CROSS LAYER SCHEDULING ALGORITHMS FOR DIFFERENT RATE TRAFFIC IN OFDM BROADBAND WIRELESS SYSTEMS

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ABSTRACT

Cross layer optimization plays a key role in radio resource management of broadband wireless systems (BWS). Maximal SNR (MaxSNR) and Round Robin (RR) are two conventional scheduling strategies which emphasize efficiency and fairness respectively. Proportional Fair (PF) provides a tradeoff between efficiency and fairness. Here, we tailor PF to OFDM-based BWS (OPF). To meet QoS requirements for multi-rate services in multimedia systems, we propose two algorithms: Adaptive OPF (AOPF) and Multimedia AOPF (MAOPF). Under time varying and frequency selective fading wireless channel, system performances of OPF, AOPF, MAOPF are evaluated and compared with conventional MaxSNR and RR. We define user satisfaction rate and average user rate as optimization indication. Joint PHY and MAC simulation results show that OPF gives a good compromise between system throughput and fairness by providing the highest user satisfaction rate; MAOPF favors the high date rate by increasing throughput while providing the highest average user rate.

KEYWORDS

Scheduling algorithm, cross layer optimization, OFDM, wireless, multimedia services, QoS

1. INTRODUCTION

The scarcity of radio resources makes the efficient use of radio resources a very challenging task for 3G/4G wireless communication systems. Meanwhile, diverse QoS requirements and wireless channel conditions complicate scheduling and radio resource management. The main function of a scheduler is to intelligently allocate radio resources to achieve high system performance. Thus, the scheduling algorithm becomes the key component in optimizing the system across the physical (PHY) and Media Access Control (MAC) layers.

There are two conventional scheduling algorithms in packet data services: Round Robin (RR) and Maximum SNR (MaxSNR).. RR runs the scheduling in a fixed cycle. All active users are identified by their ID. Each user is allocated an equal, fixed number of time slot(s) in a ring fashion. Transmission service will not be re-granted to the same user before all other users had been served. In MaxSNR, all active users are periodically ranked by their reported SNR values. At every scheduling event, the user with the highest SNR value is selected and allocated a number of time slots depending on the service requirements and the availability of slot resources. It is possible for the same user to be re-scheduled for the next available resource provided that the user still possesses the highest SNR value. It has been widely accepted that RR has the advantage of being “fair” in the sense of giving equal slots to every user. It is also known that MaxSNR provides high system throughput.

In order to improve the fairness without sacrificing throughput too much, one should incorporate both aspects in the scheduling algorithm figure of merit. PF is one of many efforts emphasizing this issue. It was proposed in [1] and [2]. The scheduler selects the user with a maximal priority metric defined as:

$$j = \arg \max_{1 \le i \le N} D_i(t) / R_i(t-1) \quad (1)$$

where i is the user index, j is the selected user, N is the toal number of users, $D_i(t)$ is the current supportable data rate by the channel of the i[th] user and

$$R_i(t) = \begin{cases} 0 & i \neq j \\ (1 - \frac{1}{T_c}) R_j(t-1) + \frac{1}{T_c} D_j(t) & i = j \end{cases} \quad (2)$$

where T_c is the observation window, $R_i(t)$ is the data rate per unit time for the i^{th} user observed at time *"t"* and averaged over *Tc*. With this algorithm, a user is selected when it has good channel to keep system throughput high. In the meantime, users in bad channels are also considered since their low average rate $R_i(t\text{-}1)$ will increase their chance of being selected for the next scheduling event. The performance of this algorithm is a function of T_c. This algorithm is implemented in High Data Rate (HDR) networks of CDMA2000 1X, and has been studied by [3] and [4]. Their researches show that this algorithm does not provide enough fairness among users.

One of the proposed solutions to the unfairness problem in PF is the PF with Data Rate Control (DRC) exponent rule in [5]. The selection criteria is modified to: $j = \arg \max_{1 \le i \le N} \frac{D_i^n}{R_i}$, where n is a weighted parameter introduced to manage the relationship between the data rates of the users with different channel conditions. However, the control parameter n is a constant for all users, and two problems exist in this approach: 1) the control parameter is fixed in time and does not adapt to the current radio conditions of each user; 2) n is set to the same value for all users, thus impossible to choose a value for n that ensures fairness among all users at the same time.

As far as we know, PF and PF-DRC are fitted for and implemented in CDMA, WCDMA and TDMA/CDMA hybrid networks, but not in OFDM system which is a competitive candidate for 3G/4G Broadband Wireless Systems (BWS) [6], [7]. OFDM is well known for its flexibility and good performance in combating the frequency selective fading. It eliminates Inter Symbol Interference (ISI) by inserting a Cyclic Prefix (CP) longer than the delay spread of the channel. At the same time, due to its feature of multi-carrier, it can fulfill the ever-increasing demand of high data rate communication as well as the multi-media requirements. In terms of scheduling, OFDM based systems have two degrees of freedom: time and frequency. In [8] and [9] the authors do not describe specific scheduling algorithms in detail. Instead they describe the system architecture and available information that enables diverse scheduling algorithms. Recently, there has been a growing interest in applying this technique to the broadband multiple access system [10], [11]. It is also adopted by IEEE standards such as 802.11a, 802.11g and 802.16a. Therefore, it is of interest to develop scheduling algorithm that provides tradeoff between throughput and fairness in the context of OFDM signaling in order to optimize the joint PHY and MAC layer performance.

Some works have been done on scheduling algorithms that provide tradeoff between throughput and fairness in OFDM BWS. In paper [12], three modified PF schedulers are elaborated and fit into an OFDM system. The criterion is exactly the same as the formula in [1] and [2]. All three schemes could be regarded as frequency domain scheduler, which divide the subcarriers into several sub-bands. Each sub-band is scheduled according to the PF criteria, and the difference between the three schemes lies in how the average rate is updated. The first two schemes update the average rate subband-by-subband while the third scheme does the update when the allocation for all subbands is finished. In the time domain, during the allocation of subbands in the same slot, the average rate is supposed to stay constant. This is not true for fast fading wireless channels, and the scheme is not optimal because some information is lost in rate updating. In the frequency domain, depending on the total bandwidth, FFT size and the multipath, the correlation function of subcarriers is hard to predict. When several subcarriers are grouped together and assigned to the same scheduler, it is under the assumption that those subcarriers are highly correlated. This is not true for frequency selective fading channel. The scheme is not optimal because some information in the frequency domain is lost.

In paper [13], a scheduling scheme was proposed for OFDM systems, which considers the user priority and urgency level. In [14], an urgency and efficiency based packet scheduling (UEPS) scheme, which utilizes the head-of-line (HOL) delay and channel quality, was proposed for OFDM systems. However, [13] and [14] only considered scheduling for the MAC layer, but did not consider adaptive resource allocation (ARA) for the physical (PHY) layer. In paper [15], the author proposed a novel packet batch dependant scheduling scheme, which takes delay satisfaction, higher QoS coefficient and more data amount into account, meanwhile ARA is also applied. But the tradeoffs between system throughput and fairness are not emphasized.
To design scheduling algorithms for multimedia service in OFDM BWS which provide tradeoff between throughput and fairness, we make these efforts: 1) modify PF to eliminate the impact of observation window and we name it OPF; 2) propose two new algorithms, namely AOPF and MAOPF for multimedia service; 3) implement conventional algorithms RR, MaxSNR, PF and newly developed OPF, AOPF, MAOPF in OFDM BWS which is compliant with the IEEE 802.16a standard. We evaluate and compare the performance in efficiency and fairness. Efficiency is in terms of throughput and delay statistics; Joint PHY and MAC optimization indicator and fairness are evaluated in terms of user satisfaction rate and average user rate.

In this paper, the system is described in section 2; scheduling algorithms are introduced in section 3; simulation results are presented in section 4 and we draw the conclusions in section 5.

2. SYSTEM DESCRIPTION

We consider the downlink in a single cell. One Base Station (BS) is located at the cell center. N users are uniformly distributed within the cell radius.

2.1. Pass Loss

We choose the path loss model described in [16], which is applicable for scenarios in urban and suburban areas outside the high-rise core, where structures are of nearly uniform height. This is a typical environment for high speed data packet switched BWS such as 802.16a:

$$L_S = 40(1-4\times10^{-3}\Delta h_b)\log_{10}R - 18\log_{10}\Delta h_b + 21\log_{10}f + 80[dB] \tag{3}$$

Where L_S is pass loss in dB, R is the distance between the BS and Mobile Station (MS) in kilometers, f is the carrier frequency in MHz, and Δh_b is the BS antenna height in meters, measured from the average rooftop level. Please note that: 1) The path loss model is valid for a range of Δh_b from 0 to 50m; 2) L_S shall in no circumstances be less than the free-space loss; 3) Log-normal shadowing with a standard deviation of 10 dB is assumed.

2.2. Channel Model

In IEEE 802.16a standard, Non Line Of Sight (NLOS) is assumed. Multipath fading is modeled as Rayleigh. Each multipath component is time varying. During one frame, channel is supposed to keep unchanged.

2.3. IEEE 802.16a Requirements

IEEE 802.16a is the Wireless Metropolitan Area Network (WMAN) standard for BWS [17]. The OFDM parameters are extracted in table 1.

Table 1. IEEE 802.16a Standard – OFDM Mode

Frequency band (GHz)	2.500-2.520
Bandwidth (MHz)	20
OFDM number of	256
Used number of carriers	200
Ratio of CP	25%
Symbol duration	16µs
FFT time and Guard	12µs and 4µs
Frame length & Physical	2ms & 64µs (4 symbol
Overhead DL frame	25% of frame time
Coding and rate	RS, CC code; rate: ½,
Modulation scheme	QPSK, 16 QAM,

2.4. System Scenario

Based on the frame structure in table 1, the number of OFDM symbols in each frame, N_{symbol}, can be calculated. For each OFDM symbol, there are M_{tone} subcarrriers. N users are uniformly distributed in the cell. The distance between user i and the BS is d_i. Based on the previous frame, each user estimates its received SNR and sends this information in the uplink to the BS. Path loss, large scale shadowing, small scale time fading, and multipath effects are included in the estimated SNR computation:

$$P_r(dB) = P_t(dBm/Hz) - F_{ch}(dB) - L_s(dB) - P_{sh}(dB) - N_f(dBm/Hz) \quad (4)$$

where $P_r(dB)$ is the estimated received symbol SNR, P_t is the transmit Power Spectrum Density (PSD) in dBm/Hz, F_{ch} stands for the channel frequency response, L_S is the large scale path loss derived from equation (3), P_{sh} is the loss due to shadowing, N_f is the noise PSD floor. We choose N_f = -174dBm/Hz [18]. We define upper and lower limits of $P_r(dB)$ to be 200dB and -200dB respectively. This is to ensure that $P_r(dB)$ is in a reasonable range.

At the beginning of each frame, BS obtains $P_r(dB)$ from each user. First, the scheduler selects one MS based on the algorithm, which will be introduced in the next section. Second, it does bit allocation to the selected MS. In bit allocation, Adaptive Modulation and Coding (AMC) scheme is applied to different tones for the selected user. That means when the channel is good, higher modulation level and the corresponding coding scheme is applied. The number of bits each tone can carry is determined as:

$$C(bits/s/Hz) = \log_2[1 + (P_r(dB) - SNR_{marg} - SNR_{required})] \quad (5)$$

where C is the tone capacity; $P_r(dB)$ is derived from equation (4). SNR_{marg} is set to 3dB [18] to combat any unexpected interference. $SNR_{required}$ is the required SNR to guarantee symbol error rate below a certain margin, such as 10^{-4}. This is to meet the QoS requirements for different types of multimedia communications service.

No power control is adopted because OFDM systems are not as sensitive to power level as CDMA systems. Every subcarrier uses the constant maximal power to transmit data to the selected MS.

3. SCHEDULING ALGORITHMS

In this section, we first describe the time-frequency diagram and the common scheduling procedure. Second, the traffic model is given. Third, for evaluation purpose, system performance definition is outlined. Then, OPF is introduced. At last, AOPF and MAOPF are elaborated. For comparison purpose, two well-known algorithms, RR and MaxSNR, are also briefly described.

3.1. Two Dimensional Resource diagram and Common Scheduling Procedure

The inherent multicarrier nature of OFDM provides a high degree of flexibility in both time and frequency domains. In Fig. 1, in the time domain, each frame is composed of OFDM symbols. In the frequency domain, each OFDM symbol is composed of a number of tones, or carriers. We regard each

time-frequency unit as the smallest unit in our two dimensional resource. The overhead percentage of the frame is stated in table 1. Only traffic stream symbols are considered in this paper.
All algorithms have the common scheduling procedure. The scheduler is called at the beginning of each frame and it does the scheduling for the whole frame. The procedure has two steps. First is user selection: at each symbol, the scheduler picks up one MS from *N* number of users based on the algorithm criterion. Second is bit allocation: the scheduler dynamically allocates different number of bits on each tone based on channel conditions and the QoS requirement of the selected MS. AMC is applied according to IEEE 802.16a standard.

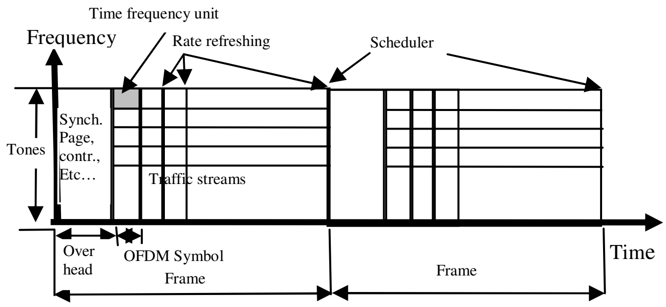

Figure 1. Time-Frequency diagram

3.2. Traffic Model

Multimedia service is one of the features of 3G/4G wireless networks. The system should give good performance for all types of traffic. We choose exponential distribution since it is a typical traffic model. The inter-arrival time between two packets is a random variable. Its probability density function with mean $1/\lambda$ is:

$$f_X(x) = \lambda e^{-\lambda x}, x > 0 \quad (6)$$

In multimedia services, there are different data rate requirements. We assume that two rates are in demand representing two kinds of services. The high rate is 10 times the low rate. In other words, the inter-arrival time of the high rate is 1/10 times the low rate.

3.3. System Performance Definitions

The performance of the system will be evaluated in terms of system throughput, mean packet delay, delay variance, user satisfaction rate and average user rate.
1) Throughput: the received number of information bits per second (bps). It equals the total number of bits that are received and accepted by the MS within a time period T divided by that time period.
2) Delay: the mean value of the packet completion time, measured from the instant of packet arrival at the BS's buffer queue until the MS receives this packet.
3) Packet loss ratio: the ratio of the number of dropped packets and the total number of generated packets. In [19], it is stated that the compressed video can tolerate a maximum delay of 100ms while preserving good real-time interactivity. To have better video quality, we define that when the packet delay is larger than 80ms, the packet is dropped.
4) User satisfaction rate: ratio of the satisfied number of MS and the total number of MS. A MS is defined as satisfied [20] when the following criteria are met: 1). The MS does not get blocked when arriving the system; 2). the ratio of the received bits and total generated traffic of this user is greater than a threshold; 3). the user's packet does not get dropped.
5) Average user rate: user rate is defined as the ratio of received bits and the total generated bits for this user. Average user rate is the average ratio of all users.

3.4. Scheduling Algorithms

1). Modified PF: OPF

As shown in equation (1) and (2), the performance of PF depends on the observation window T_c. To provide long term fairness from user activation to termination, in other words, to eliminate the impact of T_c, we define the OPF criterion as:

$$j = \arg \max_{1 \le i \le N} \frac{D_i(t)}{\overline{R_i(t)}}, \qquad (7)$$

where $D_i(t)$ is the current supportable rate to user i, $\overline{R_i(t)}$ is the average rate experienced by user i, which is calculated as the total number of bits received divided by the time from its activation to the current instant, under the condition that this user is active all the time. In the case of user termination, it de-registers from the BS and the scheduler will not count it as one of its users.

The detailed flow chart of OPF is given in Fig. 2. As shown in the time-frequency diagram, OPF is called at the beginning of each frame. $\overline{R_i(t)}$ is updated at each OFDM symbol. The "resolution" of fairness is hence improved compared with rate updating at each frame. Path loss, shadowing, time varying, frequency selective fading and background noise are included in the computation of received SNR. QoS requirements and external interference are added on top of SNR to calculate tone capacity. Supportable rate $D_i(t)$ of current symbol is the sum of all tone capacities with the constraint of AMC defined by the 802.16a standard. Through this design, OPF is supposed to achieve high system throughput while ensuring user fairness in terms of highest user satisfaction rate.

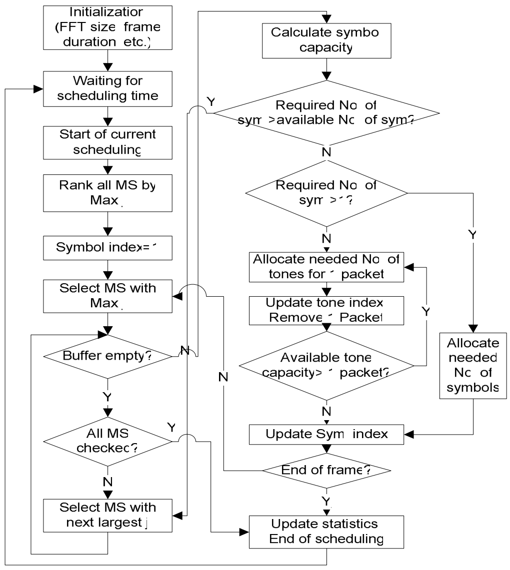

Figure 2. OPF flow diagram

2). Proposed Algorithm: AOPF

Inspired by [5], we may add an adaptive exponent to the numerator of OPF and name it Adaptive OPF (AOPF). The criterion is:

$$G_I(t) = \arg : \max_{1 \le i \le N-1} \{G_i(t)\} \qquad (8)$$

where I is the selected user index, i is the user index, and the total number of users is N. $G_i(t)$ is a function of time t:

$$G_i(t) = \alpha_i \frac{D_i^{n_i}(t)}{\overline{R_i(t)}} \qquad (9)$$

$D_i(t)$ and $\overline{R_i(t)}$ are the same as in (7). $D_i(t)$ is updated each frame, while $\overline{R_i(t)}$ is updated each symbol. α_i is equal to 1 or 0 when this user is active or idle respectively, that is:

$$\alpha_i = \begin{cases} 0 & idle \\ 1 & active \end{cases} \tag{10}$$

The n_i is the newly introduced adaptive exponent, it is also a function of time t and it is updated each symbol. For simplification, we use another index j to represent the time, or symbol index. Assume all users are active and there are M symbols per frame, we simplify the criterion as:

$$I = \arg \max_{\substack{1 \le i \le N \\ 1 \le j \le M}} \frac{D_{i,j}^{n_{i,j}}}{\overline{R_{i,j}}} \tag{11}$$

In AOPF, the crucial part is to obtain the adaptive exponent $n_{i,j}$ for each user at every symbol. That is to determine the value of adaptive exponent matrix:

$$\begin{vmatrix} n_{1,0} & n_{1,1} & \cdots & n_{1,M} \\ n_{2,0} & n_{2,1} & \cdots & n_{2,M} \\ \vdots & & & \vdots \\ n_{N,0} & n_{N,1} & \cdots & n_{N,M} \end{vmatrix}$$

When regarding OPF as a special case of AOPF where the exponent equals 1, we set the initial value of the exponent for each user to 1. For each user, the purpose of adjusting $n_{i,j}$ is to equalize $\frac{D_{i,j}^{n_{i,j}}}{\overline{R_{i,j}}}$ in current symbol to previous symbol. Take user 1, e.g. Let $c_{1,0} = \frac{{D_{1,0}}^{n_{1,0}}}{\overline{R_{1,0}}} = \frac{D_{1,0}}{\overline{R_{1,0}}}$ be the initial value for this user in symbol 0, then for the next symbol, we have: $n_{1,1} = \frac{\log c_{1,0} + \log \overline{R_{1,1}}}{\log D_{1,1}}$

In general, for user i in symbol j, we have:

$$n_{i,j} = \frac{\log c_{i,(j-1)} + \log \overline{R_{i,j}}}{\log D_{i,j}} \tag{12}$$

Thus, we determine the adaptive exponent of each user, and it is updated symbol by symbol.

3). Proposed Algorithm: MAOPF

In OPF and AOPF, we only consider the supportable data rate and experienced data rate. In other words, we are trying to offer fairness under the consideration of user location and to capture the fast changing wireless channel. We do not take the data rate requirements of each user into account. This may be fair enough for all users having the same rate. This may not be fair when users have different data rate requirement which is an important feature for multimedia services. The next step is to define a new criterion which takes data rate requirements into consideration. Hence, the new algorithm will provide fairness to different data rate users without sacrificing too much of the system throughput.

Based on AOPF, we add another parameter R_{i_r}, the required data rate of user i and name it Multimedia Adaptive OPF (MAOPF). The criterion can be written as:

$$G_I(t) = \arg : \max_{1 \le i \le N-1} \{G_i(t)\} \tag{13}$$

$$G_i(t) = \alpha_i \frac{D_i^{n_i}(t)}{\left(\frac{\overline{R_i(t)}}{R_{i\_r}(t)} \right)} = \alpha_i \frac{D_i^{n_i}(t) \cdot R_{i\_r}(t)}{\overline{R_i(t)}} \tag{14}$$

where I, i, N, $D_i(t)$, $\overline{R_i(t)}$, α_i, n_i are defined the same as AOPF. R_{i_r} is updated each symbol considering of capturing the randomness of the traffic for this user. Assume all users are active, we simplify the criterion as:

$$I = \arg \max_{\substack{1 \le i \le N \\ 1 \le j \le M}} \frac{D_{i,j}^{n_{i,j}} \cdot R_{i\_r,j}}{\overline{R_{i,j}}} \tag{15}$$

Follow the same deduction process in AOPF, we have:

$$n_{i,j} = \frac{\log c_{i,(j-1)} + \log \overline{R_{i,j}} - \log R_{i\_r,j}}{\log D_{i,j}} \quad (16)$$

The initial value is: $c_{1,0} = \frac{D_{1,0}^{\ n_{1,0}} \cdot R_{1\_r,0}}{\overline{R_{1,0}}} = \frac{D_{1,0} \cdot R_{1\_r,0}}{\overline{R_{1,0}}}$, under the setting $n_{i,0}$=1 for all users.

For comparison, we also implement RR and MaxSNR algorithms. In RR, at each symbol time, the scheduler picks up one MS by their sequence number from non-empty queues. In MaxSNR, the scheduler chooses the MS with the largest SNR value as defined in equation (4). In evaluating the system throughput, RR and MaxSNR are supposed to give the lower and upper bound respectively.

4. SIMULATION RESULTS

4.1. Common Assumption for All Algorithms

1). we regard one packet as the smallest unit of requested traffic. When the channel condition cannot support even one packet to any MS, we skip this frame;
2). Packet size is a constant and does not change;
3). Channel is supposed to remain stable during one frame;
4). Different MS cannot share the same symbol;
5). AMC is applied to each tone when MS is selected according to the specific frequency response of that tone;
6). Jake's model [21] is chosen to simulate the time varying, Rayleigh distribution fading channel. We set f_dT_s=0.001 to simulate slow fading, where f_d is the maximal Doppler frequency, T_s is the duration of an OFDM symbol.

4.2. Simulation Parameters

1). Cell radius: 1.5 kilometer;
2). Number of MSs: 15, with 8 high rate and 7 low rate;
3). Packet size: 180 bits/packet
4). Frame structure and OFDM parameters as in table 1.
5). Path loss parameters: Antenna height: 15m; MS distribution: uniform distribution within cell radius; Shadowing: lognormal distribution with mean of 10dB.
6). Channel parameter:number of multipath components: 3;
7). Threshold of user satisfaction rate: 95%
8). Delay tolerance: 80ms.
9). Other parameters: Input PSD: 145dBm/Hz; N_f: -174dBm/Hz; SNR_{margin}: 3dB; $SNR_{required}$: 7dB

4.3. Simulation Results and Discussions

Joint PHY and MAC layer simulation is fully implemented. We show the performance for the high data rate group of user. In the IEEE standard, the data rate of 802.11a can reach 54Mbps. In IEEE 802.16a, it can reach up to 70Mbps which accommodates heavier traffic. Thus, we focus on the performance of scheduling schemes at high traffic load of 55Mbps and up.

Fig. 3 is throughput vs. traffic load. Under high traffic load, as expected, RR has the lowest throughput. OPF improves the system throughput by 10Mbps compared to RR. AOPF is almost the same as MaxSNR. To our surprise, MAOPF has the highest throughput and breaks MaxSNR which was the expected upper bound. Fig. 4 is the packet loss ratio comparison. Under high traffic, both of our proposed algorithms improve a lot in the packet drop ratio performance compared with the known algorithms. Fig. 5 shows the mean delay comparison. Under high traffic, OPF is better than RR and our proposed AOPF and MAOPF outperform OPF, but not as good as MaxSNR. Fig. 6 is the delay variance with MAOPF as the best.

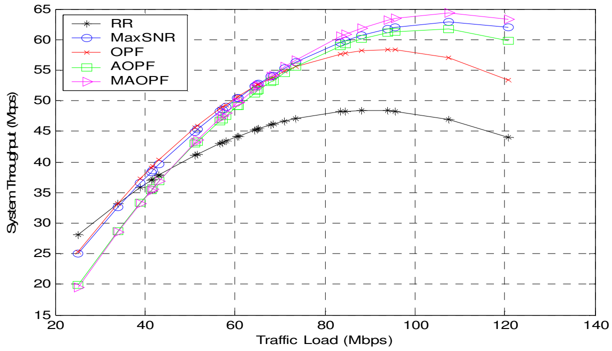

Figure 3. System throughput of the 5 algorithms

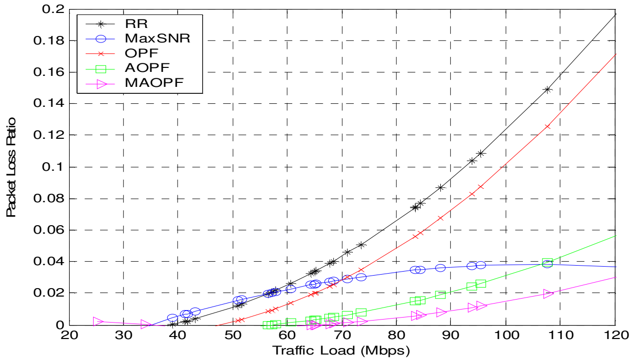

Figure 4. Packet loss ratio of the 5 algorithms

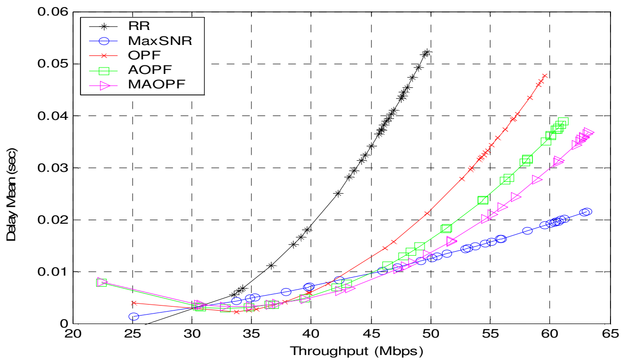

Figure 5. Mean delay of the 5 algotithms

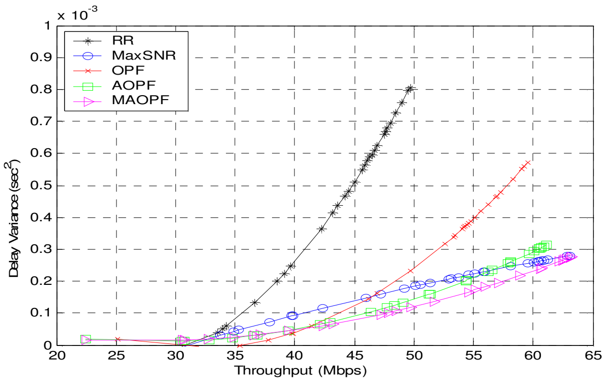

Figure 6. Delay variance of the 5 algorithms

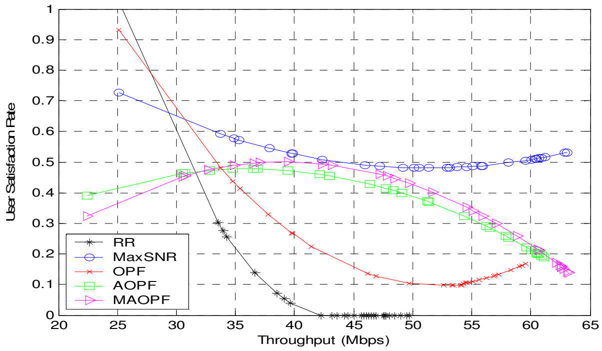

Figure 7. User satisfaction rate of the 5 algorithms

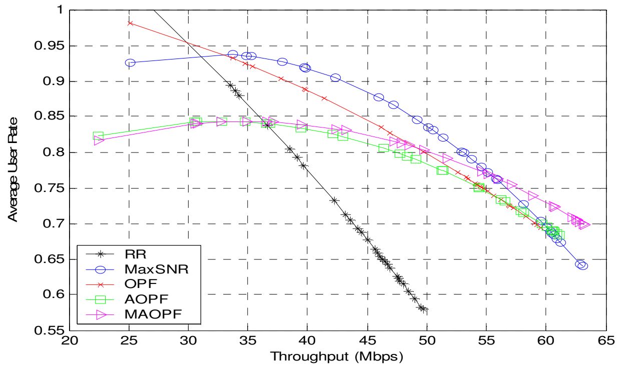

Figure 8. Average user rate of the 5 algorithms

In Fig. 7, The target user satisfaction threshold is set at 95% to give good QoS for multimedia traffic. When throughput is larger than 35Mbps, both of our proposed algorithms perform better than OPF. The criterion of OPF indicates that the scheduler not only favors those MSs with high supportable rate, but also its experienced rate. Our proposed AOPF gives extra weight to users with good channel. In addition, MAOPF considers the required rate as well. So, it is reasonable that OPF is better than RR, AOPF is better than OPF, and MAOPF is better than AOPF. Compared with OPF, both of our proposed algorithms improve the system throughput. At the same time, it also gives enough fairness to MSs whose channel is not so good in the past. In other words, if a user does not receive service at previous frame due to its bad channel condition, then its precedence will increase at the next frame. This means the served user will have less chance to access the next frame. Therefore, a user will not “starve” too long even if its channel condition is not so good.
In Fig. 8, AOPF is not as good as OPF when throughput is between 36~55Mbps, but slightly better under high traffic. MAOPF outperforms OPF when throughput is larger than 48Mbps and it becomes the best when above 55Mbps.

For low traffic load below 35Mbps, RR performs the best except the packet drop ratio. In between 35 and 55 Mbps, for throughput, OPF is the best, AOPF and MAOPF are in between MaxSNR and RR. For packet drop ratio, from good to bad are: MaxSNR, RR, OPF, AOPF, MAOPF. For delay, AOPF and MAOPF are better most of this range. For user satisfaction rate, from good to bad are: MaxSNR, MAOPF, AOPF, OPF and RR. For average user rate, MaxSNR is the best, MAOPF is better than AOPF, and MAOPF outperforms OPF above 48Mbps, RR is the worst in all this range.

We have omitted the simulation results for the case of all users. The results show that: Under high traffic, OPF improves system throughput, packet drop ratio, delay statistics as well as user satisfaction rate and average user rate. While AOPF and MAOPF only improve the throughput, packet drop ratio and delay compared with OPF, they cannot offer better user satisfaction rate and average user rate. In the future, we may develop new algorithms aiming at improving throughput and delay without losing user fairness. Also, to evaluate all the algorithms in a more complicated multimedia traffic environment is a subject for future research.

5. CONCLUSIONS

Optimizing system performance by cross layer scheduling algorithm design plays an important role in broadband wireless system (BWS). In seeking better tradeoff between throughput and fairness in scheduling algorithms for multi-rate multimedia service, we modify Proportional Fair (PF) and implement it in OFDM BWS The modified PF is named as OFDM PF (OPF). In addition we propose two new algorithms for different rate multimedia services: Adaptive OPF (AOPF) and Multimedia AOPF (MAOPF). System performance of the algorithms are evaluated and compared with conventional Round Robin (RR) and Maximal SNR (MaxSNR).

Joint PHY and MAC layer simulation results show that, under high traffic, compared with RR, OPF increases the system throughput by 10Mbps, improves packet drop ratio, delay statistics, user satisfaction rate as well as average user rate. Regarding the new developed algorithms, under high traffic, compared with OPF, AOPF and MAOPF improve system throughput, packet drop ratio, delay statistics, user satisfaction rate and average user rate. In addition, MAOPF favors high date rate group users by breaking through the current throughput upper bound. At the same time, it improves average user rate compared with MaxSNR when throughput is larger than 55Mbps. Our future work will include developing new algorithms aimed at improving throughput without losing user satisfaction rate in more complicated multimedia traffic model.

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