Optimization of a QoS Aware Cross-Layer Scheduler by Packet Aggregation

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Abstract In this paper, the performance of a QoS-aware two-stage cross-layer scheduler utilising a MIMO channel for transmission is considered along with TDMA, OFDMA and SDMA channel access methods which serves a number of users with QoS-constrained data flows. OFDMA and SDMA allow a parallel transmission of packets which can have different transmission durations due to varying physical bit rates and packet lengths. The data flow with the longest packet slows down the other flows because they have to wait until the transmission is complete. This paper proposes packet aggregation where waiting times are reduced by transmitting more than one packet per user if airtime is left. It is shown that this method significantly enhances the QoS parameters throughput and delay. For constant-size packets, shorter delays can be achieved than for variable-size packets. Aggregating non-consecutive packets further enhances the performance, however the packets have to be buffered at the receiver to put them into the correct order.

1 Introduction

Wireless LANs have to meet increasing requirements nowadays and in the future: high data rates for each user, high spectral efficiency in the sense of a high total capacity and meeting several types of QoS requirements for different applications.

Up to now, most protocols stacks are designed according to the OSI model which defines seven layers from the physical layer up to the application layer, with an increasing degree of abstraction from the physical hardware. In legacy protocol stacks, these different protocol layers have been optimised independently of each other. This separation is in particular problematic for the design of the two lowest layers, which are the MAC and the PHY layer, because there are close mutual dependencies between these two layers. The QoS requirements have already to be considered by selecting the physical transmission method. Moreover, the actual channel conditions and the effects of these conditions for a QoS aware transmission have to be known when selecting a particular packet for the transmission.

To cope with these requirements, in the framework of the xLAYER project funded by the German Research Foundation (DFG), a cross-layer transmission system for wireless LANs was introduced in [9] and extended by QoS support in [10]; the OFDMA/SDMA platform which means the parallel transmission to multiple users was introduced in [5].

The cross-layer transmission system is located in the access point which has full control of the channel access, similar to the Hybrid Coordination Function Controlled Channel Access (HCCA) specified in IEEE 802.11e. In this paper, only the downlink from the access point to the mobile stations is considered.

In the previous investigations mentioned above, a number of boundary conditions were simplified. The number of users was assumed to be constant and the load of each user as well as the packet size was assumed to be time-independent. In practice, such idealised conditions do not apply. Typical applications such as voice-over-IP (VoIP) have variable packet size and also can have varying load dependent on the used codec. This paper investigates the properties of the previously introduced QoS scheduler considering that the above-mentioned parameters can be variable. In addition, the efficiency of the scheduler is increased: When packets for different data flows are transmitted simultaneously using OFDMA or SDMA, the packet for one flow might be transmitted faster than for another flow due to different packet lengths and channel conditions. In the previous investigations, the faster flow then had to wait for the slower flow because transmission for all flows starts at the same time. The scheduler is now enhanced in the way that the faster flow can transmit one or more additional packets in the remaining time until the slower flow has finished its transmission.

Methods of legacy scheduling schemes which do not consider application requirements are discussed in [14, 8]. Schedulers which are specialized on video applications are discussed in [7, 13, 6, 2]. The metric to optimize the transmission is the quality of the video image at the mobile station. The optimum transmission is achieved by selecting the most suitable type of video codec as well as adjusting parameters in the MAC and the PHY layer. The approach given in this paper is independent on the particular application, however QoS requirements specified by the application are considered.

In [16], it is pointed out that Video-On-Demand can cope with relatively large delays which should however be constant, i. e. a low jitter is required. From the view of the QoS scheduler discussed in this paper, the jitter is kept small by enforcing a short delay. Low jitter means that a packet always has to be sent inside a short time frame, regardless if this has to be done immediately or after a certain delay. Two scheduling concepts are analyzed in [1], where one has a better support for QoS and the other one has a better support for the total throughput. In this paper, the aim of the scheduler is to satisfy the QoS requirements for a maximum number of users, because meeting the QoS requirements is the criteria which results in the highest satisfaction for the user. A user does not get personal benefit if a system optimizes the total throughput, but the performance of the own application is poor. The scheduling concept presented in [4, 3] is specially designed for OFDM-TDMA transmissions and integrates the channel state into the MAC layer scheduling. In the

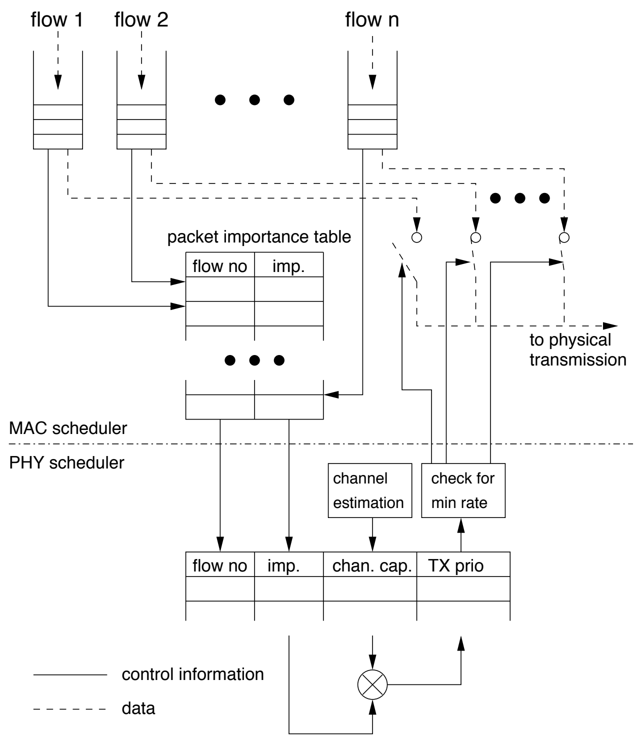

Fig. 1 Design of the parallelised cross-layer scheduler

approach presented in this paper, the PHY scheduling is separated from the MAC scheduling, however the schedulers communicate through an abstract interface, i. e. providing an importance metric instead of giving detailed information about packet lifetime etc.

The method of aggregating packets is for example used in IEEE 802.11n to avoid loss of airtime due to contention periods and acknowledgements [15]. The aggregated packets are treated as one large packet.

2 Scheduling for Varying Load Conditions

Fig. 1 shows the design of the cross-layer scheduler introduced in [9] which is used for the investigations described in this paper. The scheduler includes a hardware-independent stage in the MAC layer and a hardware-dependent stage in the PHY layer. In the hardware-independent stage, each data flow of a user is assigned its own queue. In each turn of the scheduling process, the MAC scheduler assigns a priority to the data packet at the top of each queue according to a certain scheduling scheme. Different scheduling schemes were compared in [10], where also a scheduling method with quality-of-service support was proposed and investigated. In this scheduling method, which is also used for the investigations in this paper, the throughput is monitored by observing the number of successfully transmitted packets inside a sliding window and compared against the target value. The maximum delay of the packet is mapped to its remaining lifetime, which is determined by subtracting the time which the packet already waits in the queue from this maximum delay. The difference is mapped to a priority value using a weighting function. The throughput-based and the delay-based contribution to the priority are then added for each user. The results are kept in a table for each user; this table is handed over to the PHY scheduler which determines the available channel capacity for each user which is then multiplies each entry of the list with the priority for the respective user. For the physical transmission, TDMA, OFDMA and SDMA were considered in [11]. In that work, the packet size of the user data flows is assumed to be the same and constant for all users. The operation of the scheduler becomes however more complex if variable packet size is assumed.

If the packet size is constant, it is possible to consider the queue length, i. e. the number of packets in the queue as a criterion for scheduling which was done in the previous papers for comparison. In case of variable packet size, it is needed to consider the amount of data in bytes which is kept in the queue. In case of the QoS enabled scheduler, up to now, the number of packets transmitted in the past were compared against a reference to control the throughput. In case of variable packet sizes, the sizes of packets transmitted within a time interval need to be added and compared against a reference throughput.

In case of a parallelised transmission, it was up to now assumed that exactly one packet is transmitted for each user in each turn of the scheduling process. This is not optimum even if the packet size is the same for all flows, because the users can face different channel capacities so that a user with a fast channel has to wait for a user with a slower channel until his packet was transmitted. The problem increases if the packet sizes of the users are different. For this reason, the scheduler needs to be extended: if the packet for data flow A is transmitted within a shorter time than the packet for data flow B, then the remaining time can be used to transmit other packets for A until the transmission for B is finished.

The above-mentioned task is performed by the algorithm given below, where $S_{i,1}$ is the size of the packet at the top of the queue of user i and and C_i is the channel capacity of that user.

Select the flows for which a packet should be transmitted
according to the legacy scheme:
for all flows i **do**
  calculate the transmission time $T_{i,1} = S_{i,1}C_i$
end for
find flow j which needs maximum time for transmission:
$j = \arg\max_i T_{i,1}$.
for all users $i \neq j$ **do**

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    calculate the remaining transmission time:
    T_rem,i = T_j − T_i,1
    n = packet no in queue i
    for k = 2 to n do
      T_i,k = S_i,k · C_i
      if T_i,k < T_rem,i then
        append packet i,k to packet i,1
        T_rem,i = T_rem,i − T_i,k
      end if
    end for
  end for
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By this method, more than one packet can be sent for a user within one scheduling process. However, the packets might be taken from the queue non-consecutively. Assuming the packet at the top of the queue (no. 1) was taken and the next packet (no. 2) is too big, the search continues until the end of the queue is reached. If no. 3 is also too big, but 4 and 5 fit into the gap, then packets 1, 4 and 5 are transmitted. Since the order of the packets must not be changed when being handed over to the upper protocol layers, only packet 1 may be handed over; 4 and 5 have to be kept inside the buffer until the missing packets 2 and 3 were received.

The case that the data flow uses constant-size packets is easier. In this case, consecutive packets can be taken from the queue until the gap is filled.

3 Simulation Setup

The channel capacities are determined based on the IEEE 802.11 TGn radio channel model proposed in [5] which is deployed here to implement a MIMO transmission with $M = 2$ transmit antennas at the base station and $N = 2$ receive antennas at each of the mobile stations, $K = 8$ users and $L = 52$ subcarriers. The model considers a typical indoor environment where the signal transmitted by the sender is reflected at a number of objects so that a large amount of signal components arrive at the receiver. Based on this model, the channel matrices are calculated for each user. Due to the OFDM based transmission, this matrix calculation has to be done for each subcarrier.

For TDMA, OFDMA and SDMA, the throughput and delay is compared between data flows with variable and constant packet size. In case of variable packet size, the size is uniformly distributed between 500 and 1500 bytes.

For OFDMA and SDMA, the effect of packet aggregation is compared against the case of transmitting individual packets for each user.

The simulator used for the investigations discussed in this paper is called WARP2; it implements the IEEE 802.11 protocol stack and has been extended with the two-stage MAC/PHY scheduler as described above. The simulated scenario includes an access point which serves a number of stations. The stations move between 5 m and 15 m distance towards and away from the access point with a speed of 2 m/s. They start at a randomly selected point and change the direction when one of the boundaries was reached. Each station is assigned one user with a certain traffic category. The load generator generates CBR traffic for the users 1, 2 and 4 and Poisson traffic for the other users; the traffic load is configured individually for each user. The packets for each user are stored in the respective queue until they are served by the MAC scheduler. The MAC scheduler works with the QoS enabled scheduler mentioned before.

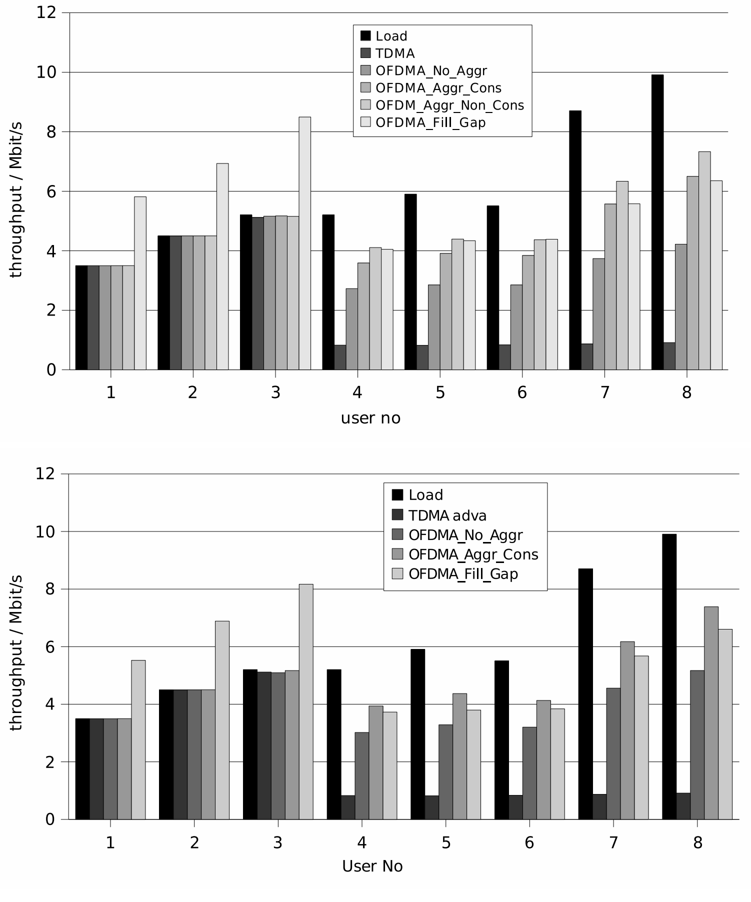

Fig. 2 Per-flow throughput for OFDMA. Up: variable, down: constant packet size

4 Results and Discussion

Fig. 2 shows the throughput for the eight routes for different OFDMA access methods and variable resp. constant packet size. For each user, the bar on the left depicts the offered traffic load. The throughput achieved by TDMA is shown right to the load bar for comparison. Four different OFDMA scenarios are considered: transmitting a single packet without aggregation (OFDMA_No_Aggr), aggregation of consecutive packets (OFDMA_Aggr_Cons) and, in case of variable packet size, aggregation of non-consecutive packets (OFDM_Aggr_Non_Cons). Finally, for comparison, the theoretical case is given that the gap is completely filled with a packet that exactly has the size of the gap. For the idealised case where the distribution function of the packet size and of the channel capacity is uniform or exponential, it was shown in [12] that the simulation results can be well described by analytical means. The results in Fig. 2 show that all OFDMA techniques yield better performance than TDMA by a factor of about 2.5 due to the parallel transmission for different users. The TDMA throughput is almost the same for all non-time-critical flows. Packet aggregation increases the throughput by another 20% due to the reduced gap size. If the packets are aggregated non-consecutively, another enhancement can be observed because the chance is higher that a packet can be found in the queue which still fits into the gap. In the theoretical case of filling the gap completely, the throughput is lower than for variable packet size for users 7 and 8. On the other hand, for the time-critical users 1 to 3, the throughput is higher than the offered load, which is due to the assumption taken in this case that the entire gap is filled with additional data not contained in the queue.

Fig. 3 shows the delay for the same scenario. Users 1 to 3 are served according to their QoS requirements. The delay for the other users inversely corresponds to the throughput, i. e. the higher the throughput for a user is, the lower is the delay. This fact is explained by the lower queueing delay which a packet experiences if the queue service rate is higher. The delay is lower for constant packet size, because for eight user data flows, it is likely that there is at least one of them which has to transmit a long packet so that the scheduling interval will be long.

In Fig. 4, the per-user throughput is shown for SDMA in case of variable and constant packet size, respectively. The different cases are the same as for OFDMA, i. e. transmission without aggregation, with consecutive and non-consecutive aggregation and filling the entire gap. The results are similar than for OFDMA, which is that packet aggregation enhances the throughput and variable-size packets result in a better throughput than constant-size packets. Also, for idealised filling the gap, the throughput is in case of users 7 and 8 lower than for packet aggregation. For the time-critical users 1 to 3, the QoS criteria are always met. For the other users, the achieved throughput is higher than in case of OFDMA. The throughput for variable packet size is about the same than the throughput for constant packet size if the variable-size packets are aggregated non-consecutively. In case of consecutive packet aggregation, the throughput for variable-size packets is lower than for constant-size packets. This is explained by the fact that in case of consecutive variable-size packets, it is less likely to fill the gap effectively than for constant-size

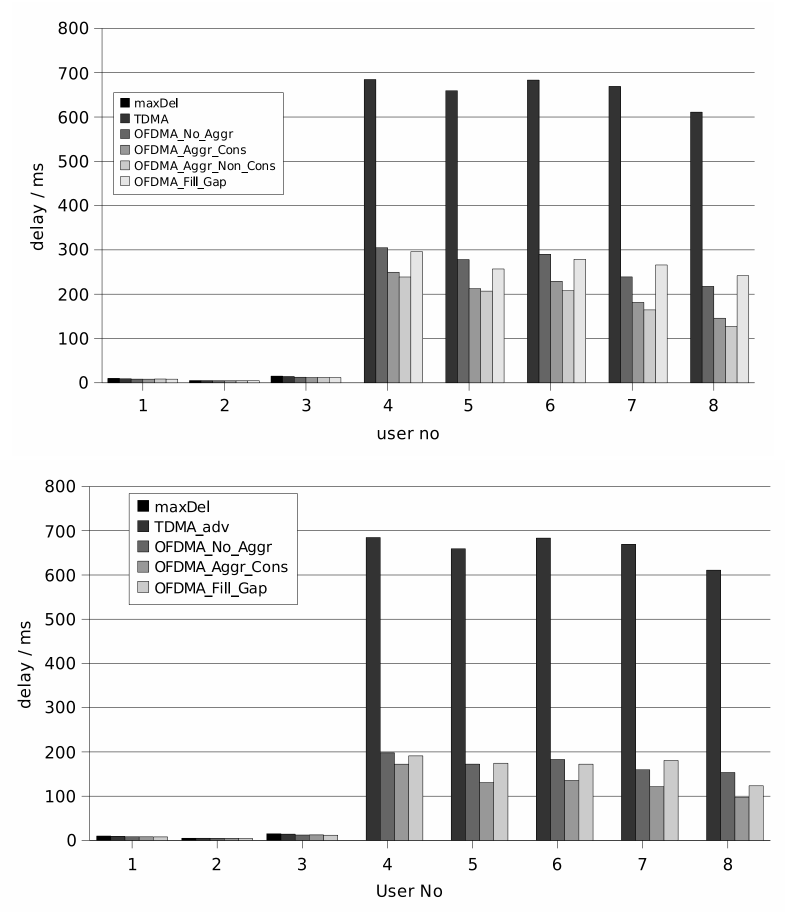

Fig. 3 Delay for OFDMA, up: variable, down: constant packet size

or non-consecutive variable-size packets. The delay is reduced for a particular user when the throughput is increased as demonstrated in Fig. 6 in a similar way as observed for OFDMA. The delay for constant packet size is always smaller than for variable packet size.

Fig. 5 depicts the behaviour of the total throughput in the different transission configurations which is similar to the per-user throughput for the non-time-critical flows.

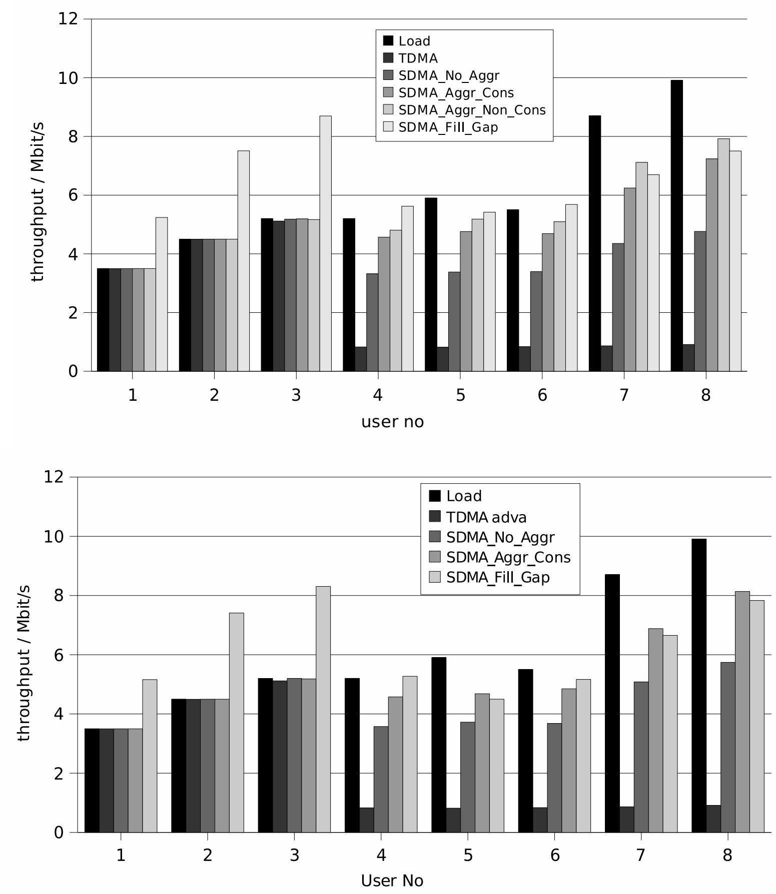

Fig. 4 Per-flow throughput for SDMA. Up: variable, down: constant packet size

5 Conclusion and Outlook

In this paper, the effect of packet aggregation on the throughput and delay performance in a wireless LAN with parallel transmission between the access point and the mobile stations was investigated by simulations. Due to variable packet length and channel capacity, airtime remains unused for a legacy transmission because a user with a good channel has to wait until the transmission of slower users is complete. This remaining airtime can be used by aggregating more than one packet from the users' queues in order to improve the Quality-of-Service parameters. In case of consecutive packet aggregation, the amount of enhancement is higher for traffic with constant packet size than for traffic with variable packet size. If non-consecutive aggregation is used, the throughput for variable and constant packet size

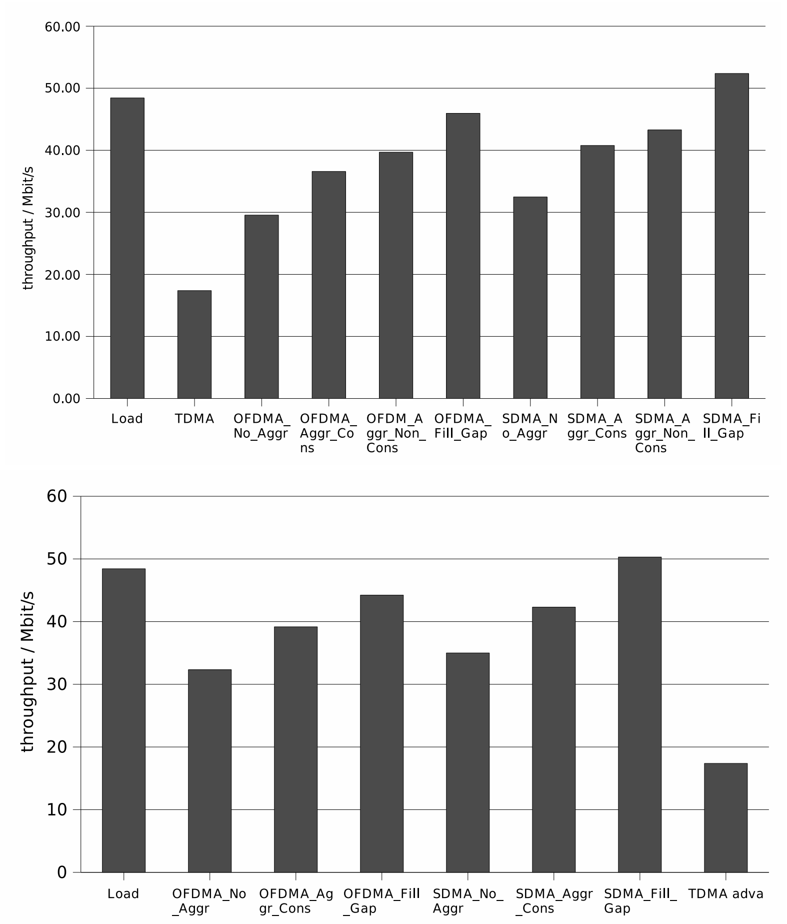

Fig. 5 Total throughput for different access methods, up: variable, down: constant packet size

is approximately the same. Along with the throughput enhancement, the queueing delay is also reduced due to the higher service rate of the queues. The packet aggregation enhances non-time-critical data flows for which the scheduler does not enforce guaranteed QoS parameters. Users with QoS-constrained transmission are in any case correctly served due to the properties of the scheduler.

Further investigations will consider the behaviour of the system in case of imperfect channel knowledge. In this case, packets can be dropped because the channel was overestimated which requires ARQ to recover the lost data. If the channel is underestimated, the physical bit rate selected for the packet transmission is unnecessarily low which reduces the achievable throughput. Thus the robustness of the scheduler against these imperfections needs to be investigated.

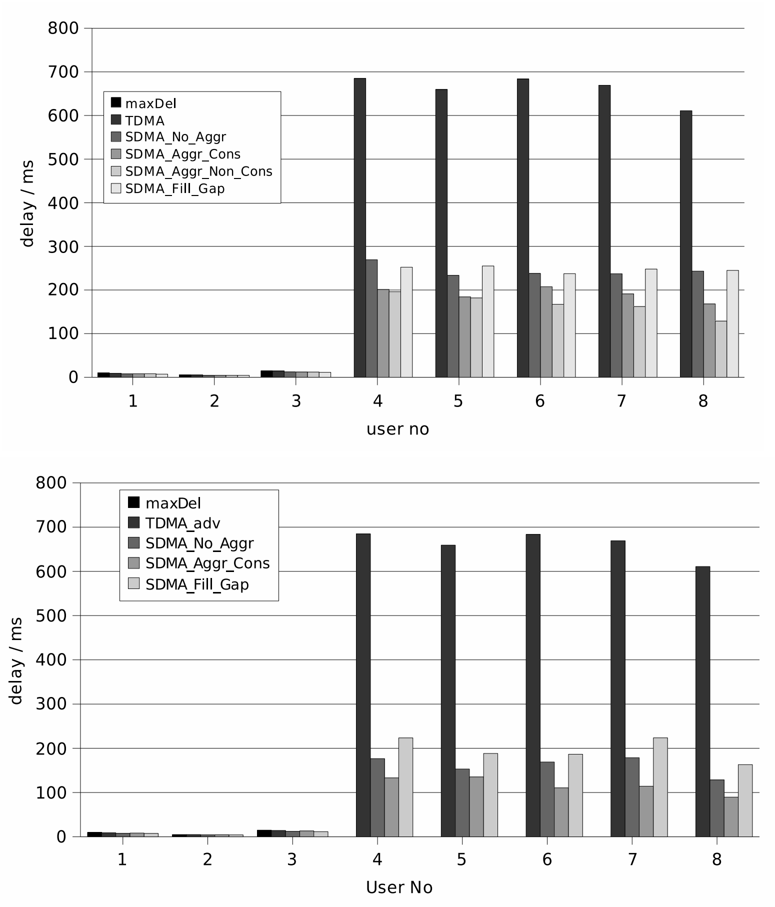

Fig. 6 Delay for SDMA. Up: variable, down: constant packet size

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