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Introduction to Packet Scheduling Algorithms for Communication Networks

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1. Introduction

As implied by the word "packet scheduling", the shared transmission resource should be intentionally assigned to some users at a given time. The process of assigning users' packets to appropriate shared resource to achieve some performance guarantee is so-called packet scheduling.

It is anticipated that packetized transmissions over links via proper packet scheduling algorithms will possibly make higher resource utilization through statistical multiplexing of packets compared to conventional circuit-based communications. A packet-switched and integrated service environment is therefore prevalent in most practical systems nowadays. However, it will possibly lead to crucial problems when multiple packets associated to different kinds of Quality of Service (QoS) (e.g. required throughput, tolerated delay, jitter, etc) or packet lengths competing for the finite common transmission resource. That is, when the traffic load is relatively heavy, the first-come-first-serve discipline may no longer be an efficient way to utilize the available transmission resource to satisfy the QoS requirements of each user. In such case, appropriate packet-level scheduling algorithms, which are designed to schedule the order of packet transmission under the consideration of different QoS requirements of individual users or other criteria, such as fairness, can alter the service performance and increase the system capacity . As a result, packet scheduling algorithms have been one of the most crucial functions in many practical wired and wireless communication network systems. In this chapter, we will focus on such topic direction for complete investigation.

Till now, many packet scheduling algorithms for wired and wireless communication network systems have been successfully presented. Generally speaking, in the most parts of researches, the main goal of packet scheduling algorithms is to maximize the system capacity while satisfying the QoS of users and achieving certain level of fairness. To be more specific, most of packet scheduling algorithm proposed are intended to achieve the following desired properties:

1. Efficiency:

The basic function of packet scheduling algorithms is scheduling the transmission order of packets queued in the system based on the available shared resource in a way that satisfies the set of QoS requirements of each user. A packet scheduling algorithm is generally said to

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be more efficient than others if it can provide larger capacity region. That is, it can meet the same QoS guarantee under a heavier traffic load or more served users.

2. Protection:

Besides the guarantees of QoS, another desired property of a packet scheduling algorithm to treat the flows like providing individual virtual channels, such that the traffic characteristic of one flow will have as small effect to the service quality of other flows as possible. This property is sometimes refered as *flow isolation* in many scheduling contexts. Here, we simply define the term flow be a data connection of certain user. A more formal definition will be given in the next section.

Flow isolation can greatly facilitate the system to provide flow-by-flow QoS guarantees which are independent of the traffic demand of other flows. It is beneficial in several aspects, such as the per-flow QoS guarantee can be avoided to be degraded by some ill-behavior users which send packet with a higher rate than they declared. On the other hands, a more flexible performance guarantee service scheme can also be allowed by logically dividing the users which are associated to a wide range of QoS requirements and traffic characteristic while providing protection from affecting each other.

3. Flexibility:

A packet scheduling algorithm shall be able to support users with widely different QoS requirements. Providing applications with vast diversity of traffic characteristic and performance requirements is a typical case in most practical integrated system nowadays.

4. Low complexity:

A packet scheduling algorithm should have reasonable computational complexity to be implemented. Due to the fast growing of bandwidth and transmission rate in today's communication system, the processing speed of packets becomes more and more critical. Thus, the complexity of the packet scheduling algorithm is also of important concern.

Due to the evolution process of the communication technology, many packet scheduling algorithms for wireless systems in literatures are based on the rich results from the packet scheduling algorithms for wired systems, either in the design philosophy or the mathematical models. However, because of the fundamental differences of the physical characteristics and transmission technologies used between wired and wireless channels, it also leads to some difference between the considerations of the packet scheduling for wired and wireless communication systems. Hence, we suggest separate the existing packet scheduling algorithms into two parts, namely, wired ones and wireless ones, and illustrate the packet scheduling algorithms for wired systems first to build several basic backgrounds first and then go to that for the wireless systems.

The rest of the chapter is outlined as follows. In Section 2, we will start by introducing some preliminary definition for preparation. Section 3 will make a overview for packet scheduling algorithms in wired communication systems. Comprehensive surveys for packet scheduling in wireless communication systems will then included in Section 4. In Section 5, we will employ two case studies for designing packet scheduling mechanisms in OFDMA-based systems. In Section 6, summary and some open issues of interest for packet scheduling will be addressed. Finally, references will be provided in the end of this chapter.

2. Preliminary definitions

The review of the packet scheduling algorithms throughout this chapter considers a packet-switched single server. The server has an outgoing link with transmission rate C. The main task of the server is dealing with the packets input to it and forwarding them into the outgoing link. A packet scheduling algorithm is employed by the server to schedule the appropriate forwarding order to the outgoing link to meet a variety of QoS requirements associated to each packet. For wireline systems, the physical medium is in general regarded as stable and robust. Thus the packet error rate (PER) is usually ignored and C can be simply considered as a constant with unit bits/sec. This kind of model is usually referred as ***error-free channel*** in literatures. On the other hands, for wireless systems, the situation can become much more complicate. Whether in wireless networks with short transmission range (about tens of meters) such as WLAN and femtocell or that with long transmission range (about hundreds of meters or even several kilometers) such as the macrocell environments based on WCDMA, WiMAX and LTE, the packet transmission in wireless medium suffers location-dependent path loss, shadowing, and fading. These impairment make the PER be no longer ignorable and the link capacity C may also become varying (when adaptive modulation and coding is adopted). This kind of model is usually referred as ***error-prone channel*** in literatures.

Each input packet is associated to a ***flow***. Flow is a ***logical*** unit which represents a sequence of input packets. In practice, packets associated to the same flows often share the same or similar quality of service (QoS) requirement. There should be a ***classifier*** in the server to map each input packets to appropriate flows.

The QoS requirement of a flow is usually characterized by a set of ***QoS parameters***. In practice, the QoS parameters may include tolerant delay or tolerant jitter of each packet, or data rate requirement such as the minimum required throughput. The choice of QoS parameters might defer flow by flow, according to the specific requirement of different services. For example, in IEEE 802.16e [47], each data connection is associated to a service type. There are totally five service types to be defined. That is, unsolicited grant service (UGS), real-time polling service (rtPS), extended real-time polling service (ertPS), non-real-time polling service (nrtPS), and best effort (BE). Among these, rtPS is generally for streaming audio or video services, and the QoS parameters contains the minimum reserved rate, maximum sustained rate, and maximum latency tolerant. On the other hands, UGS is designed for IP telephony services without silence suppression (i.e. voice services with constant bit rate). The QoS parameters of UGS connections contains all the parameters of rtPS connections and additionally, it also contains a parameter, jitter tolerance, since the service experiment of IP telephony is more sensitive to the smoothness of traffic. Moreover, for nrtPS, which is mainly designed for non-real-time data transmission service such as FTP, the QoS parameters contains minimum reserved data rate and maximum sustained data rate. Unlike rtPS and UGS, which required the latency of each packet to be below certain level, nrtPS is somewhat less sensitive to the packet latency. It allows some packets to be postponed without degrading the service experiment immediately, however, an average data rate should still be guaranteed, since throughput is of the most concern for data transmission services.

The server can be further divided into two categories, according to the ***eligible time*** of the input packets. Eligible time of a packet is defined as the earliest time that the packet begins being transmitted. Additionally, a packet is called eligible when it is available to be transmitted by the server. If all packets immediately become eligible for transmission upon arrival, the system is called ***work-conserving***, otherwise, it is called ***nonwork-conserving***. A direct consequence of a system being work-conserving is that the server is never idle whenever there are packets queued in the server. It always forwards the packets when the queues are not empty.

3. Packet scheduling algorithms in wireline systems

In this section, we will introduce several representative packet scheduling algorithms of wireline systems. Their merits and expense will be examined respectively.

3.1 First Come First Serve (FCFS)

FCFS may be the simplest way for a scheduler to schedule the packets. In fact, FCFS does not consider the QoS parameters of each packets, it just sends the packets according to the order of their arrival time. Thus, the QoS guarantee provided by FCFS is in general weak and highly depends on the traffic characteristic of flows. For example, if there are some flows which have very bursty traffic, under the discipline of FCFS, a packet will very likely be blocked for a long time by packets burst which arrives before it. In the worst case, the unfairness between different flows cannot be bounded, and the QoS cannot be no longer guaranteed. However, since FCFS has the advantage of simple to implement, it is still adopted in many communication networks, especially the networks providing *best effort* services. If some level of QoS is required, then more sophisticated scheduling algorithm is needed.

3.2 Round Robin

Round Robin (RR) scheme is a choice to compensate the drawbacks of FCFS which also has low implementation complexity. Specifically speaking, newly arrival packets queue up by flow such that each flow has its respective queue. The scheduler polls each flow queue in a cyclic order and serves a packet from any-empty buffer encountered; therefore, the RR scheme is also called flow-based RR scheme. RR scheduling is one of the oldest, simplest, fairest and most widely used scheduling algorithms, designed especially for time-sharing systems. They do offer greater fairness and better bandwidth utilization, and are of great interest when considering other scenarios than the high-speed point-to-point scenario. However, since RR is an attempt to treat all flows equally, it will lead to the lack of flexibility which is essential if certain flows are supported to be treated better than other ones.

3.3 Strict priority

Strict priority is another classical service discipline which assigns *classes* to each flow. Different classes may be associated to different QoS level and have different *priority*. The eligible packets associated to the flow with higher-priority classes are send ahead of the eligible packets associated to the flow with lower-priority classes. The sending order of packets under strict priority discipline only depends on the classes of the packets. This is why it called "strict" since the eligible packets with lower-priority classes will never be sent before the eligible packets with higher-priority classes. Strict priority suffers from the same problem as that of FCFS, since a packet may also wait arbitrarily long time to be sent. Especially for the packets with lower-priority classes, they may be even starved by the packets with higher-priority classes.

3.4 Earliest Deadline First (EDF)

For networks providing real-time services such as multimedia applications, earliest deadline first (EDF) [5][6] is one of the most well-known scheduling algorithms. Under EDF discipline, each flow is assigned a tolerant delay bound d_i; a packet j of flow i arriving at time a_{ij} is naturally assigned a deadline $a_{ij} + d_i$. Each eligible packet is sent according to the increasing order of their deadlines. The concept behind EDF is straightforward. It essentially schedules the packets in a greedy manner which always picks the packets with the closest deadline. Compare with strict priority discipline, we can regard EDF as a scheduling algorithm which provides ***time-dependent priority*** [8] to each eligible packet. Actually, the priority of an eligible packet under EDF is an increasing function of time since the sending order in EDF is according to the closeness of packets' deadlines. This fact allows the guarantee of QoS if the traffic characteristic of each flow obeys some specific constraint (e.g. the incoming traffic in a time interval is upper bounded by some amount).Define the traffic envelope $A_i(t)$ is the amount of flow i traffic entering the server in any interval of length t. The authors in [9] and [13] proved that in a work-conserving system, the necessary and sufficient condition for the served flows are schedulable (i.e. each packet are guaranteed to be sent before its deadline expires) , which is expressed by

$$\sum_i A_i(t-d_i) + l_{\max} I_{\{d_{\min} \le t \le d_{\max}\}} \le Ct \tag{3.1}$$

where C is the outgoing link capacity as described in section 2, l_{max} is the maximum possible packet size among all flows, $d_{min} = min_i\{d_i\}$, $d_{max} = max_i\{d_i\}$, $I_{\{event\}}$ is the indicator function of event E.

An important result of EDF is that it has been known to be the optimal scheduling policy in the sense that it has the largest *schedulable region* [9]. More specifically, given N flows with traffic envelopes $A_i(t)$ ($i = 1,2, \ldots, N$), and given a vector of delay bounds $\mathbf{d} = (d_1, d_2, \ldots d_N)$, where d_i is the to delay bound that flow i can tolerate. It can be proved that if $\mathbf{d}$ is schedulable under a scheduling algorithm π, then $\mathbf{d}$ will also be schedulable under EDF.

Although EDF has optimal schedulable region, it encounters the same drawback as that of FCFS and strict priority disciplines. That is, the lack of protection between flows which introduces weak flow isolation (see section 1). For example, if some flows do not have bounded traffic envelope, that is, $A_i(t)$ can be arbitrary large (or at least, very large) for some i, then the condition in (3.1) can't no longer be guaranteed to be satisfied, and no QoS guarantee can be provided to any flows being served. In the next section, we will introduce generalized processor sharing (GPS) discipline, which can provide ideal flow isolation property. The lack of flow isolation of EDF is often compensated by adopting ***traffic shaper*****s** to each flow to shape the traffic envelopes and bound the worst-case amount of incoming traffic of per flow. There are also some modified versions of EDF proposed to provide more protection among flows, such as [7] [10].

3.5 Generalized Processor Sharing (GPS)

Generalized processor sharing (GPS) is an ideal service discipline which provides perfect flow isolation. It assumes that the traffic is infinitely divisible, and the server can serve multiple flows simultaneously with rates proportional to the ***weighting factors*** associated to each flow. More formally, assume there are N flows, and each flow i is characterized by a weighting factor w_i. Let $Si(\tau,t)$ be the amount of flow i traffic served in an interval (τ,t) and a flow is ***backlogged*** at time t if a positive amount of that flow's traffic is queued at time t. Then, a GPS server is defined as one service discipline for which

$$\frac{S_i(\tau,t)}{S_j(\tau,t)} \ge \frac{w_i}{w_j}, j = 1,2,\ldots,N \tag{3.2}$$

For any flow i that is continuously backlogged in the interval (τ,t).
Summing over all flow j, we can obtain:

$$S_i(\tau,t)\sum_j w_j \geq (t-\tau)Cw_i$$

that is, when flow i is backlogged, it is guaranteed a minimum rate of

$$g_i = \frac{w_i}{\sum_j w_j}C$$

In fact, GPS is more like an idealized model rather than a scheduling algorithm, since it assumes a fluid traffic model in which all the packets is infinitely divisible. The assumptions make GPS not practical to be realized in a packet-switched system. However, GPS is still worth to remark for the following reasons:

1. It provides following attractive ideal properties and can be a benchmark for other scheduling algorithms.
   a. **Ideal resource division and service rate guarantee**
   GPS assumes that a server can serve all backlogged flows simultaneously and the outgoing link capacity C can be perfectly divided according to the weight factor associated to each backlogged flow. It leads to ideal flow isolation in which each flow can be guaranteed a minimum service rate independent of the demands of the other flows. Thus, the delay of an arriving bit of a flow can be bounded as a function of the flow's queue length, which is independent of the queue lengths and arrivals of the other flows. According to this fact, one can see that if the traffic envelope of a flow obeys some constraint (e.g. leaky buckets) and is bounded, then the traffic delay of a flow can be guaranteed. Schemes such as FCFS and strict priority do not have this property. Compare to EDF, since the delay bound provided by GPS is not affected by the traffic characteristic or queue status of other flows, which makes the system more controllable and be able to provide QoS guarantee in per-flow basis.
   b. **Ideal flexibility**
   By varying the weight factors, we can enjoy the flexibility of treating the flows in a variety of different ways and providing widely different performance guarantees.
2. A packet-by-packet scheduling algorithm which can provide excellent approximation to GPS has been proposed [1]. This scheduling algorithm is known as packet-by-packet GPS (PGPS) or weighted fair queueing (WFQ). In the later section, we will discuss the operation and several important properties of PGPS in more detail.

3.6 Packet-by-packet Generalized Processor Sharing (PGPS)

PGPS is a scheduling algorithm which can provide excellent approximation to the ideal properties of GPS and is practical enough to be realized in a packet-switched system. The concept of PGPS is first proposed in [4] under the name Weighted Fair Queueing (WFQ). However, a great generalization and insightful analysis was done by Parekh and Gallager in the remarkable paper [1] and [2]. The basic idea of PGPS is simulating the transmission order of GPS system. More specific, let F_p be the time at which packet p will depart (finish service) under GPS system, then the basic idea of PGPS is to approximate GPS by serving the packets in increasing order of F_p. However, sometimes there is no way for a work-conserving system to serve all the arrival packets in the exactly the same order as that of corresponding GPS system. To explain it, we make the following observations:

1. The busy period (the time duration that a server continuously sends packets) of GPS and PSPS is identical, since GPS and PGPS are all work-conserving system, the server will never idle and send packets with rate C when there are unfinished packets queued in the system.
2. When the PGPS server is available for sending the next packet at time τ, the next packet to depart under GPS may not have arrived at time τ. It's essentially due to the fact that a packet may depart earlier than the packets which arrive earlier than it under GPS. ***A packet may arrive too late to be send in PGPS system***, at this time, if the system is work-conserving, the server should pick another backlogged packet to send, and this would conflict the sending order under GPS system. Since we do not have additional assumption to the arrival pattern of packets here, there is no way for the server to be both work-conserving and to always serve the packet in increasing order of F_p.

To preserve the property of work-conserving, the PGPS server picks the first packet that would complete service in the GPS simulation. In other words, if PGPS schedules a packet p at time τ before another packet p' that is also backlogged at time τ, then packet p cannot leave later than packet p' in the simulated GPS system.

We have known the basic operation of PGPS, now a natural question arises: how well does PGPS approximate GPS? To answer this question, we may attempt to find the worst-case performance under PGPS compared to that of GPS. So we ask another question: how much later packets may depart the system under PGPS relative to GPS? In fact, it can be proved that let the G_p be the time at which packet p departs under PGPS, then

$$G_p - F_p \le \frac{L_{\max}}{C}$$

where L_{max} is the maximum packet length. That is, the depart time of a packet under PGPS system is not later than that under GPS system by more than the time of transmitting one packet. To verify this result, we first present a useful property:

Lemma 1 ***Let p and p′ be packets in a GPS system at time τ and suppose that packet p complete service before packet p′ if there are no arrivals after time τ. Then packet p will also complete service before packet p′ for any pattern of arrivals after time τ***

Proof.

The flows to which packet p and p' belong are backlogged at time τ. By (3.2), the ratio of the service received by these flows is independent of future arrivals. ■

Now we have prepared to prove the worst-case delay of PGPS system.

Theorem 1 ***For all packet p, let G_p and F_p be the departure time of packet p under PGPS and GPS systems, respectively. Then***

$$G_p - F_p \le \frac{L_{\max}}{C}$$

where L_{max} is the maximum packet length, and C is the outgoing link capacity.

Proof.

As observed above, the busy periods of GPS and PGPS coincide, that is, the GPS server is in a busy period if and only if the PGPS server is in a busy period. Hence it suffices to prove the theorem by considering one busy period. Let p_k be the k-th packet in the busy period to depart under PGPS and let its length be L_k. Also let t_k be the time that p_k depart under PGPS and u_k be the time that p_k departs under GPS. Finally, let a_k be the time that p_k arrives. It should be first noted that, if the sending order in a busy period under PGPS is the same as that under GPS, then it can be easily verified the departure time of the packets under PGPS system are earlier or equal to those under GPS system. However, since the busy periods of GPS and PGPS systems coincide, there are only two possible cases:

1. The departure times of all the packets under PGPS system in a busy period are all the same as those of corresponding GPS system.
2. If the departure times of some packets under PGPS system in a busy period are earlier than that of GPS, then there are also some packets with which the departure time are later than those of corresponding GPS system.

The second case implies that if there is a packet with which the departure time under PGPS system is later than the departure time of the corresponding GPS system, then the sending orders are not the same in the two systems in the busy period. According to the operation of PGPS, the difference of sending orders is only caused by some packets arrive too late to be transmitted in their order in GPS system. Thus, after these packets arrive, they may wait for the packets which should be sent later than them in GPS system to be served. Then, the additional delay caused.

Now we are clear that the only packets that have later departure time under PGPS system than under GPS system are those that arrive too late to be send in the order of corresponding GPS system. Based on this fact, we now show that:

$$t_k \le u_k + \frac{L_{\max}}{C}$$

For k = 1,2,… Let p_m be the packet with the largest index that has earlier departure time than p_k under PGPS system but has later depart time under GPS system. That is, m satisfies

$$\begin{aligned} &0 < m \le k-1 \\ &u_m > u_k \ge u_i \quad \text{for } m < i < k \end{aligned}$$

So packet pm is send before packets $p_{m+1}, \ldots, p_k$ under PGPS, but after all these packets under GPS. If no such m exists then set m = 0. For the case m = 0, it direct lead to case 1 above, and $u_k \geq t_k$. For the case m > 0, packet pm begins transmission at $t_m - L_m / C$, so from Lemma 1:

$$\min\{a_{m+1}, \ldots, a_k\} > t_m - \frac{L_m}{C}$$

That is, $p_{m+1}, \ldots, p_{k-1}$ arrive and are served under GPS system after tm-Lm/C. Thus

$$u_k \ge \frac{1}{C}(L_k + L_{k+1} + \ldots + L_{m+1}) + t_m - \frac{L_m}{C}$$

Moreover, since

$$\frac{1}{C}(L_k + L_{k-1} + \ldots + L_{m+1}) + t_m = t_k$$

we obtain the inequality

$$u_k \geq t_k - \frac{L_m}{C} \geq t_k - \frac{L_{\max}}{C}$$

which directly lead to the desired result. ■

It is worth to note that the guarantee of delay in PGPS system in Theorem 1 leads to the guarantee of per-flow throughput.

Theorem 2 *For all times τ and flows i*

$$S_i(0,\tau) - S'_i(0,\tau) \leq L_{\max}$$

where $S_i(a,b)$ and $S'_i(a,b)$ are the amount of flow i traffic served in the interval $[a,b]$, respectively.

Prove.

$$S_i(0,\tau) - L_{\max} \leq S_i(0,\tau - \frac{L_{\max}}{C}) \overset{(a)}{\leq} S'_i(0,\tau)$$

relation (a) comes from the fact that all the flow i packets transmitted before $\tau\text{-}L_{max}/C$ under GPS system will always be transmitted before τ under PGPS system, which is the direct consequence of Theorem 1. ■

Let $Q_i(\tau)$ and $Q_i(\tau)$ be the flow i backlog at time τ under GPS and PGPS system, respectively. Then it immediately follows from Theorem 2 that

Corollary 2.1 For all time τ and flow i

$$Q'_i(\tau) - Q_i(\tau) \leq L_{\max}$$

From the above results, we can see that PGPS provides quiet close approximation to GPS with the service curve never falls behind more than one packet length. This allows us to relate results for GPS to the packet-switched system in a precise manner. For more extensive analysis of PGPS, readers can refer to [1], [2], and [3].

4. Wireless packet scheduling algorithms

Recently, as various wireless technologies and systems are rapidly developed, the design of packet scheduling algorithms in such wireless environments for efficient packet transmissions has been a crucial research direction. Till now, a lot of wireless packet scheduling algorithms have been studied in many research papers. In the section, we will select four much more representative ones for illustrations in detail.

4.1 Idealized Wireless Fair Queueing (IWFQ) algorithm

The Idealized Wireless Fair Queueing (IWFQ) algorithm, proposed by Lu, Bharghavan, and Srikant [14] is one of the earliest representative packet scheduling algorithms for wireless access networks and to handle the characteristic of location-dependent burst error in wireless links. IWFQ takes an error-free WFQ service system as its reference system, where a channel predictor is included in the system to monitor the wireless link statuses of each flow and determines the links are in either "good" or "bad" states. The difference between IWFQ and WFQ is that when a picked packet is predicted in a bad link state, it will not be transmit and the packet with the next smallest virtual finish time will be picked. The process will repeat until the scheduler finds a packet with a good state.

A flow is said to be *lagging*, *leading*, or *in sync* when the queue size is smaller than, larger than, or equal to the queue size in the reference system. When a *lagging* flow recovered from a bad link state, it must have packets with smaller virtual finish times, compare to other error-free flows' packets. Thus, it will have precedence to be picked to transmit. So the compensation is guaranteed [15]. Additionally, to avoid unbounded amount of compensation starve other flows in good link state, the total lag that will be compensated among all *lagging* flows is bounded by B bits. Similarly, a flow i cannot lead more than l_i bits.

However, IWFQ does not consider the delay/jitter requirements in real-time applications. It makes no difference for different kind of applications, but in fact, non-real-time and delay-sensitive real-time applications have fundamental difference in QoS requirement, so always treat them identically may not be a reasonable solution. In addition, the choice of the parameter B reflects a conflict between the worst-case delay and throughput properties. Hence, the guarantees for throughput and delay are tightly coupled. In many scenarios, especially for real-time applications, decoupling of delay from bandwidth might be a more attractive approach [16]. Moreover, since the absolute priority is given to packets with the smallest virtual finish time, so a lagging flow may be compensated in a rate independent of its allocated service rate, violating the semantics that a larger guaranteed rate implies better QoS, which may be not desirable.

4.2 Channel-condition Independent packet Fair Queueing (CIF-Q) algorithm

The Channel-condition Independent packet Fair Queueing (CIF-Q) algorithm [17], proposed by Ng, Stoica, and Zhang. CIF-Q also uses an error free fair queueing algorithm as a reference system. In [17], Start-time Fair Queueing (SFQ) is chosen to be the core of CIF-Q. Similar to IWFQ, a flow is also classified to be *lagging*, *leading*, or *satisfied* according to the difference of the amount of service it have received to that of the corresponding reference system. The major difference between CIF-Q and IWFQ is that in CIF-Q the leading flows are allowed to continue to receive service at an average rate ar_i , where r_i is the service rate allocated to flow i and a is a configurable parameter. And instead of always choosing the packet with smallest virtual service tag like IWFQ, the compensation in CIF-Q is distributed among the *lagging* flows in proportion to their allocated service rates.

Compared with IWFQ, CIF-Q has better scheduling fairness and also has good properties of guaranteeing delay and throughput for error-free flows like IWFQ. However, the requirement of decoupling of delay from bandwidth is still not achieved by CIF-Q.

4.3 Improved Channel State Dependent Packet Scheduling (I-CSDPS) algorithm

A wireless scheduling algorithm employing a modified version of Deficit Round Robin (DRR) scheduler is called Improved Channel State Dependent Packet Scheduling (I-CSDPS), which is proposed by J. Gomez, A. T. Campbell, and H. Morikawa [18].

In DRR, each flow has its own queue, and the queues are served in a round robin fashion. Each queue maintains two parameters: Deficit Counter (*DC*) and Quantum Size (*QS*). *DC* can be regarded as the total credit (in bits or bytes) that a flow has to transmit packets. And

QS determines how much credit is given to a flow in each round. For each flow at the beginning of each round, a credit of size *QS* is added to *DC*. When the scheduler serves a queue, it transmits the first *N* packets in the queue, where *N* is the largest integer such that $\sum_{i=1}^{N} l_i \le DC$, where l_i is the size of the *i*th packet in the queue. After transmission *DC* is decreased by $\sum_{i=1}^{N} l_i$. If the scheduler serves a queue and finds that there are no packets in queue, its *DC* is reset to zero.

To allow flows to receive compensation for their lost service due to link errors, I-CSDPS adds a compensation counter (*CC*) to each flow. *CC* to keep track of the amount of lost service for each flow. If the scheduler defers transmission of a packet because of link errors, the corresponding *DC* is decreased by the *QS* of the flow and the *CC* is increased by the *QS*. At the beginning of each round, $\alpha \cdot CC$ amount of credit is added to *DC,* and *CC* is decreased by the same amount, where $0 < \alpha \le 1$.

Also, to avoid problems caused by unbounded compensation, the credit accumulated in a DC cannot exceed a certain value $DC_{\max}$. Similar to the parameter *B* in IWFQ, the choice of $DC_{\max}$ also lead to the tradeoff between delay bound and the compensation for a flow lost its service. However, this bound is very loose and is in proportion to on the number of all active flows.

4.4 Proportional Fair (PF) algorithm

In the recent years, the two most well-known packet scheduling schemes for future wireless cellular networks are the maximum carrier-to-interference ratio (Max CIR) [26] and the proportional fair (PF) [27] schemes. Max CIR tends to maximize the system's capacity by serving the connections with the best channel quality condition at the expense of fairness since those connections with bad channel quality conditions may not get served. PF tries to increase the degree of fairness among connections by selecting those with the largest relative channel quality where the relative channel quality is the ratio between the connection's current supportable data rate (which depends on its channel quality conditions) and its average throughput. However, a recent study shows that the PF scheme gives more priority to connections with high variance in their channel conditions [28]. Therefore, we pay our attention focusing on the PF scheme for illustration here.

In another point of view, in wireless communication systems, the optimal design of forward link gets more attention because of the asymmetric nature of multimedia traffic, such as video streaming, e-mail, http and Web surfing. For the efficient utilization of scarce radio resources under massive downlink traffic, opportunistic scheduling in wireless networks has recently been considered important.

The PF was originally proposed in the network scheduling context by Kelly *et al.* in [45] as an alternative for a max-min scheduler, a PF scheduling promises an attractive trade-off between the maximum average throughput and user fairness.

The standard PF scheme in packet scheduling was formally defined in [45].

Definition: *A scheduling P is 'proportional fair' if and only if, for any feasible scheduling S, it satisfies:*

$$\sum_{i \in U} \frac{R_i^{(S)} - R_i^{(P)}}{R_i^{(P)}} \le 0$$

where U is the user set and $R_i^{(S)}$ *is the average rate of user i by scheduler S.*

Also, it is known that a PF allocation P should maximize the sum of logarithmic average user rates [21], which is expressed by

$$P = \arg\max_{S} \sum_{i \in U} \log R_i^{(S)} .$$

The PF scheduling is implemented for Qualcomm's HDR system, where the number of transmission channels is one. Only one user is allocated to transmit at a time, and the PF is achieved by scheduling a user j according to

$$j = \arg\max_{i} \frac{r_i}{\overline{R}_i} ,$$

where r_i is the instantaneous transmittable data rate at the current slot of user i and $\overline{R}_i$ is the average data rate at the previous slot of user i.

Consider a model where there are N active users sharing a wireless channel with the channel condition seen by each user varying independently. Better channel conditions translate into higher data rate and vice versa. Each user continuously sends its measured channel condition back to the centralized PF scheduler which resides at the base station. If the channel measurement feedback delay is relatively small compared to the channel rate variation, the scheduler has a good enough estimate of all the users' channel condition when it schedules a packet to be transmitted to the user. Since channel condition varies independently among different users, PF exploits user diversity by selecting the user with the best condition to transmit during different time slots.

The PF algorithm was proposed after studying the unfairness exhibited when increasing the capacity of CDMA by means of differentiating between different users. Transmission of pilot symbols to the different users yields channel state information, and by allocating most resources to the users having the best channels, the total system capacity of the CDMA scheme could be increased. Such allocation of resources favors the users closest to the transmitting node, resulting in reduced fairness between the different users. The PF algorithm seeks to increase the fairness among the users at the same time as keeping some of the high system throughput characteristics.

The PF scheduling algorithm has received much attention due to its favorable trade-off between total system throughput and fairness in throughput between scheduled users [19] [20]. The PF scheduling algorithm can achieve multi-user diversity [20] [21], where the scheduler tracks the channel fluctuations of the users and only schedules users when their instantaneous channel quality is near the peak. In other words, the PF scheme is a channel-state based scheduling algorithm that relies on the concept of exploiting user diversity.

PF has extensively been studied under well-defined propagation channel conditions, such as flat fading channels with Rayleigh and/or Rician type of fading [22], or the ITU Vehicular and Pedestrian channels [24], which are typically applied in standardization work [23].

In early years, the PF scheduling is widely considered in single-carrier situations. In addition, it is pointed out in [26] that the PF scheme for a single antenna system is attractive for non-real time traffics, since it achieves substantially larger system throughput than the Round-Robin (RR) scheme. The PF scheme also provides the same level of fairness as the RR

scheme in the average sense [25]. Further descriptions of the PF algorithm can be in [29], [30], [31], [32] and [33], while a variant which offers delay constraints is described in [34].
In more recent years, as many modern broadband wireless systems with multi-carrier transmissions are rapidly developed, multi-carrier scheduling becomes a hot topic. The issue will be investigated and illustrated in detail in Section 4.

5. Case study: design of packet scheduling schemes for OFDMA-based systems

5.1 Introduction to OFDMA

Recently there has been a high demand for large volume of multimedia and other application services. Such a demand in wireless communication networks requires high transmission data rates. However, such high transmission data rates would result in frequency selective fading and Inter-Symbol Interference (ISI). As a solution to overcome these issues, Orthogonal Frequency Division Multiplexing (OFDM) had been proposed in [35].
Nowadays, the OFDM technology has widely been used in most of the multi-user wireless systems, which can be referred to research papers [36-38] for instance. When such a multiple carrier system has multi-user, it can referred to as Orthogonal Frequency Division Multiple Access (OFDMA) system. In other words, the key difference between both transmission methods is that OFDM allows only one user on the channel at any given time whereas OFDMA allows multiple accesses on the same channel. OFDMA assigns a subset of subcarriers to individual users and their transmissions are simultaneous. OFDMA functions essentially as OFDM-FDMA. Each OFDMA user transmits symbols using some subcarriers that remain orthogonal to those of other users. More than one subcarrier can be assigned to one user to support high data rate applications. Simultaneous transmissions from several users can achieve better spectral efficiency.

5.2 Token-based packet scheduling scheme for IEEE 802.16 [46]

5.2.1 Frame by frame operation scheme

Since IEEE 802.16 is a discrete-time system, time is divided into fixed-length frames, and every MS is mandatory to synchronize with the BS before entering the IEEE 802.16 network [45], our packet scheduler scheme is also a discrete-time scheme and schedule packets in a per-frame basis. Additionally, because we consider downlink traffic only, all the components and algorithms are all operated in BS.
Figure 5.1 is a simple description of the operation of our packet scheduling scheme. When a packet arrives at the BS from the upper layer, it is buffered in the BS first and the system decides whether it will be scheduled to be transmitted in the next frame. This procedure will be repeated every frame until this packet is transmitted successfully in the downlink subframe of one of the afterward frame. We assumed that a packet transmitted in the downlink subframe of a frame will receive ARQ feedback (ACK or NAK) immediately from MSs in the uplink subframe of the same frame. The result of scheduling of the next frame is broadcast via the DL-MAP which is transmitted at the beginning of the next frame.

1. **System Resource Normalization**

Since the packets of each flow may be transmitted in different Modulation and Coding Schemes (MCS), we use “slots” as a general unit of entire system to describe traffic characteristic and system resource. Suppose that the MSC used for a flow is not changed during the session’s life time.

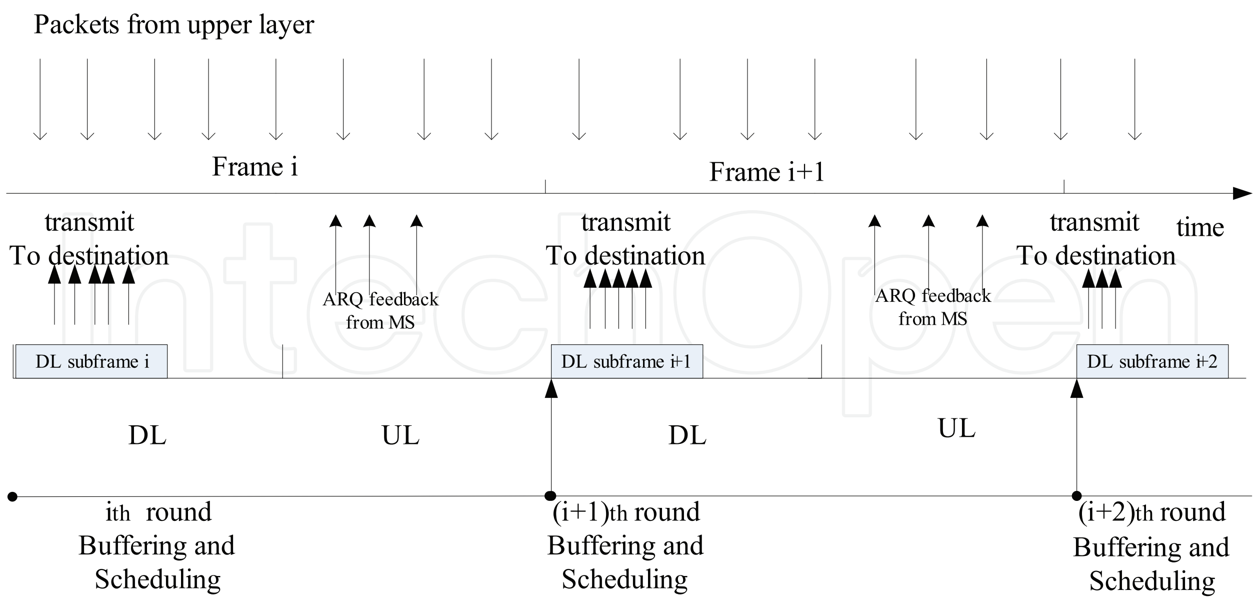

Fig. 5.1 Simple description of the operation of packet scheduling scheme

For example, we can say a leaky bucket shaper has bucket depth 10 slots and 2.251 slots/frame. Or we guaranteed a non-real-time session a minimum throughput of 5.35 slots/frame.

In this study, we assume our system has a total of C slots available for downlink traffic in a frame. We can also say that this system has a capacity of C slots/frame.

2. **System Architectures**

Figure 5.2 is our proposed packet scheduling scheme operated in the BS. It consists of several components. We describe their functions and algorithms respectively in this section.

Classifier and Traffic Profile

The classifier is responsible for classifying packets from upper layers to the appropriate service group. Two service groups are defined, that is, real-time group and non-real-time group, according to their fundamental differences of QoS requirement. There are several approaches to identify each packet's group. One suggestion is to classify each packet according to the service type of its MAC connection ID. For example, UGS, rtPS, and ertPS are belong to real-time group and nrtPS and BE are belong to non-real-time group. We also assume that each flow has a flow profile for description of its traffic characteristic. Flows of real-time group and flows of non-real-time group have different flow profiles. We introduce them respectively as follows:

Real-time group: Real-time flows are delay-sensitive traffic. A packet from real-time sessions is expected to be transmitted successfully in some delay constraint or it is regarded as meaningless and dropped. Although that, some loss rate does not degrade the application layer quality seriously and is tolerable for users. A triple $\{\delta i, \lambda i, Di\}$ is used to describe the traffic characteristic of real-time flow i. Where δi is maximum burst size (normalized to slots), λi is the minimum sustainable data rate (normalized to slots/frame), Di is the maximum tolerable packet delay (in frame). Note that when a leaky bucket policer [46] is used, δi is equivalent to the bucket depth and λi is equivalent to the average rate in the leaky-bucket policing algorithm.

Non-real-time group: Non-real-time flows are not sensitive to delay and jitter. The QoS matrix of non-real-time services is the average throughput. A parameter λ_j is used to describe the traffic characteristic of non-real-time session j. Where λ_j is the minimum reserved data rate (normalized to slots/frame).

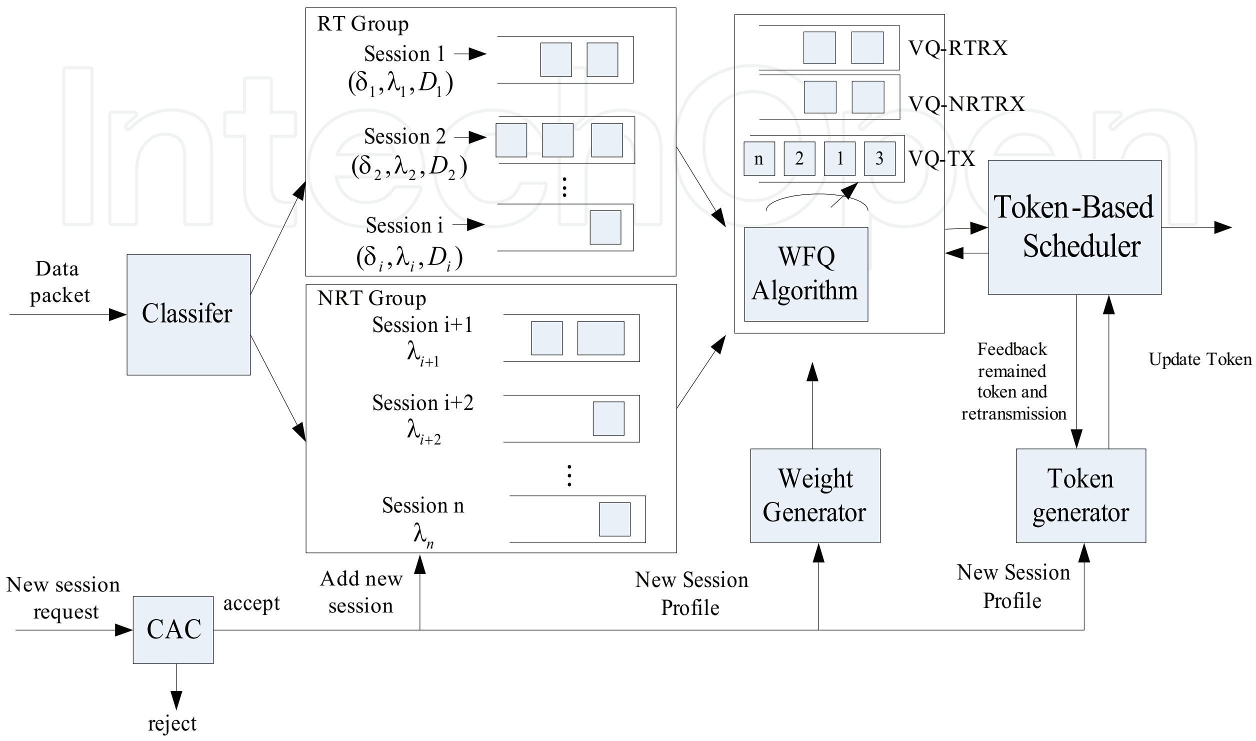

Fig. 5.2 The system architecture of our packet scheduler

Packet Scheduler and Weight Generator

In the following sections, we introduce the main part of our packet scheduling scheme. Some useful notations are as follows:

S_{RT}: The set of all real-time flows

S_{NRT}:The set of all non-real-time flows

b_i: The minimum required capacity to achieve the QoS requirement of flow *i* (normalized to slots/frame). For real-time services, the QoS marix is the tolerant delay of a packet, and for non-real-time services, the QoS matrix is the average throughput

w_i: The *weighting factor* of flow *i* in the WFQ scheduler

t_i: The current token value of flow *i*. If a packet of flow *i* is scheduled to transmit in the next frame. It must take the token value equal to the size of the packet (normalize to slot) away

T_i: The maximum token value flow *i* can keep

α_i : The protecting factor of flow *i*. A number which is larger than or equal to 1. The more α_i is, the more *protected capacity* for flow *i*.

r_i: The token incremental rate of flow *i*. At the beginning of a frame, the token value of flow *i* is updated to $max(t_i+r_i,T_i)$. r_i can be regard as the *protected capacity* for flow *i*. $r_i = b_i * \alpha_i$.

R: The sum of the *protected capacity* of real-time flows, $R = \sum_{i \in S_{RT}} r_i$

N: The sum of the *protected capacity* of non-real-time flows, $N = \sum_{i \in S_{NRT}} r_i$

N_{max}: The maximum value of the sum of *protected capacity* of non-real-time flows

C: The available slots for downlink traffic per frame. Intuitively, $C \geq R + N$

Our packet scheduling scheme has two stages. The first stage is a work conserving packet scheduler. When a packet arrives from upper layer, it first enters the first stage. The actual conditions in the lower layer such as the channel status or the allocation of slot are transparent to the first stage. It always assumes there is an error-free channel with fixed capacity C' in the lower layer. The main purpose of the first stage packet scheduler is to emulate the transmission order of a work conserving system in an ideal condition and be a reference system to our scheme. The packet order in the reference system is not certainly the actual transmission order in our packet scheduling scheme. The task of determining which packet should be scheduled to transmit in the next frame is executed by the token-based slot scheduler which is in the second stage of our scheme. The detail of the operation of the token-based slot scheduler will be illustrated in the next section.

There is no constraint to the scheduling discipline adopted in the first stage packet scheduler. But to achieve a better resource allocation and isolation among each flow, weight-based scheduling disciplines such as WFQ, VC, are suggested. In our packet scheduling scheme, we take WFQ as the reference system. When a packet arrives from the upper layer, it enters the packet scheduler in the first stage, the packet scheduler then schedules the transmission order of this packet with WFQ algorithm. The packet order scheduled by the packet scheduler is recorded in a *virtual queue*. *Virtual queue* is not really buffered the packets but store the pointers of packet which is the input of the token-based slot scheduler in the second stage.

There are three virtual queues with strict priorities. They are virtual queue for real-time retransmission (VQ-RTRX), virtual queue (VQ-NRTRX) for non-real-time retransmission, and virtual queue for first time transmission (VQ-TX) according to their priorities. The packets which have not been transmitted are recorded their pointer in the VQ. The real-time packets which have transmitted but not received successfully by the receiver, their pointers are moved from VQ to VQ-RTRX. The non-real-time packets which have transmitted but not received successfully by the receiver, their pointers are moved from VQ to NRTRVQ . The token-based packet scheduler checks the packet pointers from the virtual queue with highest priority (RTRVQ) to that with lowest priority (VQ) and determines which packets will be scheduled to transmit in the next frame. The algorithm determining which packets will be scheduled will be discussed in the next section in detail. Figure 5.3 is the queueing model of our packet schedulingscheme.

To indicate the resource sharing of the flows, each flow i associates a *weighting factor* w_i . The *weighting factor* is an important parameter as the weight in packet scheduler in the first stage and in the *debt allocation procedure* in token-based slot scheduler. We will return to discuss the procedure of weight allocation after we introduce the token-based slot scheduler and its algorithm in the next section.

Token-Based Scheduler

We use a token-based scheduler to determine which packet should be scheduled to transmit in the next frame. The fundamental operation of the token-based slot scheduler is as follows. For convenience of illustration, we define some notation as follows:

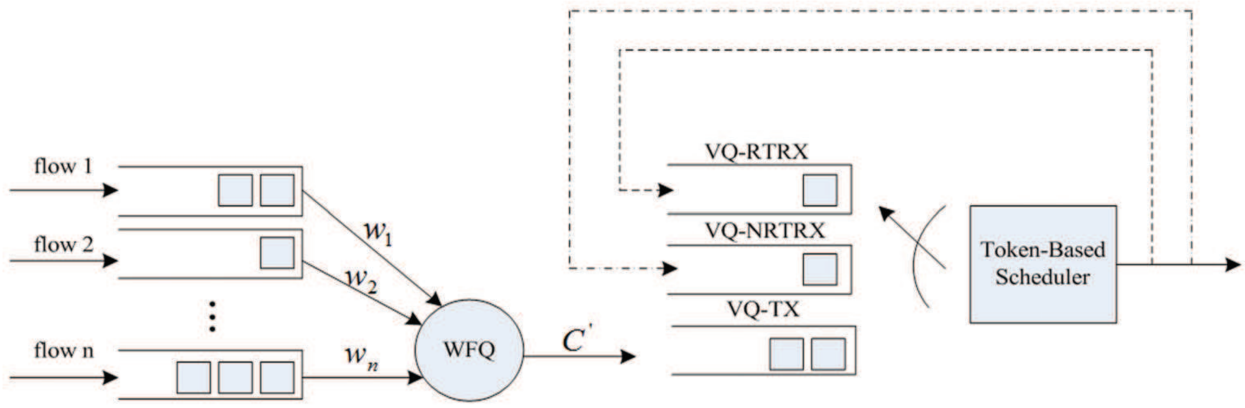

Fig. 5.3 The queueing model of our proposed packet scheduling scheme

l : The size of the packet that is checked by the token-based scheduler

l_i^j : The j_{th} packets of flow i which is scheduled in the next frame

L_i : The total length of the scheduled packets of flow i. That is, $L_i = \sum_j l_i^j$

$remained\_slots$:The remained slots available for scheduling. $remained\_slot = C - \sum_i \lceil L_i \rceil$

Assume that there are C slots available for downlink traffic each frame. Each flow i maintains a *token value* t_i. The *token value* of every session i has a fixed *token incremental rate* r_i , the unit of r_i is slots/frame. At every beginning of a frame, the *token value* of session i is increased by r_i slots. The task of updating the token value of each flow at the beginning of a frame is operated by the *token generator*. When a packet of flow i with size l (normalized to slots) is scheduled to transmit, it must take the token value equal to the amount of the size of the packet (normalized to slots) away. And the number of slots available for scheduling is also decreased by That is,
When a packet of flow i with size l is scheduled

$$t_i \leftarrow t_i\text{-}l \tag{5.1}$$

$$L_i += l \tag{5.2}$$

$$remained\_slot = C - \sum_i \lceil L_i \rceil \tag{5.3}$$

There should be an upper bound of the token value t_i , where we denote it by T_i. The setting of T_i can affect the system performance. We will discuss the issue of the effect of T_i later.
Thus, when a new frame start

$$t_i \leftarrow max(t_i+r_i,T_i)\text{, for each flow } I \tag{5.4}$$

We can regarded r_i as the *protected capacity* of flow i. The configuration of r_i can affect the system performance significantly. We introduce the detailed algorithm of token-based slot scheduler in this section. Then we will return to discuss the guideline of the setting of token incremental rate in the next section.
The basic principle of scheduling is as follows:

1. The packet which has sufficient token value to transmit it has the higher priority, or it has the lower priority
2. For the packets with the same priority, the scheduling order is according to the order in WFQ scheduler (i.e. the order of the virtual finish time in the WFQ)

The process of our token-based packet scheduler algorithm can be divided into two phases. At the beginning of scheduling, the token-based packet scheduler enters the first phase. It checks the packet pointers sequentially in each virtual queue from high priority to low priority. We call the first step *packet selection procedure*. If the checked packet has sufficient token value (that is, $t_i \geq l$) and there are suffienct slots to transmit it in the next frame (that is, $remained\_slots \geq \lceil L_i + l \rceil - \lceil L_i \rceil$). It is scheduled in the next frame. And the token value of this flow is decreased by the size of the packet.

Otherwise, the token-based slot scheduler will skip it, and to keep the packets of the same flow to be transmitted in order, other packets from the same flow which have not been checked are also skipped in the first phase. For convenience of discussion, we say that this flow is *blocked* in this phase. After all the packets are checked, the slot scheduler enters the second phase.

During the second phase, the slot scheduler continues to find other packets can be transmitted with the remained slots in the next frame. The token-based scheduler does it by checking the packets which have not been scheduled in the first phase. Again, the order of checking is the same as the first phase. If there are still sufficient slots to transmit the checked packet (that is, $remained\_slots \geq \lceil L_i + l \rceil - \lceil L_i \rceil$), the packet is scheduled in the next frame, or the packet is skipped and the flow of this packet is *blocked* which is the same as the first phase. When the checked packet is scheduled, the token value of the scheduled packet must runs out and become a negative number, it implies that the session of this packet uses more capacity than its protected capacity. The additional consumed token value exceeding the *protected capacity* is regarded as the *debt* draw from other flows. For example, if t_i=30, now flow *i* has a packet with size 50 slots and is scheduled to transmit in the next frame by the scheduler. The *debt* is 20. If t_i= -10, a packet with the same size is scheduled, the *debt* is 50.

We can represent *debt* as follows:

$$debt = -1 * \max(-l, t_i - l),\ l \geq t_i \tag{5.5}$$

In our algorithm, we prefer to give real-time sessions more opportunity to increase its token value, since if we clean the packets of real-time sessions as soon as possible, it is more likely to have more remained resource to improve the throughput of non-real-time traffic in the operation of our token-based algorithm, thus meet the QoS requirement of both. So the *debt* is allocated to the token value of all real-time flows in proportion to their weight. That is,

$$t_i = \max(T_i, t_i + \frac{w_i}{\sum_{j \in S_{RT}} w_j} * debt),\ \text{for all}\ i \in S_{RT} \tag{5.6}$$

We call the second step *debt allocation procedure*. For example, there are three flows. Flow 1 and 2 is real-time flows with *weighting factor* 0.3 and 0.2 respectively, flow 3 is non-real-time flows with *weighting factor* 0.5. And their *token value* is -10, 40, 20. Now flow 2 has a packet of size 50 slots be scheduled by the token-based slot scheduler. Since the packet size is larger than the token value of flow 2. We can calculate the *debt* is 10 and allocate it to the token value of real-time flows, that is, flow 1 and flow 2. Finally, the token value of flow 1 is $-10+\frac{10*0.3}{0.2+0.3}=-4$, the token value of flow 2 is $40-50+\frac{10*0.2}{0.2+0..3}=-6$. The token value of flow 3 is not changed. The debt allocation procedure is finished when all the packets are checked. Then in the slot the scheduler transmits the scheduled packet and receives ARQ from the receivers.

The chosen of T_i can affect system performance significantly. If the T_i is set too large, suppose flow i is in good channel status for a long time and it accumulates a large amount of token value from the token generator and the *debt* of other flows, now it incurs burst error and the channel is in bad channel for an long interval of time. Then flow i will waste a large amount of system resource to transmit error packet because it accumulates too much token value when it is in good channel. Thus is unfavorable. On the other hand, if the maximum token value is set too small. Then it is hard to differentiate the flows behave well and give it more opportunity to be scheduled. Thus is difficult to show the advantage of our algorithm. Furthermore, the traffic characteristic also should be taken into consider. Generally, we suggest that the maximum token value of non-real-time flows has better larger than that of real-time flows, because most non-real-time flow are TCP traffic, which is composed of several burst.

The flow chart of all procedures of the token-based slot scheduler is shown in Fig. 5.4. The checking procedure and the debt allocation procedure is the core of our slot scheduler. The pseudo codes of these two procedures are shown in Fig. 5.5 and Fig. 5.6, respectively.

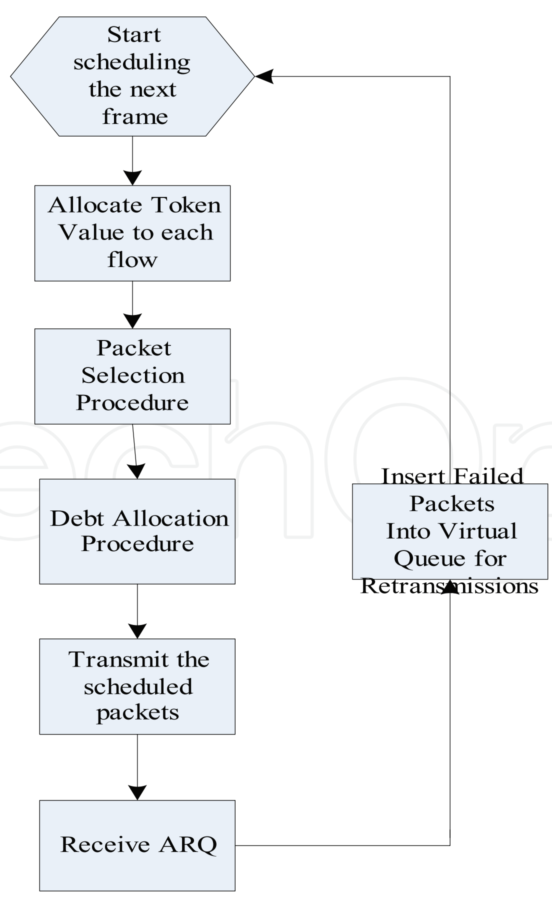

Fig. 5.4 The flow chart of the procedure of the token-based scheduler

```
packet-selection-procedure(system_capacity){
remained_slot←system_capacity;
for(each flow j){
   block_j=0;
}
for(each virtual queue, from highest priority to lowest priority)
while(packets not checked in the virtual queue){
      i← the flow the packet belong to;
      l←the size of the checked packet;
      if( remained_slot ≥ ⌈L_i + l⌉ − ⌈L_i⌉ and l ≤ t_i and block_i==0){
         schedule this packet in the next frame;
         remained_slot −= ⌈L_i + l⌉ − ⌈L_i⌉;
         t_i −= l;
      }
      else
        block_i=1; /* the other packet of flow i is also skipped in this procedure */
}
}
debt-allocation-procedure(remained_slot);
}
```

Fig. 5.5 The pseudo code of packet selection procedure

Weight Allocation and Token Generator

In this section, we discuss the functions of *token generator***,** and the relationship between . The main tasks of token generator are calculating the *token incremental **rate*** of each flow, and allocating token value to all the flows per frame. In this section, we address the issues of the chosen of *weighting factor* and *token incremental rate***.**

The different configuration of the *token incremental rate* alters the system performance. We suggest that the *token incremental rate* of flow *i* is set to its *minimum required capacity* b_i multiplies a *protecting factor* α_i. The *minimum required capacity* of flow *i* is the minimum capacity need to reserved for flow *i* to satisfy its QoS requirement. That is, the capacity to make flow *i*'s QoS acceptable in the assumption that no channel error occurs. For real-time services, the QoS matrix is the tolerable delay of a packet. A real-time flow *i* with traffic profile *{δi, λi, Di}*, the ***minimum required capacity*** is $\max(\frac{\delta_i}{D_i},\lambda_i)$. For non-real-time services, the QoS matrix is the average throughput. A non-real-time flow *j* with traffic profile *λj*, the *minimum required capacity* is *λj*. Thus, b_i is calculated as follows:

$$b_i = \begin{cases} \max(\frac{\delta_i}{D_i},\lambda_i), i \in S_{RT} \\ \lambda_i, i \in S_{NRT} \end{cases} \tag{5.7}$$

The purpose of *protecting factor* α_i is to expand the *protected capacity* of flow *i* by multiplying the *minimum required capacity* by a number larger than or equal to 1. The larger the *protecting*

```
debt_allocation_procedure(remained_slot){
  for(each flow j)
    block_i ← 0;

for(each virtual queue, from the highest priority to the lowest priority){

    while(packets not checked and not scheduled in the virtual queue){
      i ← the flow the packet belong to;
      l ← the size of the packet;
      if( remained_slot ≥ ⌈L_i + l⌉ − ⌈L_i⌉ and block_i != 1){
           schedule the packet in the next frame;
           debt ← -1 * max(t_i − l, −l);
           t_i −= l;
           remained_slot −= ⌈L_i + l⌉ − ⌈L_i⌉;
           for(all real-time flows k){
                t_k = max(t_k + (w_k * debt) / (Σ_{j∈RT} w_j), T_k);
           }
      }
      else
           block_i ← 1;  /* the other packet of flow i is also skipped in this procedure */
    }
  }
}
```

Fig. 5.6 The pseudo code of debt allocation procedure

factor, the larger the *protected capacity.* It implies that providing a flow more protection by giving it more resource than it required to compensate the loss due to wireless channel error. The tuning of *protecting factors* is also important and closely relative to system performance. If the *protecting factor* of a flow is too large, it may be unfair to other flows and also cause waste of resource, which will be harmful to overall system performance. In our scheme, we set the *protecting factors* of real-time flows to 1, and set the *protecting factors* of non-real-time flows to a number slightly larger than 1, for example, Since in our token-based scheduler, we give real-time flows more opportunity to increase their token value than that of non-real-time flows by allocating all the *debt* to real-time flows. So we compensate non-real-time flows by regulating their *protecting factors* to be larger than that of real-time flows'. Additionally, setting the *protecting factor* of real-time flows to 1 means the *protected capacity* of a real-time flow is the same as its *minimum required capacity*. It brings benefits to differentiate the flows with good channel status and the flows with bad channel status. Because when a flow suffers burst error, it will use more capacity than its minimum required, so it soon runs out of its token value, and other real-time flows in good channel status get additional token value. It makes the real-time flows in good channel status has higher priority to be transmitted, and improve the efficiency of the use of system resource.

In many real situations, we may degrade some resource sharing of non-real-time flows to make the system to accommodate more real-time flows. That is, satisfy more real-time users at the cost of some average throughput of non-real-time services. We can achieve this by bounding the sum of the *token generating rate* of non-real-time flows. When the sum of the *token generating rate* of all non-real-time flows exceeds a defined value $N_{\max}$, the *token generating ratse* of all non-real-time flows degrade proportionally to make their sum not larger than $N_{\max}$.
Thus, the sum of the token rate of all non-real-time flows N can be represented as

$$N = \min(N_{\max}, \sum_{i \in S_{NRT}} (\alpha_i * b_i)) \tag{5.8}$$

and the token rate of a non-real-time flow can be calculated as

$$r_i = N * \frac{b_i}{\sum_{i \in S_{NRT}} b_i} \tag{5.9}$$

The *weighting factor* is for indicating the resource allocation of the WFQ in the first stage of our scheme. The WFQ emulates a work-conserving system with error-free channel. The *weighting factor* of flow i is proportional to its *protected capacity* divided by its protecting factor. That is,

$$w_i = \frac{r_i / \alpha_i}{\sum_j r_j / \alpha_j} \quad \text{for all flow } i \tag{5.10}$$

5.3 PF schemes for OFDMA systems

Recently, for the higher rate data transmission, interest in wireless communications has shifted in the direction of broadband systems such as multicarrier transmission systems such like the OFDMA system. There has been a growing interest in defining radio resource allocation for a physical layer based on the OFDMA technology for 4G cellular system. While throughput–optimal scheduling can be achieved by using the multi–user diversity effect, it can generate unfairness as users with bad channel conditions have a lower probability to get a resource.
Based on the definition of the standard PF scheduling scheme [45], the theorem of the modified PF scheduling for multi-carrier transmission systems was proposed in [40]. Notice that the proof of this theorem is omitted and can be referred to [40].
Theorem: *A scheduling P is 'proportional fair' for a multicarrier transmission system, if and only if, for any feasible scheduling S, it satisfies:*

$$P = \arg\max_S \prod_{i \in U} \left(1 + \frac{\sum_{k \in C_i} r_{i,k}}{(T-1)\bar{R}_i} \right),$$

where U is the set of selected users by S, C_i *is the set of carriers allocated to user i,* $r_{i,k}$ *is the instantaneous transmittable data rate of carrier* $k \in C_i$ *at the current slot,* $\bar{R}_i$ *is the average rate of user i at the previous slot, and T is the average window size.*

With OFDMA, there are multiple transmission channels that can be used, where scheduling schemes considering the PF algorithms have widely been studied in many papers. See, for example, [39 41-42]. Papers [39] and [42] had proposed heuristic approaches by simply applying PF of the single carrier case in each subcarrier to adapt for the multi–carrier case in a suboptimal manner. Additional the QoS requirement for each user was considered in [41]. Furthermore, readers are suggested to refer to [43-44] for more complete investigation of related modified PF schemes in multi-carrier systems.

6. Summary and the discussion of open issues

Packet scheduling is one of most important radio resource management functions. It is responsible for determining which packet is to be transmitted such that the resources are fully utilized. The design of an efficient algorithm to be used for the scheduling of packet transmissions in wireless communication networks is a still a open issue for research. This Chapter has widely covered the conceptual description of many representative packet scheduling algorithms deployed in high-speed point-to-point wireline and wireless scenarios.

Well designing algorithms with low complexity offering fairness among and potentially differentiation between different data-flows is important in the evolution of communication networks. The rapidly growing demand of network nodes capable of taking into account the different QoS requirements of different flows to better utilize the available resources at the same time as some degree of fairness is maintained, makes more intelligent packet scheduling a central topic in future development of communication technologies.

7. Acknowledgement

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8. References

[1] A. Parekh, R. G. Gallager, ″A generalized processor sharing approach to flow control in integrated services networks: The single-node case,″ *IEEE/ACM Trans. on Networking,* Vol. 1, June 1993

[2] A. Parekh, R. G. Gallager, ″A generalized processor sharing approach to flow control in integrated services networks: The multiple-node case,″ *IEEE/ACM Trans. On Networking,* Vol. 2, April 1994

[3] R. Cruz, “Quality of Service Guarantees in Virtual Circuit Switched Networks,” *IEEE J. Select. Areas Commun.,* Special issue on ″Advances in the Fundamentals of Networking,″ August, 1995.

[4] A. Demers, S. Keshav, and S. Shenkar, “Analysis and simulation of a fair queueing algorithm,” *Internet. Res. Amd Exper., vol. 1, 1990*

[5] D. Ferarri, “Real-time communication in an internetwork,” *J. High Speed Networks*, vol. 1, no. 1, pp. 79-103, 1992

[6] D. Ferrari and D. Verma. "A scheme for real-time channel establishment in wide-area networks," *IEEE Journal on Selected Areas in Communications*, 8(3):368–379, April 1990.

[7] J. R. Piney, S. Sallent, "Performance Evaluation of a Normalized EDF Service Discipline," *in Proc. IEEE MELECON 2004, Dubrovnik, Croatia, May 2004*

[8] S. Chaudhry, and A. Choudhary, "Tune dependent priority scheduling for guaranteed QoS Systems," *in Proc. Sixth International Conference on Computer Communications and Networks, pp. 236-241, Sept. 1997*

[9] L. Georgiadis, R. Guerin, A. Parekh, "Optimal multiplexing on a single link: delay and buffer requirements," *IEEE Trans. On Information Theory, 43(5), pp. 1518-1535, Sep. 1997*

[10] K. Zai, Y. Zhang, Y. Viniotis, "Achieving end-to-end delay bounds by EDF scheduling without traffic shaping," *in Proc. Infocom'01*, 2001

[11] V. Sivaraman, F. M. Chiussi, Mario Gerla, "End-to-End Statistical Delay Service under GPS and EDF Scheduling: A Comparison Study," *in Proc. Infocom'01*, 2001

[12] H. Zhang, "Service disciplines for guaranteed performance service in packet-switching networks," *Proceedings of the IEEE*, vol. 83, No. 10, Oct. 1995

[13] J. Liebeherr, D. Wrege and D. Ferrari, "Exact admission control in networks with bounded delay services," IEEE/ACM Trans. Networking, vol. 4, pp. 885-901, 1996.

[14] S. Lu and V. Bharghavan, "Fair Scheduling in Wireless Packet Networks," *IEEE/ACM Trans. Neetworking*, vol. 7, no. 4, pp. 473-489, 1999.

[15] Y. Cao, and VICOR O. K. Li, "Scheduling Algorithms in Broad-Band Wireless Networks," *Proceedings of the IEEE*, Vol. 89, No. 1, Jan. 2001.

[16] F. Tsou, H. Chiou, and Z. Tsai, "WDFQ: An Efficient Traffic Scheduler with Fair Bandwidth Sharing for Wireless Multimedia Services," *IEICE TRANS. COMMUNICATIONS*, Vol. E00-A, No. 1, Jan. 2000

[17] T. S. Eugene Ng, I. Stoica, and H. Zhang, "Packet Fair Queueing Algorithms for Wireless Networks with Location-Dependent Errors," *in Proc. INFOCOM'98*, Mar. 1998, pp. 1103-1111.

[18] J. Gomez, A. T. Campbell, and H. Morikawa, "The Havana Framework for Supporting Application and Channel Dependent QOS in Wireless Networks," *in Proc. Seventh International Conference on Network Protocols*, 1999.

[19] T. E. Kolding, K. I. Pedersen, J. Wigard, F. Frederiksen, and P. E. Mogensen, "High Speed Downlink Packet Access: WCDMA Evolution," *IEEE Vehicular Technology Society News*, February 2003, pp. 4-10.

[20] P. Viswanath, D. Tse, and R. Laroia , "Opportunistic beam forming using dumb antennas," *IEEE Transactions on Information Theory*, vol. 48, no. 6, June 2002.

[21] D. Tse, "Multiuser diversity in wireless networks," *Wireless Communication Seminar*, Stanford University, April 2001.

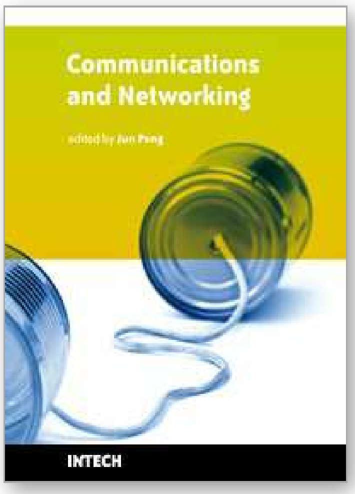
[22] J. M. Holtzman, "Asymptotic analysis of Proportional Fair algorithm," *IEEE Proc. Personal Indoor Mobile Radio Communications (PIMRC)*, September 2001, pp. 33-37.

[23] T. E. Kolding, "Link and system performance aspects of Proportional Fair scheduling in WCDMA/HSDPA," *Proceedings of 58th IEEE Vehicular Technology Conference (VTC)*, Florida USA, October 2003, pp. 1454-1458.

[24] "Guidelines for the evaluation of radio transmission technologies for IMT-2000," Recommendation ITU-R M.1225, 1997.

[25] A. Jalali, R. Padovani, and R. Pankaj, "Data throughput of CDMA-HDR a high efficiency-high data rate personal communication wireless system," *in Proc. IEEE. Veh. Technol. Conf. Spring*, Tokyo, Japan, May 2000, pp. 1854-1858.
[26] S. Borst, "User-level performance of channel-aware scheduling schemes in wireless data networks," *IEEE INFOCOM*, Mar. 2003, vol. 1, pp. 321–331.
[27] A. Jalali, R. Padovani, and R. Pankaj, "Data throughput of CDMA-HDR a high efficiency-high date rate personal communication wireless system," *in Proc. IEEE VTC*, May 2000, pp. 1854–1858.
[28] M. Kazmi and N. Wiberg, "Scheduling schemes for HSDSCH in a WCDMA mixed traffic scenario," *in Proc. IEEE Int. Symp. PIMRC*, Beijing, China, Sep. 2003, pp. 1485–1489.
[29] J. M. Holtzman, "Asymptotic Analysis of Proportional Fair Algorithm", *IEEE International Symposium on Personal, Indoor and Mobile Radio Communications*, Vol. 2, October 2001
[30] L. Erwu, and K. K. Leung, "MAC 20-5 - Proportional Fair Scheduling: Analytical Insight under Rayleigh Fading Environment", *IEEE Wireless Communications and Networking Conference*, April 2008
[31] P. Viswanath, D. N. C. Tse, and R. Laroia, "Opportunistic Beamforming using Dumb Antennas", *IEEE Transactions on Information Theory*, Vol. 48, Issue 6, June 2002
[32] P. Mueng, W. Yichuan, and W. Wenbo, "Joint an Advanced Proportionally Fair Scheduling and Rate Adaptation for Multi-services in TDD-CDMA Systems", *IEEE 59th Vehicular Technology Conference*, Vol. 3, May 2004
[33] K. Kuenyoung, K. Hoon, and H. Youngnam, "A Proportionally Fair Scheduling Algorithm with QoS and Priority in 1xEV-DO", *IEEE Symposium on Personal, Indoor and Mobile Radio Communications*, Vol. 5, September 2002
[34] O. S. Shin, and K. B. Lee, "Packet Scheduling over a Shared Wireless Link for Heterogeneous Classes of Traffic", *IEEE International Conference on Communications*, Vol. 1, June 2004
[35] J. A.C. Bingham, "Multi carrier modulation for data transmission: an idea whose time has come," *IEEE Communications Magazine*, pp. 5-14, May 1990.
[36] J. Jang and K. Lee, "Transmit power adaptation for multiuser OFDM systems," *IEEE Journal on Selected Areas in Communications*, 21(2): 171-178, Feb. 2003.
[37] Y. J. Zhang and K. B. Letaief, "Multiuser adaptive subcarrier-and bit allocation with adaptive cell selection for OFDM systems," *IEEE Transactions on Wireless Communications*, 3(4): 1566-1575, Sept. 2004.
[38] Z. Shen, J. G. Andrews, and B. L. Evans, "Adaptive resource allocation for multiuser OFDM with constrained fairness," *IEEE Transactions on Wireless Communications*, 4(6): 2726-2737, Nov. 2005.
[39] W. Anchun, X. Liang, Z. Shidong, X. Xibin, and Y. Yan, "Dynamic resource management in the fourth generation wireless systems," *in Proc. ICCT*, vol. 2, April 2003, pp. 1095–1098.
[40] H. Kim and Y. Han, "A proportional fair scheduling for multicarrier transmission systems," *IEEE Communications Letters*, vol. 9, no. 3, pp. 210–212, March 2005.
[41] Y. Lu, C. Wang, C. Yin ,and G. Yue, "Downlink scheduling and radio resource allocation in adaptive OFDMA wireless communication system for user-individual QoS," *International Journal of Electrical, Computer, and Systems Engineering* 2009

[42] N. Ruangchaijatupon and Y. Ji, “Simple proportional fairness scheduling for OFDMA frame-based wireless system,” *IEEE WCNC* 2008
[43] M. Kaneko, P. Popovski, and J. Dahl, “Proportional fairness in multi–carrier system with multi–slot frames: upper bound and user multiplexing algorithms,” *IEEE Transactions Wireless Communications*, vol. 7, no. 1, January 2008
[44] N. Ruangchaijatupon and Y. Ji, “Proportional fairness with minimum rate guarantee scheduling in a multiuser OFDMA wireless network,” *ACM IWCMC*, Leipzig, Germany, 2009
[45] F. P. Kelly, A. K. Maulloo, and D.K.H. Tan., “Rate control in communication networks: shadow prices, proportional fairness and stability,” *J. of the Operational Research Society*, vol.49, pp. 237-252, April 1998.
[46] T.-Y. Tsai and Z. Tsai, “Design of a packet scheduling scheme for downlink channel in IEEE 802.16 BWA systems,” *IEEE WCNC* 2008.
[47] IEEE 802.16e-2005, “IEEE Standard for Local and Metropolitan Area Networks – Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems – Amendent for Phisical and Medium Access Control Layers for Combined Fixed and Mobile Operation in Licensed Bands,” Dec. 7, 2005.



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This book "Communications and Networking" focuses on the issues at the lowest two layers of communications and networking and provides recent research results on some of these issues. In particular, it first introduces recent research results on many important issues at the physical layer and data link layer of communications and networking and then briefly shows some results on some other important topics such as security and the application of wireless networks. In summary, this book covers a wide range of interesting topics of communications and networking. The introductions, data, and references in this book will help the readers know more abut this topic and help them explore this exciting and fast-evolving field.