Joint Packet Scheduling and Channel Allocation for Wireless Communications

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***Abstract*— This article proposes a joint packet scheduling and channel allocation scheme for multiuser multichannel (MM) wireless networks. A fair scheduling algorithm is adapted to multichannel wireless networks by incorporating more realistic wireless channel characteristics. Channel conditions and different Quality of Service (QoS) requirements are properly reflected in the scheduling process, and allocation of multiple channels is also implemented. From both analysis and simulation, probabilistic fairness is guaranteed and utilization of each wireless channel is improved.** [1]

I. INTRODUCTION

In the protocol stack viewpoint, a wireless network consists of multiple layers, ranging from physical layer (PHY) to application layer. Among all layers, medium access control (MAC) layer is responsible for packet scheduling, medium sharing, MAC organization, power and admission control. Scheduling function selects packets to transmit when a set of traffic flows access channel. In wireline networks, GPS scheduling is a popular paradigm providing delay-bounded channel access and fairness among packet flows. A GPS scheduler is an idealized fluid flow model that services all sessions (flows) simultaneously. Consider a set of flows in a system. Flow i has a rate weight w_i over any window of time period $[t_1, t_2]$. Thus packet scheduling algorithms are to minimize the normalized throughput difference between two flows i_1 and i_2, $|\frac{R_{i_1}(t_1,t_2)}{w_{i_1}} - \frac{R_{i_2}(t_1,t_2)}{w_{i_2}}|$. Unfortunately, this conventional GPS is unrealizable in practice because it services a small part of a packet at a time.

Packet-by-packet GPS, commonly known as Weighted Fair Queuing (WFQ), assigns each packet with a start tag and a finish tag corresponding to the “virtual time” at which the first and the last bits of a flow are transmitted respectively [1]. “Virtual time” $V(t)$ is defined as

$$\frac{dV(t)}{dt} = \frac{R(t)}{\sum_i w_i},$$

where $R(t)$ is the data rate of the system at time t. This time-stamped algorithm sorts all the packets in the queue based on their increasing time tags. Hence each time a packet arrives, its time tag is calculated.

In addition to the above time-stamped scheduling technique, another solution for packet scheduling is credit-based fair queueing [2], where each flow is associated with a simple counter to store the transmission credits earned by the corresponding traffic flow. Upon service completion, scheduling is performed based on the values of the counters, the guaranteed rates, the channel condition of different traffic flows. Hence scheduling priorities are controlled by cost functions that represent how much service each flow deserves.

To adapt those existing wireline algorithms to wireless networks, features of wireless links should be taken into consideration. Recently, a number of algorithms have been proposed to adapt fair queuing to the wireless domain [3]. In those existing scheduling algorithms for wireless networks, fairness among flows can be guaranteed. However, channel throughput is not taken into consideration. Due to wireless channel fading, channel side information from PHY layer is helpful for MAC layer to improve channel utilization and maximize total throughput by avoiding poor channel conditions, whose achieved gain is commonly termed as the multi-user diversity gain [4]. By the way of how to capture channel statistics either roughly or more precisely, the gain may appear significantly different. Usually a wireless channel is modeled as a two-state Markov chain where the channel is either good or bad [3]. Then the MAC layer scheduler incorporates binary channel states. Actually, PHY layer is able to provide more accurate information on channel status such as the highest affordable data rate.

In this article, we propose a centralized scheduling MAC protocol for a multiuser multichannel wireless network. We generalize the credit-based wireline fair scheduling technique to a wireless scenario where multiple parallel subchannels are available each time. A Finite-State Markov Chain (FSMC) model for fading channels proposed in [5] is adopted to describe each channel. Credit is assigned to each flow and guarantee fairness of scheduling. Throughput is also taken into consideration by showing favor to good channels. Since multiple channels are allowed to transmit simultaneously, not only which packet but also which channel should be decided appropriately. Therefore, packet scheduling is conducted along with channel allocation in order to meet a tradeoff between fairness and throughput.

The rest of this paper is organized as follows. Section II briefly describes PHY layer features in this study. Section III describes the proposed scheduling algorithm. Some analytical results are provided in Section IV. In Section V, simulation

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results are presented and compared with analytical ones. Performance of the algorithm under different situations and effects of parameters are studied. Finally conclusions are drawn.

II. PHY Layer Features

In this section,Adaptive Modulation and Coding (AMC) schemes and wireless channel models are introduced as PHY layer features of the wireless communication system of study.

A. Adaptive Modulation and Coding

Wireless channels vary over time due to fading and interference conditions. AMC exploits these variations to maximize the data rate that can be transmitted over such channels [6]. The power of transmitted signal is held constant over a frame interval. But the modulation and coding formats are adjusted to match the current channel condition that is usually represented by received signal to noise ratio (SNR).

In this article, we adopt an AMC scheme under an assumption that all channels are perfectly known at the center node. The entire SNR range for each channel is partitioned into S non-overlapping consecutive intervals whose boundary points are γ_s for $s = 0, 1, \cdots, S$. An SNR threshold γ_1 is set to avoid deep channel fades, i.e., when SNR is smaller than γ_1, no data would be transmitted. For the mapping from SNRs to modulation/coding schemes, we adopt the transmission modes in TM1 which is recognized as part of high performance radio local area network (HIPERLAN/2) standard, a wireless LAN standard developed by the Broadband Radio Access Networks (BRAN) division of the European Telecommunications Standards Institute (ETSI). If the received SNR γ satisfies $\gamma_s \leq \gamma < \gamma_{s+1}$, the channel is said to be in mode s and data transmission rate is R_s [7].

B. Channel Model

In this article, a wireless channel is modeled as a FSMC as in [5], where a finite-state Markov chain can model a wireless channel with a finite number of states instead of early work of a two-state Markov channel known as the Gilbert Elliot channel [8]. By partitioning the range of the received SNR into a finite number of intervals, a FSMC model can be constructed for Rayleigh or Nakagami-m fading channels. As shown in Fig. 1, state transitions are assumed to occur only between adjacent states at specified probabilities.

Each steady-state probability π_s of the channel state s is determined by the distribution of the fading channel. Suppose the pdf of the channel is $p(\gamma)$. Then

$$\pi_s = \int_{\gamma_s}^{\gamma_{s+1}} p(\gamma) d\gamma.$$

If all π_s for $s = 0, 1, \cdots, S$ are arranged in a row vector π, a transition matrix P whose entries define all possible transition probabilities can be found by solving a matrix equation $\pi = \pi P$ [5]. An example of Nakagami-m fading channel is given in [7], where

$$P_{s,s+1} = \frac{N_{s+1} T_f}{\pi_s}, \quad P_{s,s-1} = \frac{N_s T_f}{\pi_s}$$

where $N_s = \sqrt{2\pi \frac{m\gamma_s}{\overline{\gamma}}} [\frac{f_d}{\Gamma(m)(\frac{m\gamma_s}{\overline{\gamma}})}]^{m-1} exp(-\frac{m\gamma_s}{\overline{\gamma}})$, f_d is the Doppler spread frequency and $\overline{\gamma}$ is the average SNR. In addition, m is the order of the Nakagami-m fading. Actually Rayleigh fading channel is a special case of Nakagami-m when $m = 1$. Due to the fact that the sum of the outgoing transition probabilities from one state always equals one, the probability of staying at the same state is

$$P_{s,s} = \begin{cases} 1 - P_{s,s+1} - P_{s,s-1}, & \text{if } 0 < s < S; \\ 1 - P_{0,1}, & \text{if } s = 0; \\ 1 - P_{S,S-1}, & \text{if } s = S. \end{cases}$$

In our case, the state intervals are determined by the AMC scheme we adopted. Each channel state s corresponds to one of the AMC modes consisting of a modulation scheme, coding rate and data transmission rate.

III. Algorithm Description

The proposed wireless network is composed of pico-nets, each of which is organized with a special node acting as a controller. Communication links are established between the center node and other nodes. The proposed scheduling algorithm also jointly assigns multiple simultaneously available channels to downlink traffic flows at the center node of a wireless network. Multiple parallel channels can be realized by different division techniques such as space division (by the means of multiple antennas), frequency division, code division, and so on. Even time division can be included in a generic sense where channel can be defined as a slot. For clarity of presentation, we focus on frequency division as an example. Different from any traditional single-link scheduling [9], [1], a traffic model of a multichannel system is described as follows.

- The entire bandwidth is divided into several independent channels. Each of the flows can access any channel.
- Independent flows in the network have independent task queues. Tasks arrive in unit of packet.
- Packet transmission is slotted, which means that only at the beginning of each time slot, the scheduler works and transmission is permitted.
- Once a channel is assigned to a packet, it will be occupied by that packet for one time slot.

Usually a downlink transmission requires higher data rate than uplink transmissions. Thus we assume the downlink traffic is very heavy and data rate is welcomed as high as possible. For a bandwidth constrained wireless network, one of the design objectives is to improve the system throughput attributed to the multi-user diversity gain [4], which is based on the fact that different flows have various transmission qualities through the same channel. Due to channel fading, stochastic bandwidth utilization is improved by showing favor to the flows with good channel conditions and less interest to those with poor channel conditions. However, sequential selection of the best channels must be balanced with fairness considerations. The fairness of wireline Fair Queuing Scheduling is evaluated by the normalized throughput difference between flows. In such a criterion, the rate $R_i(t_1, t_2)$ allocated to flow i during time interval $[t_1, t_2]$ is adjusted by minimizing the following measure to guarantee fairness of the network [10]:

$$\eta_{i_1,i_2}(t_1,t_2) = |\frac{R_{i_1}(t_1,t_2)}{w_{i_1}} - \frac{R_{i_2}(t_1,t_2)}{w_{i_2}}|, \quad \forall i_1, i_2.$$

In this definition, w_{i_1} represents the weight of the traffic flow in order to incorporate different data rate requirements of flows. The weight helps to differentiate flows' priorities and requirements.

However, it is not practical to ensure the above criterion is satisfied exactly, since packet switched networks grant channel access at the granularity of packets instead of bits. Additionally, for a random wireless channel, probabilistic instead of deterministic measure should be used to evaluate the fairness of a scheduler

$$Prob(\max_{i_1,i_2}\{\eta_{i_1,i_2(t_1,t_2)}\} < x)$$

where $\max_{i_1,i_2}\{\eta_{i_1,i_2(t_1,t_2)}\}$ is called *bias* and represents the maximum unfairness between all possible pair of flows. In the following analysis and simulation, bias will be used to represent fairness. The smaller the bias, the better fairness has been achieved. Notice that each flow will be assigned a time slot for transmission at a channel condition dependent rate each time. So in this sense, data rate can represent throughput performance. In this article, both throughput and fairness performance is pertinent to the data rate achieved by flows.

Our algorithm is extended from credit-based fair queueing scheduling [2], associated with PHY layer information and QoS consideration. Scheduling process is controlled by the cost functions of flows. To balance the service received among flows and thus achieve fairness, previous service completion is represented by the credit in the cost function. The cost function should also reflect the current channel status in order to efficiently utilize each channel in a multichannel system. The cost function $Cost_{i,j}$ can thus be defined as

$$Cost_{i,j}(t) = \frac{U_{i,j}(t) - K_i(t)}{w_i},$$

where $K_i(t)$ represents the credit of the i-th flow at time t, $U_{i,j}(t)$ represents j-th channel's cost to transmit packets of the i-th flow.

The scheduler works at the beginning of each time slot. We always assign Channel j_0 to Flow i_0 if the j_0-th channel of Flow i_0 has the the smallest cost function under the condition that $R_{i,j}$ is nonzero, which is determined by the AMC scheme. When SNR is below a certain threshold, AMC controller does not allow any transmission to guarantee bit error rate (BER) and save power. After selecting a channel, we update the value of credit and assign the next channel to some flow. If all the valid channels have been assigned, the scheduler will wait until the next frame time.

Two terms K_i and $U_{i,j}$ represent respectively previous service credit received by the flow and channel status. Introducing these two terms helps to find out a tradeoff between fairness (represented by credit K) and throughput (represented by U-function U). Thus resource is allocated according to flow's channel conditions. Rules to update them are described next.

Credit $K_i(t)$ is introduced for wireline networks to achieve fairness [11]. When flow i enters the system or becomes unbacklogged, its credit K_i is set to be zero. A flow accumulates credits when it is not scheduled and loses credits when it is scheduled. The number of accumulated (or decremented) credits is the normalized transmission rate of the current scheduled packet, which is determined by the channel condition. The relationship between transmission rate and SNR has been discussed in Section II. This value is updated whenever a flow has been assigned a channel as follows.

- For flow s who has been assigned channel j, its credit is updated as

$$K_s(t+1) = \max(K_{min}, K_s(t) - (N-1)R_{s,j}(t)),$$

  where N is the total number of flows in the system.
- For all other flows, their credits are updated as

$$K_i(t+1) = K_i(t) + min(\frac{R_{s,j}(t)}{w_s}, K_{max})w_i,$$

where K_{min} and K_{max} are respectively the minimum and maximum credits allowed by the system. $R_{s,j}$ represents the transmission rate of the selected channel, i.e., the number of packets transmitted within a unit of time. According to our credit update rule, the sum of all the weighted credits within the system is always kept zero, i.e. $\sum_i \frac{K_i}{w_i} = 0$. Thus $K_{min} < 0$, $K_{max} > 0$, and each credit is allowed to be positive or negative. Their choice depends on the statistic distribution of credit. The range $[K_{min}, K_{max}]$ should cover most of the credits that occur when no credit bound is set. Thus range is selected to satisfy $Prob\{K < K_{min}$ or $K > K_{max}\} < \delta$, where δ is a positive number much smaller than 1. However, credit is constrained to be non-negative in [2] and [12]. In those algorithms, the sum of credits of all flows would keep increasing unless all the credits are reset. Consequently, this increasing sum causes possible overflow of the memory. Difficulties in analysis also arise.

U-function $U_{i,j}(t)$ is defined to represent the j-th channel's current condition for the i-th flow. Generally, channels under better conditions should have higher priorities to be chosen. Hence cost function should be smaller under better channel conditions. The definition of U-function can be very flexible if it is a decreasing function of transmission rate $R_{i,j}$. In fact, it is also a function of SNR since $R_{i,j}$ is determined by SNR. For example, the transmission function could be defined as $U_{i,j} = -\lambda R_{i,j}$, $U_{i,j} = \alpha R_{i,j}^{-\beta}$ or $U_{i,j} = -\rho \log_\omega R_{i,j}$. In these definitions, parameters λ, α, β, ρ and ω are non-negative that can be used to adjust scheduler's performance. What type of transmission function should be selected is related to the QoS that a specific flow requests, which is similar to the well-known "utility function" [13]. For example, for a voice service, data rate increase does not make much difference if the rate is already high enough for such a service, thus a log type of transmission function may be suitable. On the other hand, a data service would like to have as high data rate as possible and in this case, a linear function is preferred. In the following discussions in this article, data service is provided and linear transmission function, $-\lambda R_{i,j}$ is adopted.

It is worth pointing out that credit K_i is only related to the index of flow, while the U-function $U_{i,j}$ is determined by both flow index i and channel index j.

IV. Fairness and Throughput Analysis

The proposed scheduling algorithm considers jointly credits of different flows and their channel conditions. The scheduling process can be modeled as a Markov chain whose state is described as a vector including all possible credits and channel states of all flows. For example for a two-flow two-channel network, a state vector V includes credits K_1, K_2 of two flows , and channel state numbers of each flow at each channel. Thus V is defined as $(K_1, K_2, C_{1,1}, C_{1,2}, C_{2,1}, C_{2,2})$.

Its steady state performance is of our interest, which is evaluated by fairness and throughput. Fairness is evaluated by $Prob(\max_{i_1,i_2}\{\eta_{i_1,i_2(t_1,t_2)}\} < x)$. It can be achieved from the probabilities of credit state, which are the sum of possibilities of all the states V_s whose credit set is $\underline{K}^q = (K_1, K_2, ..., K_N)$. As described in Section III, the fairness among different flows is measured by *bias* and a smaller bias shows a more fair situation.

On the other hand, average throughput $E\{T\}$ can be obtained through the transition probability matrix P, whose element p_{st} indicates transition probability from state V_s to V_t

$$E\{T\} = E_V\{E\{T|V\}\} = \sum_V Prob(V)E\{T|V\},$$

where $E\{\cdot\}$ represents the expected value of a variable, and $Prob(V)$ is the steady-state probability of state V which can be derived from P. Actually,

$$E\{T|V\} = \sum_{i,j} Prob(\text{Select } j\text{th channel of } i\text{th flow}|V_s)R_{i,j},$$

where the jth channel of ith flow is selected when the $Cost_{i,j}$ is the minimum, which can be easily derived for a given V.

Thus both evaluations of fairness and throughput performance require all transition probabilities from one state to another. The basic idea to obtain p_{st} is to find out all the possible next states that may occur under the condition of V_s. Compare V_t with all the possible states. If V_t is none of them, the transition probability from V_s to V_t is zero. Otherwise the transition probability is determined by the probabilities of the branches from V_s to V_t. Available channels are assigned one by one and each time all possible interim states are listed. Then for each interim state, the channels that have been assigned would be eliminated from the available set and the pair of channel and flow will be selected. When all channels have been assigned, we achieve all the possible states stemming from V_s.

The steady-state probability vector π of such a system is solved according to the fact that $\pi = \pi \cdot P$. Thus

$$\pi \cdot (I - P) = 0.$$

So π is the eigenvector of $(I - P)$ corresponding to the eigenvalue of zero. Then as described in the beginning of this section, the steady-state fairness and throughput performance can be evaluated with the transition matrix P and the steady-state probability vector π.

An example is provided as follows, where two channels are available and two flows are backlogged. In this case, average SNR $\overline{\gamma} = 15dB$, and the U-function $U_{i,j} = -R_{i,j}$. The credit range is $[-3, 3]$, indicating 7 possible credit states (K_1, K_2). In addition, the possible data rates are $0,\ 1$ and $2bit/symbol$ and each of them represents a channel state of the FSMC model as described in Section II-B. Thus vector $(C_{1,1}, C_{1,2}, C_{2,1}, C_{2,2})$ has $3^4 = 81$ possible states. So the number of possible states of V is $7{*}81 = 567$. We set an SNR threshold for all the flows so that if SNR is below the threshold the channel is regarded unavailable. In this article, when the SNR of a channel is below the value to transmit data with the 1 bit/symbol rate, the channel is not usable. The steady-state probabilities of these channel states are 0.1903, 0.6375 and 0.1722 respectively.

Although the achieved transition probability matrix is 567 by 567, which is not presentable in this article, Table I shows the steady-state probability p_k of each credit set $\underline{K}^q$, which includes credits of flows 1 and 2 respectively. The marginal probability is obtained by summing possibilities of all the states V_s whose credit set is $\underline{K}^q$. The states with smaller bias have large steady-state probabilities, i.e., under the proposed scheduling algorithm, the cases with higher fairness tend to happen as time goes on. Moreover, the probability of the states whose bias is 6 is as small as 5%, which means that unfair cases only happen occasionally. The average throughput of the example with two flows and two channels is $1.28\ bit/symbol$.

Comparisons with simulation results (both with and without credit bound represented by f'_k and f_k), are also provided in Table I. The simulation runs $20,000$ times. Other parameters in the simulation are the same as in the analysis. It is also noticed that the probabilities with credit bounds are a little bit larger than the corresponding unbounded results. It can be argued that it is due to the states omitted in the bounded case since fewer possible credit sets appear but the sum of these probability should still be 1. Small difference between results with and without credit bound suggests the credit bound of ± 3 is reasonable for such a scenario.

V. Simulation Results

In this section, we present simulation results based on the proposed scheduling algorithm. A wireless communication system with multiple independent channels is simulated. Each of the channels is simulated based on the finite-state Markov model described in Section II-B. Several experiments were carried out to characterize the fairness and throughput. It is assumed that the scheduler at the center node be aware of all the channel side information. Moreover, packets from different flows are assumed to have equal length. Theoretically, $U_{i,j}$ can be defined as any decreasing function of requested data rate $R_{i,j}$. In our simulations, it is defined as $-\lambda R_{i,j}$. In Table II and Table III, one can see the effect of λ on system fairness and throughput under different average SNRs (dB).

Efficiency is represented by the average throughput and fairness is measured by the weighted average value of bias $B = E\{\max_{i,j}(|k_i - k_j|)\}$. B_p and T_p are respectively the average bias and throughput of the proposed scheduler. B_b and

T_b are the average bias and throughput of the scheduler with binary channel status. Compared with the proposed scheduler, no matter how much data rate is allowed, a traditional binary-state scheduling regards our finely defined channel states above the SNR threshold as the same channel state. However, our algorithm allows PHY layer to differentiate the different data rates, leading to different throughput.

One can observe from Table II that the scheduler with binary channel state information achieves more fairness, i.e., smaller average bias. It can be explained as that the scheduler with binary channel state can be regarded as a special case of the proposed one where the U-function $U_{i,j}$ is always set to be 0. Thus the channel state information does not influence the scheduling process, and it is mainly the credit status that decides which packet to transmit. Hence fairness is paid more attention and better fairness performance is achieved by this binary scheduler.

Table III shows that the proposed scheduler achieves larger average throughput. It is due to the channel information part in the scheduling cost function that helps to avoid bad channel conditions and select favorable ones. In addition, it can be seen that when λ becomes larger, the weight of channel state becomes larger in the cost function and the average throughput is increased. At the same time, the fairness plays a less important role.

VI. Conclusions

The scheduling algorithm for wireless communication networks proposed in this article carefully implements scheduling with channel side information of multiple channels. Based on our design, a flow always tends to occupy channels in good conditions which provide satisfactory total throughput while fairness is also guaranteed. A scheduler can deal with flows with different types of QoS requirements simultaneously by defining *weights* and *U-functions* for flows. Weight of a flow represents the average data rate required by the flow. From both simulation and analytical results, probabilistic fairness and throughput advantages are shown.

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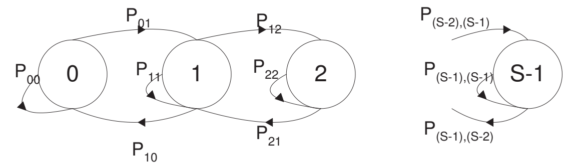

Fig. 1. A finite state Markov channel model.

TABLE I
Analysis and simulation results for a two-user-two-channel case

k_1	k_2	p_k	f'_k	f_k
0	0	0.2906	0.2897	0.2814
1	-1	0.2271	0.2297	0.2162
-1	1	0.2271	0.2271	0.2136
2	-2	0.0811	0.0806	0.0775
-2	2	0.0811	0.0791	0.0767
3	-3	0.0465	0.0471	0.0315
-3	3	0.0465	0.0467	0.0315

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TABLE II
Average Bias of schedulers

SNR	B_b	B_p					
		$\lambda = 1$	3	5	7	9	11
10	19.73	20.19	21.22	22.25	23.30	24.31	25.42
11	19.07	19.62	21.19	22.65	24.14	25.56	26.79
12	14.57	15.07	16.21	17.21	18.43	19.75	20.88
13	11.23	11.88	13.10	14.35	15.62	16.95	18.33
14	7.85	8.32	9.37	10.71	11.97	13.26	14.48
15	6.49	6.80	7.92	9.25	10.52	11.66	12.98

TABLE III
Average Throughput of schedulers

SNR	T_b	T_p					
		$\lambda = 1$	3	5	7	9	11
10	0.6768	1.51	1.52	1.53	1.54	1.55	1.56
11	0.9490	1.71	1.72	1.74	1.75	1.76	1.77
12	1.3226	1.91	1.92	1.94	1.94	1.96	1.97
13	1.6092	1.99	2.01	2.03	2.04	2.06	2.07
14	2.1074	2.40	2.41	2.43	2.44	2.45	2.47
15	2.3791	2.48	2.49	2.51	2.52	2.54	2.55