepts are analyzed in [6], where one has a better support for QoS and the other one has a better support for the total throughput. The investigations presented here focus on the delay, which is important from the view of the user, whereas a provider aims to maximize the total throughput. Fair scheduling in wireless networks is, besides the previously mentioned work [1], widely investigated, for example in [7] or [8], whereas the latter work also differentiates between different QoS classes.

The design of the cross-layer scheduler discussed in this paper is given in fig. 1. The scheduler includes two stages: in the MAC layer stage, a queue is maintained for each data flow. In each turn of the scheduler, one or more packets are selected and handed over to the physical layer scheduler, along with a priority value which is determined according to the MAC scheduling scheme, which has knowledge about the QoS requirements throughput and delay of the data flows,

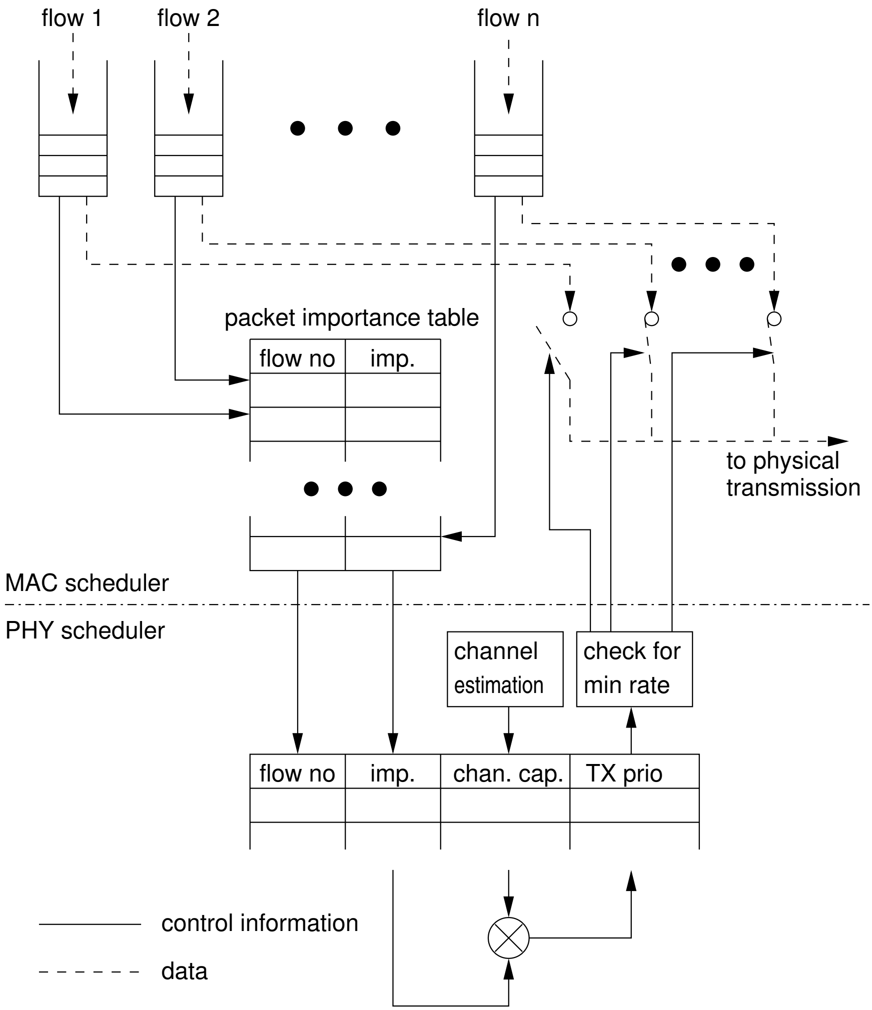

Fig. 1. Design of the cross-layer scheduler

also it keeps track about the packets which have been successfully transmitted in the past and about the packet lifetime. While the throughput is determined by a sliding window which monitors the packets transmitted within a particular time span, the delay control is maintained by assuming that the packets have a limited lifetime until they are transmitted. While the packet is waiting for transmission, the remaining lifetime is counted down; if it reaches zero, the packet is discarded. In order to calculate the final priority out of the throughput and delay measurements, the latter is converted by a weighting function; after that the results both for the throughput and the delay are added to determine the priority. After this operation has been completed for each data flow which has a packet ready to be sent, the selected packets are handed over to the physical layer part of the scheduler along with their priorities. The PHY layer scheduler supports different transmission schemes – TDMA, OFDMA, SDMA – which can be selected by the user. Dependent on the active scheme and the channel conditions, one or more packets which have been handed over by the MAC layer scheduler are selected for transmission. The MAC layer scheduler gets a feedback which packets were actually sent so that it can update the state of its queues.

For the investigations, only the downlink is considered. In this paper, the above scheduler model which has been used in previous investigations is simplified to allow an analytic description of the delay performance: the priority is calculated only based on the measurements of the remaining lifetime.

3 Analytical Modeling

Analytical considerations regarding the throughput performance of the above-mentioned scheduler have been given in [5]. In this paper, the delay characteristics of the scheduler described above are investigated. n data flows are considered, where each of them is maintained by its own queue. It is however assumed that the load generator generates a packet whenever the previous one has been successfully transmitted, so that the queue length is 1 and each station always has a packet to transmit. TDMA is used as the transmission method so that exactly

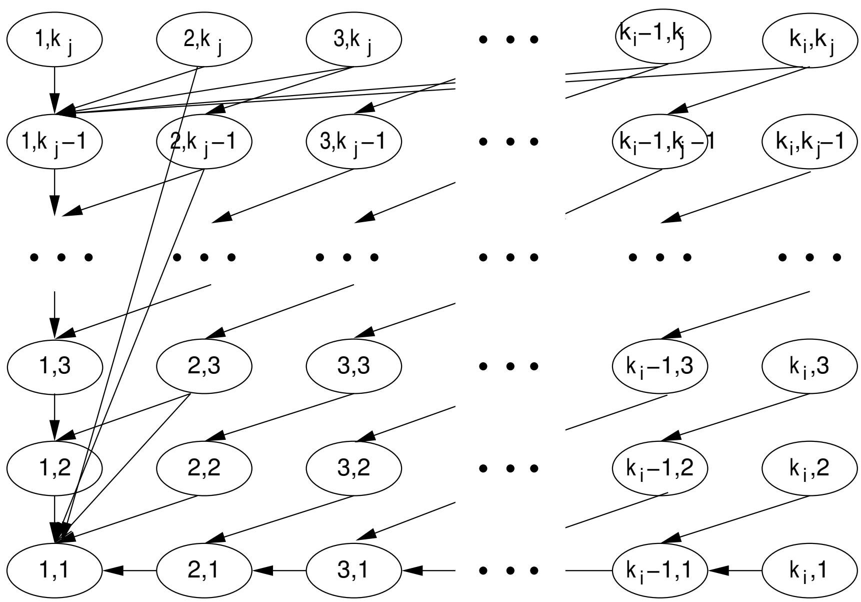

Fig. 2. Transition diagram for two data flows. The oval symbols identify the states of the system, the numbers identify the remaining lifetime for flows i and j.

one flow is served at each time. Constant packet size is assumed, the channel is time-variant in such a way that the transmission time of a packet is uniformly distributed with the mean value T, the minimum $T_{\min}$ and the maximum $T_{\max}$. It is assumed that $T_{\min}$ and $T_{\max}$ are the same for all flows. The throughput control of the above scheduler is not considered for the analytical investigations, because it affects the priority of individual packet dependent on events in the past. Since the presented approach is based on a Markov chain, it is however required that the next state of the system only depends on the current one.

Considering the fact that exactly one packet of one flow is served at a time, this means that in average the other flows have to wait for the time interval T until the scheduler serves the next packet. For simplicity, it is assumed that the maximum age of a packet is an integer multiple of the slot length T; the maximum age of a packet belonging to flow i can then be expressed as the maximum number of time slots $k_{\max,i}$ the packet may wait in the queue until it expires.

With these prerequisites, the scheduling policy can be modeled by an n-dimensional Markov chain, where each dimension represents the number of time slots that the packet of a particular flow has left until it expires. Each state in the chain is identified by an n-tuple giving the number of time slots k_i which are remaining for the next packet of flow i until expiration, i. e. $(k_1, k_2, \ldots, k_n)$. If flow i is served and thus the next packet is provided by the load generator, then k_i is set to the lifetime of the new packet $k_{i,\max}$, whereas for the other flows $j \neq i$ the respective k_j are decremented by 1. If the packet of flow i expires before transmission, the next packet is provided and k_i is reset as in the case of a successful transmission. Fig. 2 shows the state diagram for two flows i, j. In a system of n flows, the diagram shows two dimensions out of an n-dimensional cube. For reasons of overview, the tuples which identify the states only give the indices for the two dimensions i, j shown in the diagram.

If none of these two flows is served, the lifetime for both packets inside flow i and j is reduced by 1, so that the system moves from (k_i, k_j) to $(k_i - 1, k_j - 1)$, i.e. along the arrows running in a diagonal, pointing to the lower left. If flow i is served or the lifetime of flow i has expired, the system moves from (k_i, k_j) to $(k_{i,\max}, k_j - 1)$ which results in an arrow pointing horizontally to the rightmost column of the diagram. Likewise, a service of flow j means moving to $k_i, k_{j,\max}$ denoted by an arrow pointing vertically to the row at the bottom of the figure.

In order to determine the QoS characteristics of the scheduler, the probability must be known that a flow is served before its lifetime expires. It is assumed that each flow has a uniform distribution of the transmission time with the same $T_{\min}$ and $T_{\max}$ as specified above. In order to determine the transmission priority β_i of flow i, the transmission time T_i which the flow experiences at the current moment due to the channel conditions is divided by a weighting factor $w(k_i)$ which is dependent on the remaining lifetime of the packet k_i:

$$\beta_i = T_i / w(k_i). \tag{1}$$

The flow with the smallest priority value $\min_i \beta_i$ is served next. $w(k_i)$ must be a strictly monotonic increasing function; in this paper, the proportionality $w(k) = k$ is used.

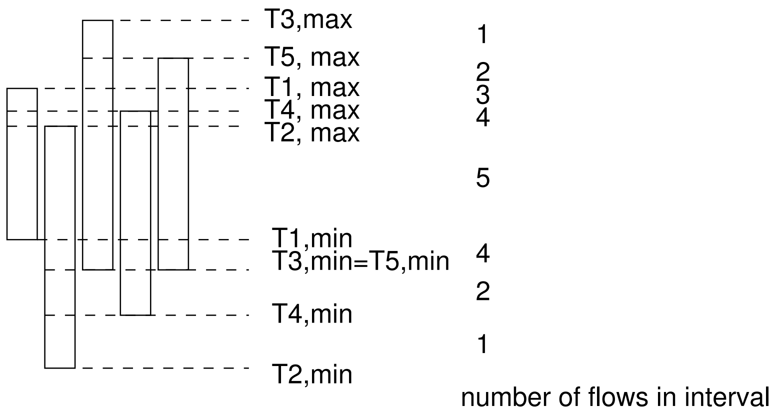

Fig. 3. Calculating the transition probability

The priority β is dependent on the randomly distributed transmission time T, hence the priority itself is a random variable. In order to determine the transition probability from one state to another one, for each of the n flows the probability that the respective flow is served in the next turn must be determined. For n flows with uniformly distributed transmission times and the same $T_{\min}$ and $T_{\max}$, where in each scheduling turn the flow with the smallest transmission time T is selected, the service probability for any of the flows is $1/n$ because of symmetry reasons. As already mentioned, the transmission times are mapped to priorities by weighting factors which are dependent on the packet age of the respective flow. If flow i has a remaining lifetime of k_i, its priority values range between $\beta_{i,\min} = T_{\min}/k_i$ and $\beta_{i,\max} = T_{\max}/k_i$ dependent on the channel condition. The different flows usually experience different weighting factors so that the resulting intervals $[\beta_{i,\min}, \beta_{i,\max}]$ might or might not be overlapping. In order to calculate the service probability for a particular flow, the $\beta_{\min}$ and $\beta_{\max}$ of all n flows are sorted in a common list in ascending order: $\beta'_1 \leq \beta'_2 \leq \ldots \leq \beta_{2n}$. Out of this list, intervals $[\beta'_1, \beta'_2]$, $[\beta'_2, \beta'_3]$, $\ldots$, $[\beta'_{2n-1}, \beta'_{2n}]$ are formed, which is illustrated in fig. 3. For each interval mentioned above, there is a certain probability q_i that a random sample β_i of a particular flow i is inside the interval. q_i is greater than zero if the following condition is met:

$$\beta_{\min,i} \leq \beta'_j \quad \text{and} \quad \beta'_{j+1} \leq \beta_{\max,i} \tag{2}$$

and can be calculated in this case as

$$q_i = \frac{\beta'_{j+1} - \beta'_j}{\beta_{\max,i} - \beta_{\min,i}}. \tag{3}$$

In a particular scheduling turn, for each flow i a random sample of β_i is drawn. Due to the probabilities q_i, in case of n flows, there are 2^n combinations for which flows the current random sample is inside the interval $[\beta'_j, \beta'_{j+1}]$. The flow indices i are mapped to values i' in a way that those l flows with random samples inside the interval are changed to $1 \leq i' \leq l$ and the other $n-l$ flows to $l+1 \leq i' \leq n$. Considering that for a given combination of l flows whose β is distributed inside the same interval for each flow, the probability that a particular flow is served is $1/l$ as mentioned earlier.

The probability has to be further conditioned due to the fact that only combinations within intervals which fulfil the inequality

$$\beta'_{j+1} \leq \min_i \beta_{\max,i} \tag{4}$$

contribute to the service of a flow. Expressed in words this means that combinations located in intervals not covered by above condition do not provide any service. For those latter intervals, there is always at least one flow which never occurs inside the interval because it has shorter transmission times and thus a smaller β. The probability that service is provided to flow l with $\beta_l \in [\beta'_j, \beta'_{j+1}]$ is then

$$p([\beta'_j, \beta'_{j+1}], l) = \frac{1}{l} \prod_{\alpha=1}^{l} q_\alpha \cdot \prod_{\alpha=l+1}^{n} (1 - q_\alpha) \quad \text{if eqn. (4) is met, 0 else.} \tag{5}$$

To determine the transit probability for one particular flow, all $p([\beta'_j, \beta'_{j+1}], l)$ for this flow have to be summed up for all possible combinations.

If the current system state is $(k_1, \ldots, k_i, \ldots, k_n)$ and flow i is served, the next state is then $(k_1, \ldots, k_{i,\max}, \ldots, k_n)$. When defining $M = \{\text{flow1}, \ldots, \text{flow}l\}$, the transition probability $p_{\text{tr},i}$ for this flow is then:

$$p_{\text{tr},i} = \sum_{\text{all } [\beta'_j, \beta'_{j+1}], M} \; \sum_{\text{all combinations}} p([\beta'_j, \beta'_{j+1}], l). \tag{6}$$

After calculating the transition probabilities, the stationary probabilities of the states have to be determined. In order to do so, first the Markov chain with m dimensions with the lengths $k_{1,\max}, k_{2,\max}, \ldots, k_{m,\max}$ is mapped to a one-dimensional one with the length $A = \prod_{i=1}^{m} k_i$ by transforming the tuple $(k_1, \ldots k_m)$ of a state to a scalar index with the range $1 \ldots$ A. Out of this chain, a squared transition matrix with the size $A \times A$ can be written which has the form

$$\begin{pmatrix} p(1|1) & \ldots & p(1|A) \\ \ldots & \ldots & \ldots \\ p(A|1) & \ldots & p(A|A) \end{pmatrix}. \tag{7}$$

This matrix must fulfil the condition that the sum of the matrix coefficients in each row must be one. With the further condition that the sum of all stationary probabilities must be 1, a linear equation system can be formed which can be solved by numerical means.

An important QoS metric is the loss probability due to expiry of the packet lifetime. With the known stationary probabilities, the loss probability is the probability of transiting from state 1 back to state n, which is $p(1) \cdot p(n|1)$.

By means of the Markov chain, it can be determined what are the requirements that all packets can be transported and what is the delay in this case, as well as it can be determined under which conditions packet loss occurs and what is the amount of this packet loss. The probability that a packet for flow i is transmitted after exactly r trials can be determined by summing up the stationary probability for all states which include $k_i = k_{i,\max} - r$. In this way, the discrete distribution function for each flow can be easily calculated.

The probability for a packet loss of flow i is the sum of the transition probabilities for all states with $(k_1, \ldots, k_i = 1, \ldots, k_n)$ to the respective corresponding state $(k_1, \ldots, k_i = k_{i,\max}, \ldots, k_n)$.

4 Comparison between Analytical and Simulated Results

In the simulation setup, the CCDF for the waiting time which each of a number of flows experiences is determined. 4 flows are considered, where one has a delay

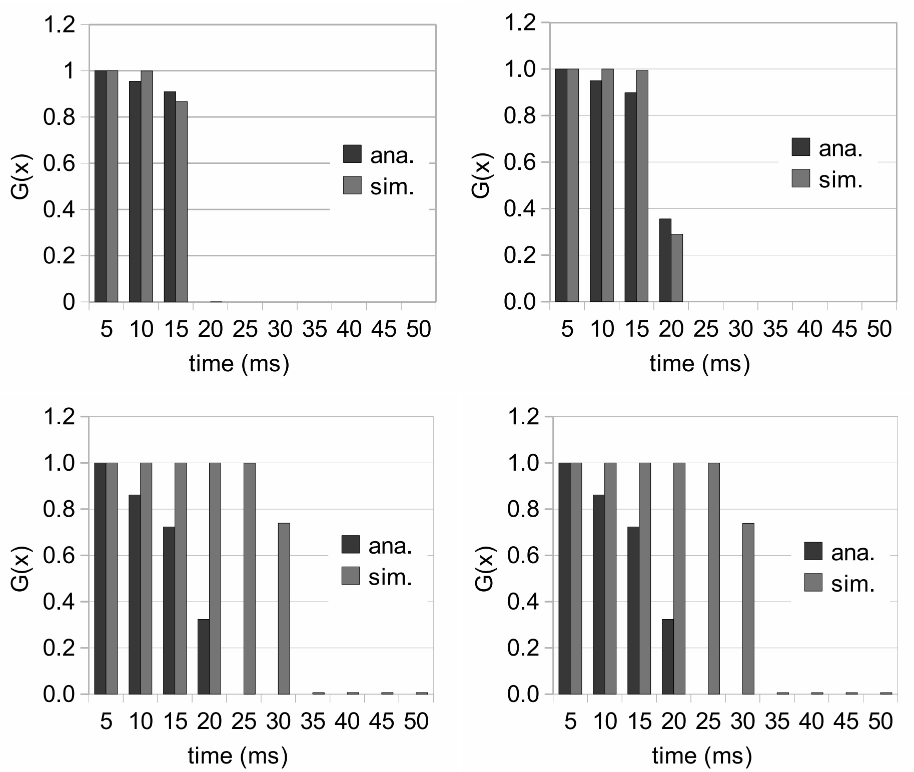

Fig. 4. CCDF of the waiting time for the four flows

constraint of 20 ms, one has 30 ms and the other two have 40 ms. The transmission time varies between 4.5 and 6.5 ms which means with a mean value of 5 ms. Fig. 4 shows the distribution function for the waiting time for each of the four flows. Precisely, the value which is counted to generate the figure is number of scheduling cycles which elapse until the next packet for a particular flow has been served. Assuming the mean value of 5 ms for a packet transmission, the number of scheduling cycles can however be interpreted as the transmission time. The simulation results for the delay and the packet loss rate are compared with the results from the theoretical analysis.

It can be seen that the scheduling scheme keeps the constraints for the respective delays between 20 and 40 ms. Only a small amount of packets exceeds the limit and has to be discarded.

For the flows with a limit of 20 or 30 ms, the analytical results are well matched by the simulation. For the flows with 40 ms delay, the simulations yields higher delays than the analytical consideration. The reason is that the assumption of constant transmission time for each packet which was needed for the theoretical analysis is an approximation. In the theoretical approach, the priorities which are calculated for the packets hence are discrete values. In the stochastic simulation, the priority values are continuous.

5 Conclusions

The scheduler discussed in this paper differs from the well-known WFQ scheduler in two ways: on the one hand, it is possible to specify a given maximum delay for the packets being delivered over the radio network. On the other hand, it cannot be guaranteed that all packets can be delivered within a given time, hence the scheduler behaves in a probabilistic way. In contrast to this, the WFQ scheduler is deterministic in the sense that it can deliver all packets within an upper boundary for the delay that can be analytically deduced, however it is not possible to specify a maximum delay that the scheduler has to meet; the delay is a result of the weighting factors which determine the throughput which the scheduler should allocate for each data flow, i.e. it does not allow to configure the delay independently from the throughput. For applications requiring bitwise precise transmissions, this means the WFQ is more suitable due to its deterministic behavior of transporting all packets. However, for real-time applications, the proposed scheduler can be more useful, since it can meet delay boundaries while the application in many cases tolerates packet loss by a certain amount. The probability of a packet loss is dependent on the total channel load, it increases the more data has to be transported.

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