中文摘要

隨著資訊化社會的出現，未來異質性接取網路將結合個人區域網路、無線區域網路，以及蜂巢式行動網路，並且提供行動多媒體服務，依據使用者的位置、移動性、傳輸需求，和服務品質等級。因為無線頻帶具稀少性，因此需要一套可有效管理系統資源的技術，包括：傳輸功率、接取控制等。在本計畫中，我們將致力於發展新型無線資源管理技術，應用於結合 WCDMA 和 WLAN 之 B3G 異質多接取網路。在第二年度計畫中，我們著力於發展 WCDMA 系統之排程演算法、封包接取管理機制，以及 WLAN 網路之輪詢機制。

首先，我們提出一效用函數式（utility-based）排程演算法。此演算法應用於多媒體 CDMA 網路中，考慮了三項因素：無線資源使用效率、服務品質需求達成率，以及加權式公平性。此外，此演算法也可支援特定通道與分享式通道傳輸模式。利用求取系統整體之效用函數之最大化，此排程演算法可安排系統資源之使用權，並且確保加權式公平性和服務品質的服務需求。同時，此效用函數具備了 polymatroid 數學結構，所以可利用推導連線之資源配置向量來計算出最佳排程解。

接著，我們利用『乏晰 Q-learning』技術，設計了一新型封包接取管理機制，可應用於多細胞 WCDMA 系統，稱為 FQ-SDAM。FQ-SDAM 包含了一乏晰 Q-learning 式剩餘資源預估器（FQ-RCE），以及一封包傳輸排程器（DRS）。FQ-RCE 可準確估計系統剩餘資源，而 DRS 採用改良式指數排程原則（modified exponential rule），考慮了用戶位置的傳輸因素，可有效的為非及時性用戶配置傳輸資源。模擬結果顯示，因為 FQ-SDAM 具備了『環境感知狀態』的功能，所以此機制可確實有效降低多細胞環境下的封包錯誤率。

同時，我們也針對 802.11e HCF 傳輸模式提出一新型輪詢機制，稱為 ODP 輪詢機制，藉以提供整合式語音/數據服務。在 HCF 的通道接取上，ODP 輪詢機制結合了輪詢機制和競爭機制。當有一語音用戶處於通話狀態時，ODP 輪詢機制會將其列入輪詢表單中；當該用戶進入通話暫停狀態時，就會被移出該表單，進入睡眠狀態，進以節省功率消耗。同時，當該用戶由暫停狀態回復一般正常通話狀態時，可採用競爭機制來加入輪詢表單的行列。也就是說，ODP 輪詢機制充分利用了雙向語音通話的特性，可以節省具 ON/OFF 特性之語音服務的功率消耗。模擬結果顯示，ODP 輪詢機制可減少語音用戶的功率消耗，並且增加數據用戶的資料傳輸量。

Abstract

In the coming years, the future heterogeneous access network consists of PAN, WLAN, and cellular network and will provide multimedia services according to users' location, mobility, rate requirements and service QoS requirements. Since the wireless spectrum is scarce, it is necessary to manage efficiently the system resource, including: spectrum, transmission power, access control and etc. Therefore, in this project, we are motivated to propose novel radio resource management techniques for the B3G heterogeneous networks, including: a packet scheduling algorithm and a data access mechanism for WCDMA system and a novel polling algorithm for WLAN.

Firstly, we propose a utility-based scheduling algorithm. Three factors: radio resource efficiency, QoS requirement achievement, and weighted fairness for multimedia CDMA networks are considered as a key design issue for the proposed scheduling algorithm. Also, both dedicated and shared channels serve different classes of services are considered. The system resource is scheduled via maximizing the overall system utility while the weighted fairness and the QoS requirements are kept. The optimal scheduling solution is derived as a resource assignment vector for active connections by solving the optimization of overall utility function via a polymatroid structure.

Then, we design a novel data access manager applies the fuzzy Q-learning technique (FQ-SDAM) for multi-cell WCDMA systems, which contains a fuzzy Q-learning-based residual capacity estimator (FQ-RCE) and a data rate scheduler (DRS). The FQ-RCE accurately estimates the situation-dependent residual system capacity and the DRS effectively allocates the resource for non-real-time terminals by adopting a modified exponential rule which takes the location-dependent link capacity into consideration. Simulation results show that, with the capability of situation-awareness, the proposed FQ-SDAM indeed can reduce the packet error probability in the multi-cell environment.

Also, we e propose an innovative polling scheme, named on-demand polling (ODP) scheme, over 802.11e HCF for integrated voice and data services. The ODP scheme combines the polling-based and the contention-based mechanisms over HCF. In the ODP scheme, a voice station is in the polling list when it is in the active mode. During the idle mode, it is configured to operate in the sleep mode for saving power. Also, it adopts the contention-based mechanism to join the list again when returning from the idle mode. The ODP scheme exploits the nature of the two-way voice communication; therefore, it is power efficient for ON/OFF voice services. Simulation results show that the ODP scheme can reduce the power consumption for voice stations and enhance the throughput for data stations.

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Chapter 1

Introduction

The future trend of wireless communication system is to provide personal, broadband, and high mobility wireless multimedia services. The mobile users' service requirements and terminal capabilities diverse such that it is hard to fulfill all the goals within a single communication system. Here comes the "heterogeneous access network". It is an integrated system that consists of PAN, WLAN, and cellular network, and can provide seamless service transmission according to users' location, mobility, rate requirements and service QoS requirements. Since the wireless spectrum is scarce, it is necessary to manage efficiently the system resource, including: spectrum, transmission power, access control and etc. In the 3G WCDMA system, within a single core network, there is a plan to support multiple radio access technologies, including: FDD mode, TDD mode and even HiperLAN/2. In this project, the heterogeneous access network that integrates WCDMA FDD/TDD and WCDMA /WLAN are considered.

To provides wireless multimedia services with heterogeneous service types, rate requirements, and QoS requirements, and also effectively predict the traffic varying of the integrated services, a *radio resource allocation algorithm* is studied to manage the system resource while the QoS of all the connections can be fulfilled and the spectrum efficiency is optimized. In the first year, we proposed two schemes for radio resource allocation in the WCDMA systems: radio resource index (RRI) scheme [1] and Q-learning-based multirate controller for multirate transmission control (Q-MRTC) scheme [2]. RRI is a unified resource allocation metric. From computer simulation results, it can be found that RRI is a flexible and simple mapping from traffic parameters and QoS requirements for any number types of services. Moreover, the achievable resource utilization efficiency is about 80%. Also, several bounds in our lemmas have tighter ones in some specific conditions. On the other hand, the Q-MRCT scheme successfully employs the Q-learning algorithm for the multi-rate transmission control. Also, the feature extraction method is applied to efficiently map the original state space into the resultant interference profile. The computer simulation results show that, compared with the interference-based scheme, Q-MRTC can improve the throughput of the WCDMA system by an amount of 87% under the constraint of the QoS requirement. In addition, the Q-MRTC provides better users' satisfaction by an amount of 50%.

In the second year, we extend our previous works on the call-level radio resource allocation and consider the problems on the burst-level control. Therefore, we propose a novel utility-based scheduling algorithm and a situation-aware data access control scheme for multimedia WCDMA systems. The proposed utility-based scheduling algorithm assuming three design aspects of radio resource efficiency, QoS requirement achievement, and weighted fairness for multimedia CDMA networks. Both dedicated and shared channels serve different classes of services are considered. The utility function of each connection is the radio resource function weighted by both the QoS requirement achievement function and weighted fairness function. The system resource is scheduled via maximizing the overall system utility while the weighted fairness and the QoS requirements are kept. The optimal scheduling solution is derived as a resource assignment vector for active connections by solving the optimization of overall utility function via a polymatroid structure. On the other hand, the novel situation-aware data access manager applies the fuzzy Q-learning technique (FQ-SDAM) for multi-cell WCDMA systems. The FQ-SDAM contains a fuzzy Q-learning-based residual capacity estimator (FQ-RCE) and a data rate scheduler (DRS). The FQ-RCE accurately estimates the situation-dependent residual system capacity and the DRS effectively allocates the resource for non-real-time terminals by adopting a modified exponential rule which takes the location-dependent link capacity into consideration.

As to the WLAN system, the weighted fairness for differentiated services is studied in the first year of the project [3] [4]. An analytical method is proposed to obtain parameters required to achieve weighted fairness for services operating under the enhanced distributed coordinator function (EDCF) mode. In the queuing analysis, a discrete-time Markov-chain was adopted to model the behavior of backoff counters for the two classes and the steady-state probabilities were derived. The analytic model can interpret the relationship between access probability and contention window. The accuracy of the analytical solution is verified by simulation for different number of active stations. It can be concluded that, for different combination of high- and low-class STAs, the weighted fairness is easily achieved by employing the proposed method. Based on the research results, we further consider a MAC protocol with the provisioning of integrated voice and data services. In the second year of the project, we propose a power-efficient MAC protocol, named on-demand polling (ODP) scheme, over 802.11e HCF.

The ODP scheme combines the polling-based and the contention-based mechanisms over HCF. In the ODP scheme, a voice station is in the polling list when it is in the active mode. During the idle mode, it is configured to operate in the sleep mode for saving power. Also, it adopts the contention-based mechanism to join the list again when returning from the idle mode. Simulation results show that the ODP scheme can reduce the power consumption for voice stations and enhance the throughput for data stations.

The rest of this report is organized as follows. In chapter 2, a utility-based scheduling algorithm with differentiated QoS provisioning for multimedia CDMA networks is described. In chapter 3, the detail of the FQ-SDAM is given. The ODP scheme is presented in chapter 4. Finally, a concluding remarks and future works are given in chapter 5.

Bibliography

[1] S. Shen and C. J. Chang, "The Radio Resource Index for Real-time Service in WCDMA Cellular Systems," IEEE VTC'2002-Fall, Vancouver, Canada, vol. 1, pp. 299-303, Sep., 2002.

[2] S. Chen, C. J. Chang, F. C. Ren, "A Q-learning-based multi-rate transmission control scheme for RRM in multimedia WCDMA systems," *IEEE Trans. Veh. Technol.*, vol. 53, no. 1, pp. 38-48, Jan. 2004.

[3] C. Y. Shih, R. G. Cheng, C. J. Chang, and Y. S. Chen, "Achieving Weighed Fairness for Wireless Multimedia Services," Proc. of VTC2003 - Fall, Orlando, FL, Session 7E-5.

[4] R. G. Cheng, C. J. Chang, C. Y. Shih, and Y. S. Chen, "A New Scheme to Achieve Weighted Fairness for WLAN Supporting Multimedia Services", submitted to *IEEE Transactions on Wireless Communications.*

Chapter 2

A Utility-based Scheduling Algorithm with Differentiated QoS Provisioning for Multimedia CDMA Cellular Networks

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In this chapter, we propose a utility-based scheduling algorithm assuming three design aspects of radio resource efficiency, QoS requirement achievement, and weighted fairness for multimedia CDMA networks. The utility function of each connection is the radio resource function weighted by both the QoS requirement achievement function and weighted fairness function. Per class QoS requirements are defined by both packet level, such as BER, and call level, such as, delay bound, packet dropping ratio, and minimum transmission rate. The system resource is scheduled via maximizing the overall system utility while the weighted fairness and the QoS requirements are kept. The optimal scheduling solution is derived as a resource assignment vector for active connections by solving the optimization of overall utility function via a polymatroid structure.

2.1 INTRODUCTION

In future wireless networks, heterogeneous and customized services with diverse of QoS requirements and traffic characteristics are expected to be provided via a number of air interfaces. Multimedia applications are commonly accepted as the enabling services, these diversity of applications can be categorized as four classes [1]. To meet the various traffic characteristics and QoS requirements of these potential applications, a sophisticated scheduling algorithm plays an important role in the determination of system resource allocation, while retaining a pre-defined fairness among them.

Many scheduling algorithms have been widely studied for wireline networks [2]-[3]. In the wireless communication networks, the radio channel have quite different characteristics from those in wireline networks. The transmission error probability is of several order greater than that in wireline links, and the available maximum transmission rate to each connection is location-dependent and time-varying due to link loss, shadowing, and multi-path fading. The QoS requirements and the weighted fairness among all connections should be modified. In the downlink, the total transmission power from base station is usually limited by a maximum amount so that the efficiency of power allocation among all connections is an important issue.

The literature studied the resource scheduling and allocation among connections with consideration of physical layer processing, power control range, and link conditions [4]-[5]. Bhargharvan, Lu, and Nandagopal [6] proposed a framework to achieve long-term fairness in wireless network. Varsou and Poor [7] proposed another class of scheduling algorithm from EDF concept in wireless environment. This class of scheme considers delay bound as its

QoS requirement. In [8], a throughput-optimal scheduling algorithm for delay bounded system was proposed and proved. Shakkottai and Stolyar [9] considered both link quality and QoS requirements as the criteria and derived the exponential form of scheduling function via fluid Markovian techniques.

In this chapter, we propose a utility-based scheduling algorithm considering three generalized design factors: efficiency of radio resource allocation, achievement of QoS requirement, and weighted fairness. The efficiency of the radio resource allocation reflects the link quality and also indicates the achievable maximum rate with considering the link adaptation scheme. Per class QoS requirements are considered in both packet level, such as bit-error-rate (BER), and call level, such as delay bound, packet dropping ratio, and minimum transmission rate. The weighted fairness is defined for non-real-time connections in terms of both traffic characteristics and location factor. We consider a multimedia CDMA cellular system with various service classes where dedicated channels are used for real-time class connections and shared channels are used for non-real-time class connections. We formulate a general mathematical model for the utility-based scheduling algorithms. In this model, the weighted fairness is defined in the long-term and location-dependent sense. In order to achieve the long term fairness among connections in the same service class, a set of target fairness weighting factors for each connection is determined in terms of the source characteristics. The target weighting factor is further modified according to the location-dependent factor so that the radio resource in terms of power is fairly allocated and the resulting rate is strongly dependent on the location. Since dedicated and shared channels are involved in the resource scheduling, a priority bias for each service class scales the weighting factor so that the connections using dedicated channels may have relative priority over those using shared channels. We also define a QoS requirement achievement weighting function to guarantee the QoS of each connection in a short term sense.

The utility function of each connection is the radio resource function weighted by both the QoS requirement achievement function and weighted fairness function. The system resource is scheduled in terms of the resource assignment vector via maximizing the overall system utility so that the system throughput is maximized within the QoS guaranteed region. We prove that the mathematical structure of the resource function is a polymatroid [14]. This radio resource assignment vector is finally obtained by the direct result of a polymatroid, and the scheduled transmission rate is therefore determined. We investigate the performance of the proposed algorithm by comparing with *Exponential Rule* [9] for systems using both dedicated and shared channel for different class of services. The results show that the utility-based scheduling scheme is efficient and effective for multimedia CDMA networks.

2.2 System Model and general problem formulation

For the downlink transmission in the multimedia CDMA cellular system with chip rate W, assume there are N connections with different values of QoS requirements, which are categorized into J service classes. For a connection with service class j, four QoS requirements are defined by both packet level one, such as BER_j^*, and call level ones, such as delay bound D_j^*, packet dropping ratio $P_{D,j}^*$, and minimum transmission rate $R_{m,j}^*$. Without loss of generality, we denote $\vec{\gamma}_i^* = (BER_j^*, D_j^*, P_{D,j}^*, R_{m,j}^*)$ as the requirement vector for connection i in service class j.

In the downlink transmission, real-time connections transmit on the dedicated channels and non-real-time connections transmit on the shared channels. For every active connection using either dedicated or shared channels, a fixed number of code channels with their corresponding spreading factors are given in the connection setup phase. Therefore, there exists an available peak transmission rate $R^*_{max,i}$ related to connection i. For real-time connections, if channel is in bad condition, the scheduler can drop the packet while the long-term QoS requirements are satisfied. The connections in both dedicated channels and shared channels are scheduled according channel condition, radio resource efficiency, fairness and QoS requirement achievement. The radio resouce efficiency of a connection i is denoted by $\mathcal{R}_i(t)$ which indicates the maximum achievable transmission rate for connection i corresponding to their channel and physical layer conditions and requirements when total downlink power P^*_{max} is allocated and modulation scheme κ_i is adopted. The link condition $\zeta_i(t)$ and interference $\mathcal{I}_i(t)$ information are assumed to be available at base station. Adaptive and coding scheme will be selected according to the reported $\zeta_i(t)$ and $\mathcal{I}_i(t)$. The weighted fairness for connection i is measured by the *service measurement function* $\bar{w}_i(t)$ indicating the amount of service received by connection i. And the QoS achievement of connection i is measured via the head-of-line delay $D_i(t)$ and the the queue length $Q_i(t)$ at time t.

The scheduling algorithm is extended from purely time-division fashion to jointly deal with the conventional resource allocation problem. Since the maximum transmission power of base station is P^*_{max}, the scheduler allocates the power ratio to each connections, denoted by the resource assignment vector $\vec{c} = (c_1,\ c_2,\ c_3,\ \ldots, c_{N_u})$ with constraint $\Psi_1 = \{\vec{c}(t) : \sum\limits_n c_n(t) = 1\}$. The scheduler decides the $\vec{c}$ for all connections and the actual transmission rate $r_i(t)$ of connection i will be determined via downlink power control. Note that two actual factors, minimum spreading factor and reamining queue size, should be involved in the determination of $\vec{r}(t)$ and $\vec{c}(t)$. No further utility can be gained if the assigned rate exceeds the supported rate or the necessary one to transmit the remaining packets. Denote by $\hat{r}_i(t) = Q_i/T_{frame}$ the virtual data rate required for transmission of all remaining packets in one frame time. Then the vector $\vec{c}$ has one further constraint expressed by $\Psi_2 = \{\vec{c}(t) : c_i(t) \leq min\{\frac{\hat{r}_i(t)}{\mathcal{R}_i(t)}, \frac{R^*_{max}}{\mathcal{R}_i(t)}\}, \forall i\}$.

2.3 THE UTILITY-BASED SCHEDULING ALGORITHM

The system resource is scheduled via maximizing the overall system utility so that the system throughput is achieved as high as possible within the QoS guaranteed region. Then the scheduling algorithm is

$$\vec{c}^*(t) = arg \max_{\vec{c}(t)} \left\{ \sum_i^N U_i(t) \right\},\ \text{subject to} \Psi_1 \bigcap \Psi_2 \tag{2.1}$$

where $\mathcal{U}_i(t)$ is the *individual utility function* for connection i at time t. $\mathcal{U}_i(t)$ is defined by the radio resource function weighted by the QoS requirement achievement and the weighted fairness if a fraction of total radio resource of transmission power $c_i(t)$ is assigned. Therefore, the individual utility function of connection i at time t $\mathcal{U}_i(t)$ is modeled by

$$\mathcal{U}_i\,(t) = c_i(t) \cdot \mathcal{R}_i\,(t) \cdot \mathcal{A}_i(Q_i(t), D_i(t)) \cdot \mathcal{F}_i\,(w_i, \bar{w}_i(t))\,. \tag{2.2}$$

In the following, we will discuss the radio resource function $\mathcal{R}_i(t)$, the QoS requirement achievement $\mathcal{A}_i(t)$, and the weighted fairness function $\mathcal{F}_i(t)$.

2.3.1 Radio Resource Function

Assume that the interference $\mathcal{I}_i(t)$ and the link gain condition $\zeta_i(t)$ is perfectly known. With a specific modulation scheme and the corresponding E_b/N_0 to satisfy the BER_i^* requirement, the following inequality should hold

$$\frac{W}{R_{s,i}(t)} \cdot \frac{c_i(t)P_{max}\zeta_i(t)}{(1-\alpha)P_{max}\zeta_i(t) + \sum_b P_{max}\zeta_{i,b}(t) + N_0 W} \geq \frac{E_b}{N_o}, \tag{2.3}$$

where $R_{s,i}(t)$ is the symbol rate function for connection i, α is the othorgonality factor for downlink, b is the index referring to the adjacent base stations, $\zeta_{i,b}(t)$ is the link gain from base station b to conneciton i, and the denominator can be replaced by $\mathcal{I}_i(t)$. Denote M_{κ_i} as the order of the adaptive modulation scheme. The upper bound of BER for M_{κ_i}-QAM in fading channel can be expressed by $BER \leq 0.2 \int_\gamma exp\left\{\frac{-1.5\gamma}{M_{\kappa_i}-1}\right\} f_\gamma(\gamma)d\gamma$, where γ is the instantaneous $(E_b/N_0)_i$ received by connection i, and $f_\gamma(\gamma)$ is the pdf of γ. Since the channel state is assumed to be known and to remain constant during a frame time, the upper bound is simply $BER \leq 0.2exp\left\{\frac{-1.5(E_b/N_0)}{M_{\kappa_i}-1}\right\}$ [11]. For given $BER_i^* = BER^*(j)$ requirement for class j, we can set the smallest E_b/N_0 to be

$$(E_b/N_0)_{\kappa_i} = \frac{-(M_{\kappa_i}-1)\cdot ln\{5BER_i^*\}}{1.5}. \tag{2.4}$$

By replacing E_b/N_0 with (2.4) and rearranging (2.3), we can define the *maximum symbol rate* with given $\zeta_i(t)$ and modulation scheme κ_i by $R_{s,i}\left(M_{\kappa_i}|\zeta_i(t)\right) \leq \frac{W}{(E_b/N_0)_{\kappa_i}} \cdot \frac{c_i(t)P_{max}\zeta_i(t)}{\mathcal{I}_i(t)+N_0W}$. However, for a given spreading factor SF_i for the allocated code channel, the maximal symbol rate is limited by $\frac{W}{SF_i}$, then the symbol rate function is obtained by

$$R_{s,i}(t) = \min\left\{R_{s,i}\left(M_{\kappa_i}|\zeta_i(t)\right), \frac{W}{SF_i}\right\}. \tag{2.5}$$

According to (2.5), the most efficient modulation scheme index κ_i is selected with given $\zeta_i(t)$ and $\mathcal{I}_i(t)$ if the following iequality holds,

$$M_{(\kappa_i-1)} \leq \frac{SF_i \cdot c_i \cdot (t) P_{max} \cdot \zeta_i(t)}{(\mathcal{I}_i(t) + N_0 W) \cdot \left(\frac{ln5 - lnBER_j^*}{1.5}\right)} \leq M_{\kappa_i}. \tag{2.6}$$

Note that $R_{s,i}(t)$ is increasing function with respect to $c_i(t)$. From the definition in section 2.2, the *radio resource function* $\mathcal{R}_i(t)$ is the maximum available transmission rate for the connection i and the information bit of one symbol is $log_2 M_{\kappa_i}$. Therefore, $\mathcal{R}_i(t)$ can be obtained by

$$\mathcal{R}_i(t) = \frac{1.5W \cdot log_2 M_{\kappa_i}}{(M_{\kappa_i} - 1)\left[ln\frac{1}{BER_j^*} - ln5\right]} \cdot \frac{P_{max}\zeta_i(t)}{\mathcal{I}_i(t) + N_0 W}. \tag{2.7}$$

If the ratio $c_i(t)$ of P_{max} is assigned to connection i, then the transmission rate is therefore the product $c_i(t) \cdot \mathcal{R}_i(t)$.

2.3.2 The Weighted Fairness Function

The *weighted fairness function*, $\mathcal{F}_i(t)$, indicates the unfairness of resource sharing among all connections. The unfairness is defined by the difference between the targeting amount of resource and the received resource. Therefore, the $\mathcal{F}_i(t)$ is defined as follows,

$$\mathcal{F}_i(t) = \beta_i \cdot (w_i - \bar{w}_i) \cdot u(w_i - \bar{w}_i), \tag{2.8}$$

where w_i is target weighting factor, β_i is the priority bias for differentiation among services classes, $\bar{w}_i(t)$ as the moving-average of transmission rate received by the connection i, and $u(x) = 1$ as $x > 0$ and $u(x) = 0$ otherwise.

To the target weighting factor, w_i, the fairness among all connections should be associated with the characteristics of traffic source. For real-time connections, dedicated channels are used and the scheduling of radio resource is QoS-requirement driven. The fairness among all real-time connections is ignored and the fairness weighting factor for each connection is therefore set to be 1. For non-real-time connections, no strict delay requirement is set and the fairness is the key design criteria for them to keep the buffer size of each connection similar. The target weighting factor is intuitively proportional to the source rate of each connection. In the CDMA downlink transmission, the total transmission power is the transmission resource to be shared among all connections, and the fairness weighting factors of all connections will base on the radio resource in terms of power. From this viewpoint, the long-term fairness can be defined in the expression of the fairness weighting factors by

$$\frac{\mathbf{E}\left[c_i(t) \cdot P_{max}\right]}{\mathbf{E}\left[c_k(t) \cdot P_{max}\right]} = \frac{w_i}{w_k}, \tag{2.9}$$

for $\forall i, k$ in the same service class. Since the allowable transmission rate is inversely proportional to link loss and interference, this definition of fairness and its resulting weighting factor may have scheduler starve connections in bad link conditions frequently. Therefore, for a measured average link state $\bar{\zeta}_i$ and a average interference level $\bar{\mathcal{I}}_i$, the fairness weighting factor w_i for connection n is modified by

$$w_i = max\{\frac{w_0 \bar{\zeta}_n}{\bar{\mathcal{I}}_i \cdot (\frac{E_b}{N_0})} \cdot w_i^{'}, R_{m,i}^{*}\}, \tag{2.10}$$

where $w_i^{'}$ is the targeting weighting factor considering the source characteristics only, and $w_0 = \frac{P_{max}}{\sum_i w_i^{'}}$. This weighting factor will make connections with transmission rate above the minimum rate share the resource according to their source characteristics and location-aware factor. The $w_i^{'}$ is given by the effective bandwidth with *hurst parameter* H,

$$r_i^*(\theta) = \lim_{t\to\infty} \frac{1}{\theta t^{2(1-H)}} \log \mathbf{E}\left[\theta \frac{\Omega_i(t)}{t^{(2H-1))}}\right], \tag{2.11}$$

where θ^* is the solution of $\Lambda^{'}(\theta) = \frac{\tilde{R}_{max} - (1-H)\mathbf{E}[\sum_i \Omega_i(t)]}{H}$, $\Lambda^{'}(\theta)$ is the aggregation of the $\Omega_i(t)$ for all i, and $\tilde{R}_{max}$ is the maximum capacity.

The β_i is the priority bias so that the resource separation is with relative priority among real-time and non-real time connections. To ensure that a real-time connection can receive enough resource to fulfill its packet dropping ratio, the only condition that a real-time connection will release the resource is that the radio resource efficiency with respect to current channel state is bad enough. Consider a real-time connection i. For $\zeta_i(t) \geq \zeta_{sus,i}^{th}$, real-time connections will have absolute priority over NRT connecitons; for $\zeta_i(t) < \zeta_{sus,i}^{th}$, real-time connections may release radio resource to other connections, where $\zeta_{sus,i}^{th}$ referring to the link suspension threshold for real-time class and is given by $\mathbf{P}\left\{\zeta_i(t) \leq \zeta_{sus,i}^{th}\right\} \leq P_{D,i}^*$. According to the above criteria, we have the relative priority bias between real-time connection i and any other non-real time connection k as

$$\frac{\beta_i}{\beta_k} = \frac{\mathbf{E}[\zeta_k(t)]}{\zeta_{sus,i}^{th}} \cdot \frac{\gamma_i^*}{\gamma_k^*}. \tag{2.12}$$

2.3.3 The QoS Requirement Achievement Function

As we discuss in section II, the QoS requirements are defined by both packet level and call level requirements. The resource function $\mathcal{R}_i(t)$ mainly indicates the maximum achievable rate constrained by the BER requirement. For the remaining call requirements, the *requirement achievement weighting function*

$\mathcal{A}_i(t)$ will weight the utility of the resource function according to the difference of the performance measures and their corresponding call level requirements.

For a real-time connection i, a hard delay bound D_i^* is imposed to each packets. Since QoS over wireless interface can be provided in a soft fashion, a packet dropping ratio $R_{D,i}^*$ is defined and can be modelled by $\mathbf{P}\left\{D_i(t) > D_i^*\right\} \leq P_{D,i}^*$, where $D_i(t)$ is the waiting delay for head-of-line packet at time t. For a non-real time connection i, a different notion of QoS requirement is that a minimum transmission rate must be guaranteed by $\mathbf{E}\left[\Omega_i(t)\right] \geq R_{m,i}^*$. The $R_{m,i}^*$ is set for each types of service and might be ignored for several non-real time connections. For example, the $R_{m,i}^*$ requirement for best-effort connections is set to be 0, but for several interactive connections, the minimum rate requirement is greater than zero.

From [13], the proposed *Modified Largest Weighted Delay First* (M-LWDF) algorithm suggests that an exponential rule [9] is the form with throughput optimal for the above call level QoS requirement constraints. The *requirement achievement weighting function* $\mathcal{A}_i(t)$ jointly considers the above two soft QoS requirements and follows the exponential form by

$$\mathcal{A}_i(t) = \begin{cases} exp\left\{\phi_{D,i}\frac{D_i(t)}{\alpha_D+[\bar{D}(t)]^\eta}\right\}, & i \in \{\text{RT class}\} \\ exp\left\{\phi_{R,i}\frac{\frac{Q_i(t)}{R_{m,i}^*}}{\alpha_R+[\bar{Q}(t)]^\eta}\right\}, & i \in \{\text{NRT class}\}, \\ 1, & R_{m,i}^* = 0 \end{cases} \tag{2.13}$$

where $\phi_{D,i}$ and $\phi_{R,i}$ are the positive constants for connection i, $\eta \in (0,1)$, α_D, and α_R are the constant, $\bar{D}(t) = \frac{1}{N}\sum_i \phi_{D,i}D_i(t)$, and $\bar{Q}(t) = \frac{1}{N}\sum_i \phi_{R,i}\frac{Q_i(t)}{R_{m,i}^*}$.

2.4 The Optimal Solution of Utility-based Scheduling Algorithm

To the solution in Eq. (2.2), the polymatroid structure is used and the optimal solution is directly the key feature of a polymatroid. Here, we briefly describe the methodology to model the scheduling algorithm by a polymatroid structure and eventually arrive at the solution, and detailed proofs are ignored

Define by $\mathcal{N}_a = \{1, 2, \ldots, N\}$ the index set of all active connections and $2^{\mathcal{N}_a} = \{S : S \subset \mathcal{N}_a\}$ to be the collection of all subset. For $\forall S \subset \mathcal{N}_a$ and $\vec{X} = (x_i, i \in \mathcal{N}_a)$, a set function is defined for $\vec{X}$ by $\vec{X}(\cdot) : 2^{\mathcal{N}_a} \rightarrow \Re_+$, and a norm operator for it is also defined by $\left\|\vec{X}(S)\right\| = \sum_{i \in S} x_i$.

Definition 1: The Rate Region Conditioning on $\vec{c}(t)$

For given resource assignment vector $\vec{c}(t)$ and channel condition vector $\vec{\zeta}(t)$, the reasonable rate region is expressed by

$$\mathcal{C}_R\left(\vec{c}(t), \vec{\zeta}(t)\right) = \left\{\vec{r}(t) : \|\vec{r}(S)\| \leq \left\|\vec{\mathcal{R}}(S)\right\|, \forall S \subset \mathcal{N}_a\right\}, \tag{2.14}$$

where $\vec{r}(t)$ is the available rate assignment vector.

Definition 2: The Overall Reasonable Rate Region $\bar{\mathcal{C}}_R$

The overall reasonable rate region consists of all reasonable rates contained in $\mathcal{C}_R(\vec{c}(t), \vec{\zeta}(t))$, for all reasonable resource assignment vector $\vec{c}(t)$, where the reasonability of $\vec{c}(t)$ is the constraints on it. From the two constraints on $\vec{c}$, Ψ_1 and Ψ_2, we can formulate the over reasonalbe rate region $\bar{\mathcal{C}}_R$ by

$$\bar{\mathcal{C}}_R = \bigcup_{\vec{c}(t) \in \Psi_1 \cap \Psi_2} \mathcal{C}_R\left(\vec{c}(t), \vec{\zeta}(t)\right). \tag{2.15}$$

We can prove that the reasonable rate region $\mathcal{C}_R(\cdot)$ defined above is a polymatroid. Furthermore, for a given vector $\vec{\mu}(t) \in \Re_+^N$ and a resource assignment vector, the optimization problem

$$\max_{\vec{r}(t)} \vec{\mu} \cdot \vec{r}(t), \quad \text{subject to} \quad \vec{r}(t) \in \mathcal{C}_R \tag{2.16}$$

has a solution $\vec{r}^*$ associated with a permutation, $\pi^* : \mathcal{N}_a \to \mathcal{N}_a$, so that $\mu_{\pi^*(1)} \geq \mu_{\pi^*(2)} \geq \ldots \geq \mu_{\pi^*(N)}$. The solution $\vec{r}^*$ can be derived by greedy algorithm defined by the radio resource function. Denote by $\mu_i(t) = \mathcal{A}_i(t) \cdot \mathcal{F}_i(t)$ the prodcut of QoS requirement achievement function and weighted fairness function. Then the problem formulaion of scheduling algorithm defined in Eq. (2.1) of section 2.3 is rewritten by

$$\vec{c}^*(t) = arg \max_{\vec{c}(t)} \left\{ \vec{\mu}(t) \cdot \vec{r}(t) \right\}, \quad \vec{r}(t) \in \bar{\mathcal{C}}_R. \tag{2.17}$$

Moreover, for a given $\vec{\mu}(t)$, $\vec{r}^*(t)$ is the solution to the above problem, then there exists a $\lambda \in \Re^N_+$, reasonalbe rate vector $\vec{r}(t)$, and a radio resource assignment vector $\vec{c}(t)$ so that $(\vec{r}(t), \vec{c}(t))$ is the solution to the optimization problem

$$\begin{aligned} &\max_{(\vec{r}(t), \vec{c}(t))} \left\{ \vec{\mu}(t) \cdot \vec{r}(t) - \vec{\lambda}(t) \cdot \vec{c}(t) \right\}, \\ &\text{subuect to } \vec{r}(t) \in \mathcal{C}_R \left(\vec{c}(t), \vec{\zeta}(t) \right), \end{aligned} \tag{2.18}$$

where $\vec{r}(t) = \vec{r}^*(t)$ and $\vec{c}(t) \in \Psi_1 \cap \Psi_2$. The rank function $\left\| \vec{\mathcal{R}}(\cdot) \right\|$ defined for reasonable rate region $\mathcal{C}_R(\vec{c}(t), \vec{\zeta}(t))$ is *generalized symmectric* and can be expressed by

$$\left\| \vec{\mathcal{R}}(S) \right\| = \left\| \mathcal{G} \left(\vec{c}(S) \right) \right\|. \tag{2.19}$$

Moreover, the formulation in Eq. (2.18) can be rewritten in terms of Eq. (2.19), and from the constraint Ψ_1, the second term of Eq. (2.19) $\vec{\lambda}(t) \cdot \vec{c}(t)$ is constant. Then the optimization problem in Eq. (2.18) can be shown to be

$$\max_{\vec{c}(t)\in\Psi_1\cap\Psi_2} \left\{\vec{\mu}(t)\cdot\vec{r}(t)\right\},$$
$$\text{subuect to } \left\|\vec{r}(S)\right\| \in \left\|\mathcal{G}\left(\vec{c}(S)\right)\right\|, \forall S \subset \mathcal{N}. \tag{2.20}$$

Furthermore, the optimal rate vector $\vec{r}^*(t)$ can be obtained using greedy algorithm while the $\vec{c}(t)$ reaches the boundary on $\Psi_1 \cap \Psi_2$ with repect to the best permutation π^*. Finally, we can obtain the optimal solution by the greedy algorithm as follows,

Algorithm 1: Optimal Resource Scheduling Solution

Define $\pi^* : \mathcal{N}_a \to \mathcal{N}_a$ is the permutation so that $\lambda_{\pi^*(1)} \geq \lambda_{\pi^*(2)} \geq \ldots \geq \lambda_{\pi^*(N)}$ where $\lambda_i = \mu_i \cdot \left\|\nabla_i \mathcal{G}\left(\vec{c}(S)\right)\right\|$. And $r_{\pi^*(1)}(t) = \left\|\mathcal{R}\left(\{\pi^*(1)\}\right)\right\|$, $r_{\pi^*(i)}(t) = \left\|\mathcal{R}\left(\{\pi^*(1),\ldots,\pi^*(i)\}\right)\right\| - \left\|\mathcal{R}\left(\{\pi^*(1),\ldots,\pi^*(i-1)\}\right)\right\|$. The $c_{\pi^*(i)}(t)$ is therefore set to support the rate $r_{\pi^*(i)}(t)$ till $\sum_i c_{\pi^*(i)}(t) = 1$.

2.5 Results and Conclusions

To investigate the performance of the proposed utility-based scheduling algorithm, we compare its average throughput with the *Exponential Rule* scheduling algorithm. Two typical types of services are simulated in the scenario. Type-1 service is assumed to be real-time class of traffic with traffic rate 15kbps, activity 0.45, $P_D^* = 0.05$, $D^* = 40$ms, and $BER^* = 0.01$. Type-2 service is assumed to be non-real-time class of traffic with batch Poisson process of which the mean rate is 12kbps, peak rate is 128kbps, and $R_m^* = 12$kbps and BER^*=0.001. Type-1 connections use dedicated channels, while type-2 connections transmit in the shared channel. Four modulation schemes, BPSK, QPSK, 16QAM, and 64QAM, are available for transmission, as the

BER requirement can be fulfilled and the remaining queue is enough. In this simulation scenario, the ratio of the number of type-1 and type-2 connections is kept 1.5. We increase the number of connections and examine the average throughput of the utility-base scheduling algorithm compared with the Exponential Rule scheduling algorithm.

Fig. 2.1 shows the average throughput of utility-based scheduling algorithm compared with Exponential rule scheduling algorithm. It can be found that the utility-based scheduling scheme has higher throughput than that of exponential rule scheme when both dedicated and shared channels are used in the system. As the number of connections increases, the gain in throughput goes higher. The optimal radio resource assignment vector $\vec{c}(t)$ is the most efficient way to schedule the resource for connections with highest system utility in terms of the link quality, the QoS achievement, and the fairness. The location-dependent fairness and the priority bias further make the utility-based scheduling scheme reduce the resource consumption on the connections with bad link quality. The utility-based scheduling algorithm is effective for multimedia CDMA systems with diverse of QoS requirements and with dedicated and shared channels support. More complicated simulation scenarios with variant service types will be carried out to further investigate the performance and the features of the utility-based scheduling algorithm.

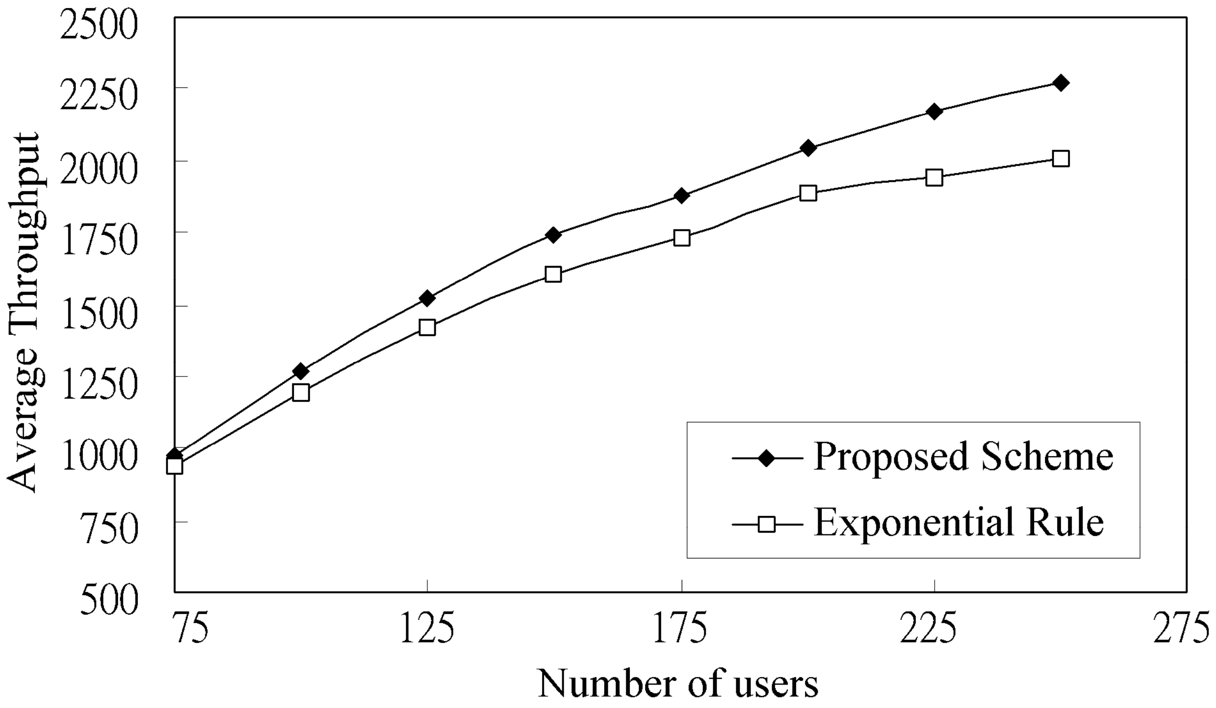

Figure 2.1: The average throughput of utility-based scheduling algorithm compared with Exponential rule scheduling algorithm.

Bibliography

[1] 3rd Generation Partnership Project, Technical Specification Group Services and System Aspects, Working Group 2 (WG2), "QoS Concept and Architecture," *3GPP TS 23.107*, V4.1.0, Jun. 2001.

[2] A. Parekh and R. G. Gallager, "A generalized processor sharing approach to flow control in integrated services networks: the single node case," *IEEE/ACM Trans. Networking*, vol. 1, no. 3, pp. 344-357, JUN. 1993.

[3] D. Ferrari and D. Verma, "A scheme for real-time channel establishment in wide-area networks," *IEEE J. Select. Area Commun.*, vol. 8, no. 3, pp. 368-379, Apr. 1990.

[4] Y. M. Lu and R. W. Brodersen, "Integrating power control, error correction coding, and scheduling for a CDMA downlink system," *IEEE J. Select. Area Commun.*, vol. 17, no. 5, pp. 978-989, May 1999.

[5] V. K. N. Lau, and Y. K. Kwok, "On generalized optimal scheduling of high data-rate bursts in CDMA systems," *IEEE Trans. Commun.*, vol. 51, no. 2, pp. 261-266, Feb 2003.

[6] V. Bharghavan, S. W. Lu, and T. Nandagopal, "Fair queuing in wireless networks: issues and approaches," *IEEE Personal Commun.*, vol. 6, no.1, pp. 44-53, Feb. 1999.

[7] A. C. Varsou and H. V. Poor, "HOLPRO: a new rate scheduling algorithm for the downlink of CDMA networks," *Proc. IEEE VTC'2000*, pp. 948-954.

[8] A. L. Stolyar and K. Ramanan, "Largest weighted delay first scheduling: large deviations and optimality," *Annal Appl. Prob.*, vol. 11, no. 1, pp. 1-48, 2001.

[9] S. Shakkottai and A. L. Stolyar, "Scheduling algorithms for a mixture of real-time and non-real-time data in HDR," *Bell Lab Reports.*

[10] C. S. Chang and J. A. Thomas, "Effective Bandwidth in High-Speed Digital Networks," *IEEE J. Select. Area Commun.,* vol. 13, no. 6, AUG. 1995.

[11] A. J. Goldsmith and S. G. Chua, "Variable-rate variable-power MQAM for fading channels," *IEEE Trans. Commun.*, vol. 45, no. 10, pp. 1218-1230, Oct. 1997.

[12] L. S. Chou and C. S. Chang, "Experiments of the theory of effective bandwidth for markov sources and video traces," *Proc. IEEE INFOCOM'96*, pp.

[13] M. Andrews, K. Kumaran, K. Ramanan, A. Stolyar, P. Whiting, and R. Vijayakumar, "Providing quality of service over a shared wireless link," IEEE Commun. Mag., vol. 39, no. 2 , pp. 150-154, Feb. 2001.

[14] D. N. C. Tse and S. V. Hanly, "Multiaccess fading channels - part I: polymatroid structure, optimal resource allocation and throughput ca-

pacities," *IEEE Trans. Info. Theory*, vol. 44, no. 7, pp. 2796-2815, Nov. 1998.

Chapter 3

A Situation-Aware Data Access Manager for Multi-cell WCDMA Systems Using Fuzzy Q-learning Technique

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In this chapter, a novel situation-aware data access manager using fuzzy Q-learning technique (FQ-SDAM) is proposed for multi-cell WCDMA systems. The FQ-SDAM contains a fuzzy Q-learning-based residual capacity estimator (FQ-RCE) and a data rate scheduler (DRS). The FQ-RCE accurately estimates the situation-dependent residual system capacity and the DRS effectively allocates the resource for non-real-time terminals by adopting a modified exponential rule which takes the location-dependent link capacity into consideration. Simulation results show that, compared to the link and interference-based demand assignment (LIDA) scheme [7], the proposed FQ-SDAM can effectively reduce the packet error probability.

3.1 Introduction

The WCDMA cellular system will support integrated services with mixed QoS (quality of services) requirements: real-time services require continuous transmission and intolerance to time delay, while non-real-time services require bursty transmission and tolerance to moderate time delay. For maximizing the system capacity and fulfilling the complementary QoS requirements, an adequate radio resource management (RRM) is required. The RRM adopts a call admission control scheme to ensure that the system will not be overloaded, based on the long-term availability of radio resources. On the other hand, it employs a *data access control scheme* to provide bursty transmission permission for non-real-time services, based on the short-term availability of radio resources.

The capacity of WCDMA system is interference-limited and is affected by the up-link multiple access interference (MAI). In [1], a simulation study showed that the interference and SIR (signal to interference) variations in an integrated services environment are much larger than those in a voice-only service one. The behavior of interference variation is also influenced strongly by factors, such as: user location and path loss/shadowing; therefore, one service profile could result in various kinds of interference level in different kinds of transmission scenarios. Taking advantage of the short-term variation of the network dynamics, the data access control scheme can effectively improve the system throughput.

The main purpose of the *data access control scheme* in WCDMA systems supporting integrated services is to maximize the throughput of non-real-time services while maintaining the transmission quality of real-time services

[2]-[5]. To achieve this goal, dynamic access probability schemes [2]-[4] and a base station-controlled scheduling scheme [5] were proposed, where the residual system capacity for non-real-time services was firstly estimated and then shared to non-real-time terminals. A single-cell environment was considered in [2]-[4], while a multi-cell environment was studied in [5]. In the multi-cell scheme [5], the interference generated from other-cell terminals was treated as if from several home-cell ones, and consequently the multi-cell environment was regarded as a single-cell one. However, the mutual-affected behavior of radio resource allocation in the multi-cell environment was still not considered. Note that, in the multi-cell WCDMA system, the increment of data transmission power in one cell would cause the rising of interference level in the adjacent cells. If each cell allocates the whole possible residual capacity for bursty transmission without taking the interference influence from adjacent cells into account, the system then would trap into an overloading condition.

The over-loading phenomenon could be alleviated by an appropriate coordination scheme among cells [6]. Knowing the information of radio resource of all cells, a centralized data access scheme for the multi-cell WCDMA system can maximize the system throughput by applying a global optimization method. Unfortunately, the coordination procedure takes long time to transact the resource information between cells so that it is infeasible for practical implementation. Usually, the data access control scheme operates in the short-term time scale, *e.g.* frame time, which makes a distributed one preferable. Kumar and Nanda [7] proposed a distributed scheme called load and interference-based demand assignment (LIDA). The LIDA is a kind of resource reservation-based scheme in which some portions of resource in each cell are reserved against the interference variation. Also, it employs the concept of burst admission threshold for high-rate transmission in a cell to avoid excess interference to adjacent cells . In this scheme, only when the strength difference between the received pilot signals from home cell and adjacent cell is larger than the threshold, the bursty transmission is permitted. The effectiveness of this scheme relies on the selection of the reservation threshold, which should be dynamically performed according to the system loading and the received interference power level.

Additionally, a rate scheduling scheme is also embedded in the data access control scheme to allocate the residual capacities for non-real-time terminals according to a service principle. Ramakrishna and Holtzman adopted a *maximization throughput* criterion for the scheduling scheme [8]. Indeed, this criterion can maximize the system throughput, but the low-class users may suffer from starvation. Alternatively, Jalali, Padovani, and Pankai proposed a *proportional fairness* criterion [9] for a down link scheduling scheme in a CDMA-HDR (high data rate) system. In the scheme, a utility function was defined as a ratio of the supported data rate and the average data rate. The supported data rate was determined by channel condition, while the average data rate was the window average of the transmitted throughput. The terminal with the highest utility value would transmit in the next frame time. This algorithm may lead to large transmission delay for some terminals. Also, Shakkottai and Stolyar proposed an *exponential rule* criterion [10] for the utility function of the scheme to make a good balance between the system throughput and the transmission delay. However, applying the exponential rule to the uplink transmission should take the location factor into consideration such that the adjacent cell interference could be maintained to a sustained level.

This chapter proposes a situation-aware data access manager using fuzzy Q-learning technique (FQ-SDAM) for multi-cell WCDMA systems. The FQ-SDAM scheme consists of two parts: *fuzzy Q-learning-based residual capacity estimator* (FQ-RCE) and *data rate scheduler* (DRS). The FQ-RCE, by fuzzy Q-learning technique, estimates the appropriate situation-dependent residual system capacity, in term of interference power, for non-real-time services; the DRS assigns transmission rates for non-real-time terminals by a modified exponential rule.

Fuzzy inference system (FIS) and reinforcement learning technique have been separately applied to solve network resource management problems [11], [12], [13], and [14]. A fuzzy resource allocation controller was proposed, where the FIS technique was adopted to estimate the resource availability [12]. A reinforcement learning technique, Q-learning, was applied respectively to deal with the dynamic channel assignment in [13] and multi-rate transmission control problems in [14] for wireless communication systems. By means of learning from the system environment, the Q-learning technique can converge to a pre-defined optimal control target. In [15], Jouffle proposed a reinforcement learning technique for FIS, called fuzzy Q-learning (FQL). The FQL technique combines the benefits of FIS and reinforcement learning, where the FIS provides a good function approximation for the FQL and *a priori* knowledge can be easily applied to the system design; also, the reinforcement learning provides a model-free approach to obtain a control target. By applying the FQL technique, the radio resource, therefore, can be managed under partial, uncertain information, and the optimal resource management can be reached in an incremental way.

The FQ-RCE chooses three essential measures of interferences: the received real-time interference from home cell, the received non-real-time interference from home cell, and the received interference from adjacent cells as input linguistic variables to estimate the situation-dependent residual capacity in the multi-cell environment. Note that the received interference from home cell is regarded as different different variable from the received interference from adjacent cells in order to distinguish their according variations. Therefore, by extent of the measuring the adjacent-cell interference, the FQ-RCE in the home cell can implicitly *perceive* the extent of the radio resource allocation by those FQ-SDAMs in the adjacent cells. The FQ-RCE can be aware of the loading situation of the multi-cell WCDMA environment and precisely estimate the residual resource in a distributed fashion; thus, an explicit action coordination scheme is negligible.

On the other hand, the DRS adopts a modified exponential rule to assign the transmission rates for non-real-time terminals, based on the residual capacity estimated by FQ-RCE. The modified exponential rule is a utility function-based scheduling algorithm which considers factors of transmission delay, average transmission rate, and link capacity. Its main difference from the original exponential rule [10] is on the definition of the link capacity. For the modified exponential rule, the link capacity is defined as the maximum available rate under current link condition while the transmission power emitted to the adjacent cells is below a threshold. With the feature of location awareness, the modified exponential rule is more suitable for applications in the uplink transmission of multi-cell WCDMA systems. Simulation results show that the proposed FQ-SDAM outperforms the LIDA scheme and it can effectively reduce the packet error probability.

The rest of the chapter is organized as follows. The system model is described in Section 4.2. The concept of fuzzy Q-learning is briefly stated and the design of FQ-SDAM is proposed in Section 4.3. Simulation results are presented in Section 4.4, where the performance comparison between the FQ-SDAM and the *conventional scheme* is made. Finally, concluding remarks are given in Section 4.5.

3.2 System Model

A multi-cell WCDMA system containing N cells is considered, where each cell has a base station with an omni-directional antenna to take charge of communicating with real-time and non-real-time terminals within its coverage area. The reverse link that supports slotted transmission is adopted. Each terminal transmits at the same frequency band and is distinguished by its own spreading code. It holds two communication channels: dedicated physical data channel (DPDCH) and dedicated physical control channel (DPCCH). The DPDCH is used to carry data generated by layer 2 protocol, while the DPCCH is used to carry control information. The channel is defined in a frame-based structure, where the frame length $T_f = 10$ ms is divided into 15 slots with length $T_{\text{slot}} = 2560$ chips, each slot corresponding to one power control period. Hence, the power control frequency is 1500 Hz. The spreading factor (SF) for DPDCH can vary between 32 $\sim$ 256 by SF $= 256/2^k, k =$

$0, 1, \cdots, 6$, carrying 10×2^k bits per slot, and the SF for DPCCH is fixed at 256, carrying 10 bits per slot.

Two types of traffic are considered in this chapter: real-time traffic as type-1 and non-real-time traffic as type-2. The system provides continuous transmission for real-time traffic and bursty transmission for non-real-time traffic. The real-time terminals may transmit at any possible data rate while necessary; on the other hand, the transmission of non-real-time terminals is controlled by the base station. Considering terminal's link gain and the received interference strength from both home and adjacent cells, the base station assigns an appropriate data rate for each non-real-time terminal. For the bursty transmission, the available data transmission rates are 1X, 2X, 4X and 8X, and 1X transmission rate is called the basic rate. A strength-based power control scheme is assumed such that the required transmission power is directly proportional to the transmission rate. Also, the overall capacity is determined by the upper bound of the total received interference power, and the residual capacity is defined as the allowable received interference power from the non-real-time terminals.

The link gain between terminal i to base station j, denoted by h_{ij}, is usually determined by the long-term fading FL_{ij} and the short-term fading FS_{ij} [18], which is given by

$$h_{ij} = FL_{ij} \times FS_{ij}. \tag{3.1}$$

The long-term fading FL_{ij}, combining the path loss and shadowing, is modelled as

$$FL_{ij} = k \times r^{-\alpha} \times 10^{\eta/10}, \tag{3.2}$$

where k is constant, r is distance from mobile i to base station j, α is path loss exponent whose value usually lies between two and five for mobile environment ($\alpha = 4$ in this chapter), and η is normal-distributed random variable with zero mean and variance σ_L^2. The parameter σ_L is affected by the configuration of the terrain and ranges from 5 to 12 (σ_L^2=10 in this chapter). The short-term fading FS_{ij} is mainly caused by multi-path reflections, and it is modelled by Rayleigh distribution.

The real-time service is modelled as an ON-OFF Markov process with a transition rate μ from ON to OFF and λ from OFF to ON state. The non-real-time service is modelled as a batch Poisson process; that is, the arrival process of the data burst is in Poisson distribution and the data length is assumed to be with a geometric distribution. An erroneous real-time packet will be dropped since there is no re-transmission for real-time packets, while the erroneous non-real-time packets will be recovered via ARQ (automatic repeat request) scheme. The measure of the packet error probability, denoted by P_e, and packet transmission delay, denoted by D_d, are regarded as the system performance indices. Also, the maximum tolerable packet error probability, denoted by P_e^*, is defined as the system QoS requirement.

3.3 Design of FQ-SDAM

The FQ-SDAM contains two functional blocks of a fuzzy Q-learning-based residual capacity estimator (FQ-RCE) and a data rate scheduler (DRS). The FQ-RCE estimates the residual interference power budget, and then the DRS allocates the resource for the non-real-time terminals. In the following, the fuzzy Q-learning and the detailed design of the two function blocks are described.

3.3.1 The Fuzzy Q-Learning (FQL)

Denote **S** the set of state vectors for the system, **S**=$\{S_i,\ i = 1, 2, \cdots, M\}$; each state vector S_i is constituted by L fuzzy linguistic variables selected to describe the system. Denote **A** the set of actions that are possibly chosen by system states, **A**=$\{A_j,\ j = 1, 2, \cdots, N\}$. For an input state vector $\mathbf{x}$ containing the L linguistic variables, the rule representation of FQL for state S_i is in the form by

if $\mathbf{x}$ is S_i, **then** A_j with $q[S_i, A_j]$, $1 \le i \le M$ and $1 \le j \le N$,

where A_j is the j-th action candidate that is possibly chosen by state S_i, and $q[S_i, A_j]$ is the Q-value for the state-action pair (S_i, A_j). The number of state-action pairs for each state S_i is equal to the number of the elements in the action set; *i.e.*, there are N possible consequence parts for the same antecedent. Every fuzzy rule has to choose an action A_i out of the action candidates set **A** by an action selection policy. In the FQL, the action selection policy for each fuzzy rule may be *select-max* or other exploration strategy. As to the defuzzification of the M fuzzy rules, the inferred action $a(\mathbf{x})$ for the input vector $\mathbf{x}$ is expressed as

$$a(\mathbf{x}) = \frac{\sum_{i=1}^{M} \alpha_i \times A_i}{\sum_{i=1}^{M} \alpha_i}, \tag{3.3}$$

where α_i is the truth value of the rule representation of FQL for state S_i. Also, the Q-value for the state-action pair $(\mathbf{x}, a(\mathbf{x}))$ is

$$Q(\mathbf{x}, a(\mathbf{x})) = \frac{\sum_{i=1}^{M} \alpha_i \times q[S_i, A_i]}{\sum_{i=1}^{M} \alpha_i}. \tag{3.4}$$

For the current system state $\mathbf{x}$, after applying the chosen action $a(\mathbf{x})$, the next-stage system state is assumed at $\mathbf{y}$ and the system reinforcement signal is $c(\mathbf{x}, a(\mathbf{x}))$. To update the Q-value, the next-stage optimal Q-value, $Q^*(\mathbf{y}, a(\mathbf{y}))$, is defined as

$$Q^*(\mathbf{y}, a(\mathbf{y})) = \frac{\Sigma_{i=1}^{M} \alpha_i \times q[S_i, a_i^*]}{\Sigma_{i=1}^{M} \alpha_i}, \tag{3.5}$$

where $q[S_i, a_i^*]$ is the Q-value of state-action pair $(S_i,\ a_i^*)$ and $a_i^* = \underset{A_j}{\text{argmax}}$ $\{\text{q}[S_i,\ A_j]\}$. According to the Q-learning rule [17], the Q-value update in the FQL can be expressed as

$$q[S_i, a_i] = q[S_i, a_i] + \eta \Delta q[S_i, a_i], \tag{3.6}$$

where η is the learning rate, $0 \leq \eta \leq 1$, and

$$\Delta q[S_i, a_i] = \{c(\mathbf{x}, a(\mathbf{x})) + \gamma Q^*(\mathbf{y}, a(\mathbf{y})) - Q(\mathbf{x}, a(\mathbf{x}))\} \times \frac{\alpha_i}{\Sigma_{k=1}^{M} \alpha_k}. \tag{3.7}$$

$c(\mathbf{x}, a(\mathbf{x}))$ in (3.7) is the reinforcement signal.

3.3.2 Fuzzy Q-learning-based Residual Capacity Estimator (FQ-RCE)

The received interference in the WCDMA system is a good indicator of the system loading. Also, in WCDMA system, the interference generated from home cell can be distinguished by PN codes and the interference from adjacent cells can be distinguished by long scrambling codes [20]. Therefore, three interference measures: the received real-time interference from home cell (I_{h1}), the received non-real-time interference from home cell (I_{h2}), and the received interference from adjacent cells (I_o), are chosen as the system state vector for FQ-RCE. Accordingly, the vector $\mathbf{x}$ containing the three linguistic variables input to FQ-RCE is defined as

$$\mathbf{x} = (I_{h1}, I_{h2}, I_o). \tag{3.8}$$

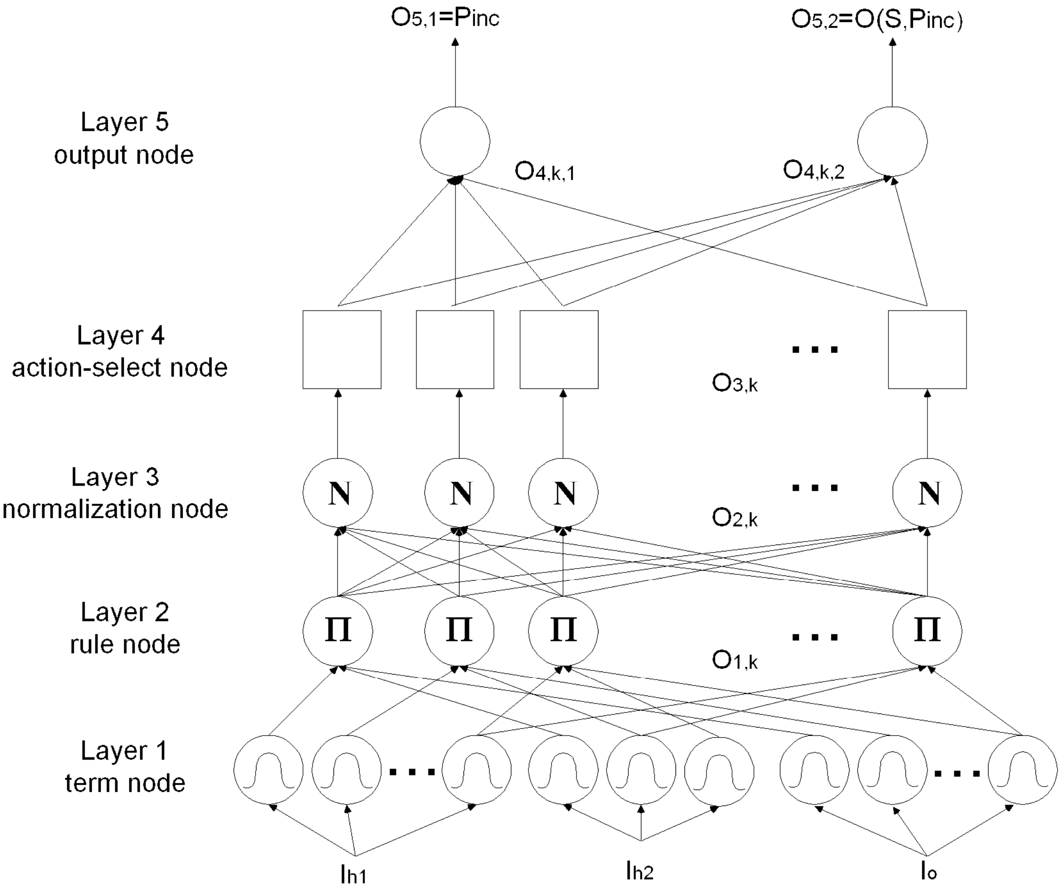

Figure 3.1: Structure of FQ-RCE

After comprehensive simulation experiments, it is found that five terms for both I_{h1} and I_o, and three terms for I_{h2} are proper. Hence, their fuzzy term sets are T(I_{h1})={Largely High, HiGh, MeDium, LoW, Largely Low}={LH, HG, MD, LW, LL}, T(I_{h2})={HiGh, MeDium, LoW}={HG, MD, LW}, and T(I_o)={Largely High, HiGh, MeDium, LoW, Largely Low}={LH, HG, MD, LW, LL}. From the fuzzy set theory, the fuzzy rule base forms with dimensions $|\mathrm{T}(I_{h1})|\times|\mathrm{T}(I_{h2})|\times\mathrm{T}|(I_o)|$. Accordingly, M=75. On the other hand, the step-wise incremental/decremental action of the interference power budget for the non-real-time services, denoted by P_{inc}, is selected as the output linguistic variable. Here, seven levels of increment actions (N=7) are decided, and the corresponding fuzzy term set is T(P_{inc})={PI_1, PI_2, PI_3, PI_4, PI_5, PI_6, PI_7}. After the interference increment is estimated by the FQ-RCE, the residual system capacity (RC) allocated for the non-real-time services is given by

$$RC = I_{h2} + P_{inc}. \tag{3.9}$$

Also, the reinforcement learning signal $c(\mathbf{x}, a(\mathbf{x}))$ is defined as

$$c(\mathbf{x}, a(\mathbf{x})) = [\frac{P_e(\mathbf{x}, P_{inc}) - P_e^*}{P_e^*}]^2, \tag{3.10}$$

where $P_e(\mathbf{x}, P_{inc})$ is the packet error probability of real-time services for the state-action pair $(\mathbf{x}, P_{inc})$, which is a performance measure of the system, and P_e^* is the QoS requirement of real-time packet error probability.

Fig. 3.1 shows the structure of FQ-RCE; it is a five-layer adaptive-network-based implementation of fuzzy inference system. In the FQ-RCE, *layer 1* to *layer 3* are the antecedent part of the FIS while *layer 4* and *layer 5* represent the consequence part. The node function in each layer is described as follows.

Layer 1: Every node k, $1 \leq k \leq 13$, in this layer is a term node which represents a fuzzy term of an input linguistic variable, where k= 1, $\cdots$, 5 (6, 7, 8) (9, $\cdots$, 13) denotes node k being the k-th ((k-5)-th) ((k-8)-th) term in $T(I_{h1})$ ($T(I_{h2})$) ($T(I_o)$), respectively. The node function is defined to be the membership function with bell shape for the term. Thus, for an input linguistic variable x, the output $O_{1,k}$ is given by

$$O_{1,k} = b(x; m^k, \sigma^k), \tag{3.11}$$

where $b(\cdot)$ is the bell-shaped function, and m^k and σ^k is the mean and the variance of the node k.

Layer 2: Every node k, $1 \leq k \leq 75$, in this layer is a rule node which represents the truth value of k-th fuzzy rule; it is a *fuzzy*-AND operator. Here, the product operation is employed as the node function. Since each fuzzy rule has three input linguistic variables, the node output $O_{2,k}$ is the product sum of three fuzzy membership values corresponding to the inputs. Therefore, $O_{2,k}$ is given by

$$O_{2,k} = \prod\{O_{1,l}\}, \forall l \in P_k, \tag{3.12}$$

where P_k=$\{l|$ all ls that are the pre-condition nodes of the k-th fuzzy rule$\}$.

Layer 3: Every node k, $1 \leq k \leq 75$, in this layer is a normalization node which performs a normalization operation; it makes the summation of the all truth values be unity. After the normalization, the output of this node $O_{3,k}$ is given by

$$O_{3,k} = \frac{O_{2,k}}{\sum_{l=1}^{75} O_{2,l}} \tag{3.13}$$

Layer 4: Every node k, $1 \leq k \leq 75$, in this layer is an action-select node which represents the consequence part of k-th fuzzy rule. Based on the action selection policy and Q-values of the possible action candidates (PI_j, $j = 1, 2, \cdots 7$), the node is to choose an appropriate action. Since improper initial setting of fuzzy parameters would lead to a bad learning result, the Boltzmann-distributed exploration strategy in [19] is employed to explore the set of all the possible action candidates. In the Boltzmann-distributed exploration, the node chooses the state-action pair (S_k, a_k), $a_k \in T(P_{inc})$, for the k-th rule, with the probability $\xi(S_k, a_k)$ given by

$$\xi(S_k, a_k) = \frac{e^{q[S_k, a_k]/T}}{\sum_{j=1}^{7} e^{q[S_k, PI_j]/T}}, \tag{3.14}$$

where T is the *temperature* which reflects the randomness of action selection. After the action is chosen, the node sends two outputs $O_{4,k,1}$ and $O_{4,k,2}$ to the action node and Q-value node in layer 5, respectively. $O_{4,k,1}$ and $O_{4,k,2}$ are represented by

$$O_{4,k,1} = O_{3,k} \times a_k, \tag{3.15}$$

and

$$O_{4,k,2} = O_{3,k} \times q[S_k, a_k]. \tag{3.16}$$

Layer 5: There are two output nodes in this layer: action node $O_{5,1}$ and Q-value node $O_{5,2}$, which represent the fuzzy defuzzification of FQ-RCE. Here, the center of the area defuzzification method is applied. Since the truth value of the antecedent part of the i-th fuzzy rule is normalized in layer 3, the node functions in this layer are summation of the inputs from layer 4. Hence, $O_{5,1}$ and $O_{5,2}$ are given by

$$O_{5,1} = P_{inc} = \sum_{k=1}^{M=75} O_{4,k,1}, \tag{3.17}$$

and

$$O_{5,2} = Q(\mathbf{x}, P_{inc}) = \sum_{k=1}^{M=75} O_{4,k,2}. \tag{3.18}$$

After the action is performed, the FQ-RCE computes the reinforcement signal $c(\mathbf{x}, a(\mathbf{x}))$ by (3.10) and updates the Q-value of each state-action pair according to (3.6).

It is noted that the convergence property of Q-learning is only held for the single-agent (learner) case but may not be held for multiple-agent cases. The convergence of Q-learning would be a difficult task for the multi-cell WCDMA systems because the decision policies of all cells change concurrently during the learning phase. To deal with this problem, the perceptual coordination mechanism (PCM) is applied to design the input linguistic variables. In FQ-RCE, the input variables are the received interference which are separated into two parts: the received interferences from home cell, I_{h1} and I_{h2}, represent the current state of the radio resource usage in home cell; the received interference from adjacent cell, I_o, represents the radio resource allocations performed in the adjacent cells. The radio resource allocations of the adjacent cells result in the interference fluctuation; by measuring the adjacent-cell interference, the FQ-RCE in the home cell can implicitly *perceive* the radio resource allocation (action) in the adjacent cells. That is, the action policies are transacted implicitly in the multi-cell environment. By PCM, the multi-cell learning environment could be simplified as a single-cell one. The convergence property for the FQ-RCE can still be held henceforth.

3.3.3 The Data Rate Scheduler (DRS)

A modified exponential rule scheduling algorithm is proposed for DRS. The formula of the modified exponential rule is given by

$$j = \underset{i}{\text{argmax}} \, \{\frac{r_i}{\bar{r}_i} \times e^{\frac{W_i - \overline{\mathbf{W}}}{1+\sqrt{\overline{\mathbf{W}}}}}\}, \tag{3.19}$$

where r_i, $\bar{r}_i$, and W_i are the link capacity, the average transmission rate, and the waiting time of the i-th data terminal, respectively, and $\overline{\mathbf{W}}$ is the average waiting time of all the data terminals. The main difference between the modified and the original exponential rules is the definition of the link capacity. The original exponential rule was proposed for downlink transmission in the CDMA HDR system [9], where the link capacity was defined as the maximum transmission rate under current link condition. However, in the multi-cell WCDMA environment, the uplink transmission power would interfere the adjacent cells. The closer the terminal near the cell boundary is, the larger the interference power will be. Therefore, the modified exponential rule algorithm takes this factor into consideration, and the location-dependent link capacity r_i is defined as the maximum available transmission rate that satisfies the following condition:

$$P(r_i) \times h_k \leq P_d \tag{3.20}$$

and

$$h_k = \max\{G_{ij} | j \in \text{adjacent cells}\}, \tag{3.21}$$

where $P(r_i)$ is the transmission power of rate r_i, h_k is the best adjacent-cell link gain of the terminal, and P_d is a guard threshold. Here, the condition (3.20) is used to keep the adjacent-cell interference emitted from single terminal lower than a pre-defined threshold. In the strength-based power control scheme, the transmission power $P(r_i)$ is given by

$$P(r_i) = \frac{r_i \times (E_b/N_0)^* \times I_{max}}{PG \times h_i}, \tag{3.22}$$

where $(E_b/N_0)^*$ is the signal-to-noise requirement, I_{max} is the maximum received interference power, PG is the processing gain, and h_i is the home-cell link gain of the terminal. Also, h_i and h_k can be measured by monitoring the received pilot strength from the home and adjacent cells. Hence, the modified exponential rule can be considered as the uplink version of the original exponential rule, which is now interpreted as: *the terminal with higher maximum available transmission rate, lower average transmitted rate, and longer delay will get higher transmission priority.* As the terminal moves toward the cell boundary, the emission power to the adjacent cells will go high. In this case, the transmission priority will be low and the waiting time will accumulate. However, as the terminal's waiting time is long, the transmission priority will be high. Therefore, the modified exponential rule can make a balance between the link gain, the location, and the waiting time of terminals.

The DRS performs the rate allocation according to terminal's priority. The terminal with highest priority gets the rate allocation first, and other terminals get the allocation in order. The operation of the DRS will stop until all the data power budget is used out. Its procedure is described in the following:

[**The DRS Algorithm**]

Step 1 Calculate the overall situation-dependent residual system capacity (RC) for non-real-time services by (3.9).

Step 2 Choose highest-priority data terminal j by (3.19).

Step 3 Compute the remaining RC by

$$RC = RC - P(r_j)/PG.$$

If the remaining RC is larger than 0, go back to **Step 2**. Otherwise, go to
Step 4.

Step 4 Inform terminals the assigned data rate via DPCCH. **End**

3.4 Simulation Results and Discussion

In the simulations, a concatenated 19-cell (N=19) environment is configured as the multi-cell WCDMA system. The central cell is labelled as cell 1, the cells in the first tier are cell 2 $\sim$ cell 7, and the cells in the second tier are cell 8 $\sim$ cell 19. Three kinds of real-time traffic are considered: voice traffic,

Table 3.1: TRAFFIC PARAMETERS IN THE MULTI-CELL WCDMA SYSTEM

Traffic Type	Traffic Parameters
2-level real-time voice	Mean talkspurt duration: 1.00 seconds Mean silence duration: 1.35 seconds
High-bursty real-time data traffic	Peak rate ($R_{p,h}$): 4-fold of basic rate Mean rate: 1-fold of basic rate ρ_h: 0.25
Low-bursty real-time data traffic	Peak rate ($R_{p,l}$): 2-fold of basic rate Mean rate: 1-fold of basic rate ρ_l: 0.5
Non-real-time data traffic	Mean data burst size: 200 packets $r_{\min}$: 1-fold of basic rate $r_{\max}$: 8-fold of basic rate

high-bursty real-time data traffic and low-bursty real-time data traffic. The voice traffic assumes 2-level transmission rate traffic which is modelled by a 2-level MMDP (Markov modulated deterministic process). The real-time data traffic is modelled by an ON/OFF traffic stream with specific burstiness $1/\rho_h$ ($1/\rho_l$) and peak rate $R_{p,h}$ ($R_{p,l}$) for high-bursty (low-bursty) real-time traffic. The two real-time data traffics have the same mean rate but different burstiness. On the other hand, the non-real-time data traffic is considered to have a Poisson arrival process with data burst length in geometric distribution. All the detailed traffic parameters are listed in Table. 3.1. A basic rate in the WCDMA system is assumed to be a physical channel with SF=256. For each connection, DPCCH is always active to maintain the connection reliability. To reduce the overhead cost of interference produced by DPCCHs, the transmitting power of a DPCCH is smaller than its respective DPDCH by an amount of 3 dB. The QoS requirement of the packet error parameter, P_e^*, is set to 0.01.

The conventional resource reservation scheme proposed in [7], LIDA (load and interference demand assignment), is used as a benchmark for performance comparison. The basic concept of the LIDA scheme is two-folded: firstly, a portion of interference power budget, β, is reserved to avoid the over-lading situation; secondly, burst-mode admission is applied for the high-rate traffic. In the LIDA scheme, the allocation of the incremental of transmission power, P_{inc}, to the non-real-time data traffic is given by

$$P_{inc} = (1-\beta)I_{max} - I_{h1} - I_{h2} - I_o. \tag{3.23}$$

Note that the performance of the LIDA scheme highly relies on the choice of reservation threshold, β. We consider, in the simulations, three different degrees of reservation threshold, β = 0%, 5%, and 10%, and apply the modified exponential rule to the LIDA scheme with P_d=2dB. Also, a scheme which combines the FQ-RCE with the original exponential rule, called FQ-RCE/EXP, is considered to further evaluate the effectiveness of the modified exponential rule. Noted that all the considered schemes are applied for non-real-time terminals. All the real-time terminals initiate data transmission whenever they have packets in queues.

3.4.1 Homogeneous Case

In the homogeneous case, all cells are assumed to contain 22 voice terminals, 40 real-time data terminals and 20 non-real-time data terminals. The forty real-time data terminals consists of $N_{D,h}$ ($N_{D,l}$) high-bursty (low-bursty) data users; obviously, $N_{D,h}$+$N_{D,l}$=40.

Fig. 3.2 shows the packet error probabilities versus the number of high-bursty real-time data users. From the figure, it can be found that the packet error probability of the LIDA scheme will violate the QoS requirement and the LIDA scheme without reservation (β=0%) has the largest packet error probability. The results justifies the necessity of a precise residual capacity estimation to avoid the overloading condition in the multi-cell WCDMA environment. As to the FQ-SDAM and FQ-RCE/EXP schemes, their packet error probabilities always satisfy the QoS requirement. The FQ-RCE estimates the residual system capacity by monitoring the loading status of the home cell and the interference variation of adjacent cells such that it can adaptively determine the residual capacity according to the current loadings in the home cell and the radio resource allocation in the adjacent cells. Also, whatever the number of $N_{D,h}$ is, the FQ-SDAM scheme always achieves lower packet error probabilities than the FQ-RCE/EXP scheme does. This is because the up-link transmission powers emitted from terminals interfere the users from home cell and adjacent cells in the multi-cell environment. With the awareness of location, the modified exponential rule effectively curbs the interference to a sustainable level, which would reduce the error probabilities consequently.

3.5 Concluding Remarks

For multi-cell WCDMA systems, a novel situation-aware data access manager using fuzzy Q-learning technique (FQ-SDAM) is proposed, which contains a fuzzy Q-learning-based residual capacity estimator (FQ-RCE) and a data rate scheduler (DRS). By applying the perceptual coordination method (PCM), the FQ-RCE treats the received home-cell interference and adjacent-cell interference as two separate linguistic variables such that it can adaptively determine the residual capacity according to the current loadings in the home cell and the radio resource allocation in the adjacent cells. Simulation results

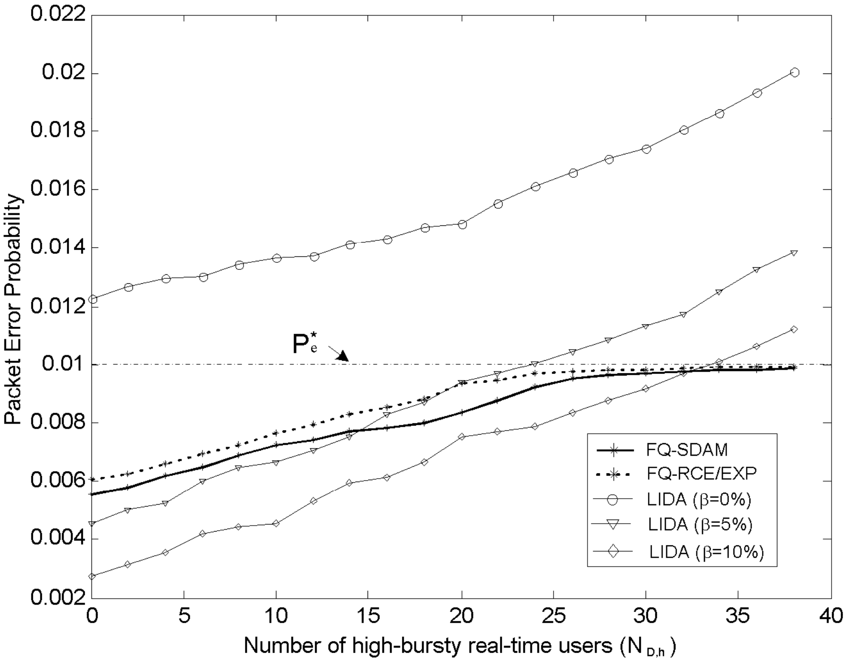

Figure 3.2: Packet error probabilities: homogeneous case

show that, compared to the LIDA scheme [7], the proposed FQ-SDAM can effectively reduce the packet error probability.

The present is still on-going. we will continue to evaluate the aggregate data throughput. Also, the performance of FQ-SDMA under non-homogeneous will be evaluated. All the remaining works will be presented in the final project report.

Bibliography

[1] K. Das and S. D. Morgera, "Interference and SIR in integrated voice/data wireless DS-CDMA networks - a simulation study," *IEEE J. Select. Areas. Commun.*, vol. 15, no. 8, pp. 1527-1537, Oct., 1997.

[2] T. K. Liu and J. A. Silvester, "Joint admission/congestion control for wireless CDMA systems supporting integrated services," *IEEE J. Select. Areas. Commun.*, vol. 16, no. 6, pp. 845-857, Aug., 1998.

[3] C. Comaniciu and N. B. Mandayam, "Delta modulation based prediction for access control in integrated voice/data CDMA systems," *IEEE J. Select. Areas. Commun.*, vol. 18, no. 1, pp. 112-122, Jan., 2000.

[4] A. Sampath and J. M. Holtzman, "Access control of data in integrated voice/data CDMA systems: benifits and tradeoffs," *IEEE J. Select. Areas. Commun.*, vol. 15, no. 8, pp. 1511-1526, Oct., 1997.

[5] C. Comaniciu, N. B. Mandayam, D. Famolari, and P. Agrawal, "Wireless access to the world wide web in an integrated CDMA system," *IEEE J. Select. Areas. Commun.*, vol. 2, pp. 472-483, May, 2003

[6] L. Chen , H. Kayama, and N. Umeda, "Power resource cooperation control considering wireless QoS for CDMA packet mobile communication systems," *IEEE Int'l symposium on personal, indoor and mobile radio communications* (PIMRC 2002), vol. 3, pp. 1092 -1096.

[7] S. Kumar and S. Nanda, "High data-rate packet communication for cellular network using CDMA: algorithm and performance," *IEEE J. Select. Areas. Commun.*, vol. 17, no. 3, pp. 472-492, Mar., 1999.

[8] S. Ramakrishna and J. M. Holtzman, "A scheme for throughput maximization in a dual-class CDMA system, " *IEEE J. Select. Areas. Commun.*, vol. 16, no. 6, pp. 830-844, Aug., 1998.

[9] A. Jalali, R. Padovani and R. Pankaj, "Data throughput of CDMA-HDR a high efficiency-high data rate personal communication wireless system," *IEEE Vehicular Technology Conference* (VTC2000- Spring), Tokyo, May 2000, pp.1854-1858.

[10] S. Shakkottai and A. L. Stolyar, "Scheduling algorithms for a mixture of real-time and non-real-time data in HDR," *17th International Teletraffic Congress* (ITC-17), Sep., 2001.

[11] L. Wang(Ed), Soft Computing in Communications, Springer, 2003.

[12] Y. S. Chen and C. J. Chang, "A resource allocation scheme using adaptive-network based fuzzy control for mobile multimedia network," *IEICE Trans. Commun.*, vol. E85-B, no. 2, pp. 502-513, Feb. 2002.

[13] J. Nie and S. Haykin, "A Q-learning-based dynamic channel assignment technique for mobile communication systems," *IEEE Trans. Veh. Technol.*, vol. 48, no. 5, pp. 1676-1687, Sep. 1999.

[14] Y. S. Chen, C. J. Chang, and F. C. Ren, "A Q-learning-based multi-rate transmission control scheme for RRM in multimedia WCDMA systems," *IEEE Trans. Veh. Technol.*, vol. 53, no. 1, pp 38-48, Jan. 2004.

[15] L. Jouffle, “Fuzzy inference system learning by reinforcement methods,” *IEEE Trans. Syst. Man. Cybern.*,, vol. 8, no. 3, pp. 338-355, Aug. 1998.

[16] O. Abul, F. Polat, and R. Alhajj, “Multiagent reinforcement learning using function approximation,” *IEEE Trans. Syst. Man. Cybern.*,, vol. 30, no. 4, pp. 485-497, Nov. 2000.

[17] C. J. C. H. Watkins and P. Dayan, “Q-learning,” *Machine Learning*, vol. 8, pp. 279-292, 1992.

[18] G. L. Stüber, Priniciple of mobile communication, Kluwer Academic Publishers, 1996.

[19] S. Haykin, Neural Networks 2nd. Prentice Hall, 1999.

[20] 3rd Generation Partnership Project. (2002, Spe.) Spreading and Modulation (FDD). 3GPP TS 25.213 [Online]http://www.3gpp.org

Chapter 4

A Power-Efficient MAC Protocol for Integrated Voice and Data Services over 802.11e HCF

A novel power-efficient MAC protocol, named on-demand polling (ODP) scheme, is presented for supporting integrated voice and data services over WLAN. The ODP scheme combines the polling-based and the contention-based mechanisms over HCF. In the ODP scheme, a voice station is in the polling list when it is in the active mode. During the idle mode, it is configured to operate in the sleep mode for saving power. Also, it adopts the contention-based mechanism to join the list again when returning from the idle mode. In this chapter, we investigate the RR-Brady, the RR-CBR and the proposed ODP schemes and derive the power consumption and the aggregate throughput for voice and data stations, respectively. Simulation results show that the ODP scheme can reduce the power consumption for voice stations and enhance the throughput for data stations.

4.1 Introduction

The integration of WLANs and 3G networks has recently evolved into a very hot topic. The service continuity is one of the issues needed to be resolved for the 3G/WLAN heterogeneous networks. In order to provide service continuity, it is essential for WLAN to support voice services. To support integrated voice and data services over WLAN, the WLAN has to support differentiated quality of services (QoSs) to guarantee the requirements of voice users. In addition to provide the differentiated QoS, the 3G/WLAN voice-enabled device needs to increase its stand-by time by reducing power consumption. Existing WLAN card may operate in either of three operation modes. In the active mode, all of the RF and based band chipsets are turning on and the power consumption is the highest. In the sleep mode, most of the functions are disabled and the power consumption could be minimized.

In order to support differentiated QoSs, the task group E of the IEEE 802.11 standardizes the MAC enhancements for WLANs, denoted as 802.11e [1]. The IEEE 802.11e defines a hybrid coordination function (HCF) which supports a contention-based and a polling-based channel accesses. The enhanced DCF (EDCF) is the contention-based channel access, which allows the QoS access point (QAP) to exchange frames with QoS stations (QSTAs) based on CSMA/CA mechanism. The HCF controlled channel access adopts a polling-based mechanism, which allows the QAP to enable the contention-free frame exchange with the QSTAs during a contention period. Several papers [2-4] studied the performance of integrated voice and data services over WLANs. In these chapters, the operation of WLANs is limited to a point coordinator function (PCF) and a distributed coordinator function (DCF), instead of the HCF. Moreover, a round-robin (RR) polling scheme was adopted to schedule the voice sources. It may cause power waste due to the excess polling of silence stations (STAs).

In this chapter, a novel power-efficient MAC protocol, named on-demand polling (ODP) scheme, is proposed for 802.11e HCF. In the proposed ODP scheme, a superframe of WLAN is divided into controlled access phase (CAP) and contention phase (CP), in which the voice services are transmitted at CAP and data services at CP. The operation of the ODP scheme is designed according to the properties of two-way voice conversation which can be mimicked by a four-state Brady model [5]. In the ODP scheme, a voice QSTA is allowed to transmit/received only when it is polled by a QAP. The QAP keeps a polling list to maintain the state of each voice QSTA. A voice QSTA will be temporarily removed from the list when it is silence. The voice QSTA may uses EDCF to request the QAP to include it to the polling list again when it starts its talkspurt. Also, the performance of the ODP scheme is evaluated in terms of power efficiency and data throughput. The power consumption of voice QSTAs in the ODP scheme can be derived by analyzing the mean duration of CAP and mean contention time of voice QSTAs. The throughput of the data QSTA can be further obtained by applying two-dimensional Markov chain queuing analysis. The performance of the proposed ODP scheme is compared with the method proposed in [4]. Simulation results show that the ODP scheme could reduce the power consumption for voice terminals and enhance the throughput for data terminals.

This chapter is organized as follows. In Section 3.2 HCF mode and Brady voice model are introduced. In Section 3.3 the concept of the proposed ODP scheme is described. The analyses of the ODP scheme are showed in Section 3.4. The performance evaluation of the ODP scheme by various simulation results are presented in Section 3.5. Finally, concluding remarks are given in Section 3.6.

4.2 System Overview

We assumed that all of QAP and QSTAs is operated in HCF mode for their QoS data transmission. HCF allows exchanging frames during both the contention period (CP) and contention free period (CFP) In the CP, the EDCF mechanism is employed for the contention-based access. The HCF-controlled channel access mechanism provides a polling-based access in the CFP.

The behavior of each voice conversation pair in the coverage of a QAP is assumed to be independent and identically distributed. The property of each conversation pair is modeled by a four-state Brady model, which consists of *double-talk* state, *mutual-silence* state, *downlink-only* state, and *uplink-only* state. The double-talk state indicates that uplink and downlink are both talking; mutual-silence state indicates that uplink and downlink are both silent; uplink-only state indicates that uplink is talking and downlink is silent; and, downlink-only state indicates that uplink is silent and downlink is talking. In Brady model, the sojourn time of each state is assumed to be exponentially distributed. The state transition rates, λ_{ij} , can be obtained by fitting the mean duration of talkspurt, pause, double talk, and mutual silence, given in [5].

4.3 The On-Demand Polling Scheme

In the ODP scheme, non-real-time data services are only transmitted by EDCF. On the other hand, based on the ON/OFF property of voice model, voice packets are transmitted by HCF controlled channel access or EDCF under different conditions. After accepting a new voice call, the QAP would add the QSTA in polling list. Then, the QAP will periodically poll QSTAs according to the list and wait for transmission of uplink voice packets. In order to enhance transmission efficiency, the QSTAs entering the silence period will be removed from the polling list. When the QSTAs are initiating a talkspurt, they will use higher access priority in EDCF to send voice packets for joining the polling list.

The period of a superframe in the ODP scheme is identical to the time interval of two successive voice packets generated by a QSTA. The superframe is divided into two periods: controlled access phase (CAP) and contention phase (CP) as shown in Fig. 1. In CAP, QAP polls QSTAs to get uplink voice packets or transmits downlink voice packets directly without acknowledge based on HCF controlled channel access. In CP, the voice QSTAs which are initiating a talkspurt and data QSTAs with packets in queue contend the channel based on EDCF with different priorities. The CAP within a superframe are further divided into three transmission periods: bi-direction voice transmission, uplink-only voice transmission (UL-only), and downlink-only voice transmission (DL-only). An active voice QSTA will be polled one of the three periods according to their current states.

For the bi-direction voice transmission period, the QAP combines the QoS(+)CF-Poll frame and the downlink voice packet into a single data frame by the QAP. After receiving the frame, the QSTA transmits an uplink voice frame to the QAP. For the UL-only voice transmission period, the QAP sends a sole QoS(+)CF-Poll frame to the QSTA and then waits for a uplink voice packet. For DL-only voice transmission period, the QAP consecutively sends remaining downlink voice packets without acknowledgement response from the corresponding QSTAs. Note that, during bi-direction voice transmission and UL-only voice transmission periods, the QSTA will be regarded as entering the silence period and removed from polling list if two consecutive QoS Null frames are received by QAP. After the CAP, the remaining time of a superframe is allocated for the CP which is shared by EDCF.

Moreover, QAP should assign different$AIFS$,$CW_{\min}$, and $CW_{\max}$for voice and data stations. For the guarantee of access delay, the AIFS of a voice QSTA, $AIFS_{RT}$, is set to be $PIFS$. And the$CW_{\min}$, denoted as $CW_{\min\_RT}$,and $CW_{\max}$, denoted as $CW_{\max\_RT}$, are given by

$$\mathrm{CW}_{\mathrm{min\_RT}}=\mathrm{CW}_{\mathrm{max\_RT}}=\max(2,\lceil E[N_V]+1\rceil), \tag{4.1}$$

where N_V is number of voice QSTA contending in a superframe and $\lceil . \rceil$ is a ceiling function. For data QSTAs, in case of contending with uplink voice QSTAs, the$AIFS$, denoted as$\mathrm{AIFS}_{\mathrm{NRT}}$, is given by

$$\mathrm{AIFS}_{\mathrm{NRT}}=\mathrm{PIFS}+\mathrm{CW}_{\mathrm{min\_RT}}. \tag{4.2}$$

The minimum and maximum contention window of data QSTA, $CW_{\min\_NRT}$ and $CW_{\max\_NRT}$, could be variable values set by QAP. Here, all the data QSTAs are with the same $CW_{\min\_RT}$ and set to beW_0. Let M denote the maximum backoff stage and Mbe the value such that $\mathrm{CW}_{\mathrm{max\_NRT}}=2^{\mathrm{M}}\cdot W_0$

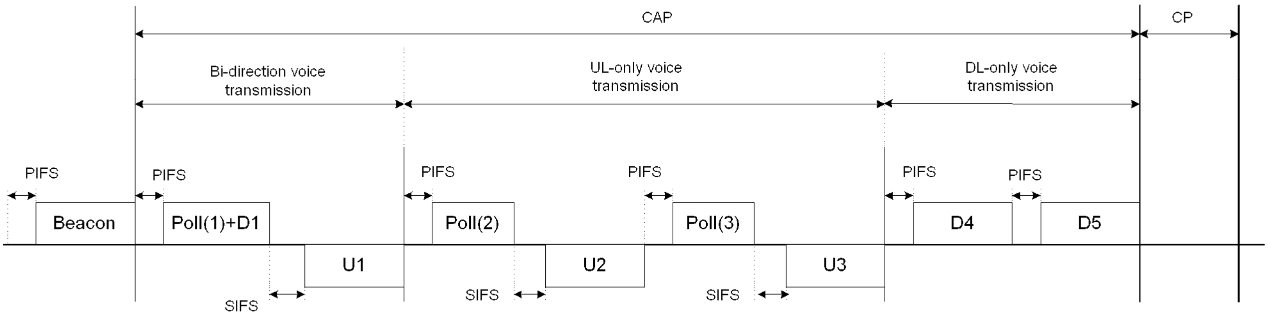

Figure 4.1: An example of the timing diagram of ODP scheme

4.4 Performance Analysis

In the analysis, N_V real-time voice QSTAs and N_D non-real-time data QSTAs are considered in the coverage of a QAP. An ideal channel condition without hidden terminals and with error-free transmission is also assumed. The mean duration of CAP in a superframe and mean contention time for a voice QSTA are analyzed in IV.A and IV.B, respectively. Using these two analytic results, the power consumption of a voice QSTA in the ODP scheme is directly derived in IV.C. Then, the frame transmission probability and aggregate throughput of data QSTAs are analyzed in IV.D.

4.4.1 Mean duration of CAP

The four-state Brady model is assumed to be symmetrical in the sense that the statistical behaviors of downlink-only state and uplink-only state are the same. By balance equation, we can obtain the state probability of a single communication pair staying in mutual-silence state (state A), downlink-only state (state B), uplink-only state (state C), and double-talk state (state D), denoted by p_a, p_b, p_c, and p_d, respectively, which yields

$$(\lambda_{ab} + \lambda_{ac})p_a = \lambda_{ba}p_b + \lambda_{ca}p_c, \tag{4.3}$$

$$(\lambda_{ba} + \lambda_{bd})p_b = \lambda_{ab}p_a + \lambda_{db}p_d, \tag{4.4}$$

$$(\lambda_{ca} + \lambda_{cd})p_c = \lambda_{dc}p_d + \lambda_{ac}p_a, \tag{4.5}$$

$$(\lambda_{db} + \lambda_{dc})p_d = \lambda_{bd}p_b + \lambda_{cd}p_c. \tag{4.6}$$

Given normalization condition for stationary probabilities,

$$p_a + p_b + p_c + p_d = 1, \tag{4.7}$$

the state probabilities are expressed as

$$p_a = \frac{\lambda_{ba} \cdot \lambda_{db}}{\lambda_{ba} \cdot \lambda_{db} + 2\lambda_{ab} \cdot \lambda_{db} + \lambda_{ab} \cdot \lambda_{bd}}, \tag{4.8}$$

$$p_b = p_c = \frac{\lambda_{ab} \cdot \lambda_{db}}{\lambda_{ba} \cdot \lambda_{db} + 2\lambda_{ab} \cdot \lambda_{db} + \lambda_{ab} \cdot \lambda_{bd}}, \tag{4.9}$$

$$p_d = \frac{\lambda_{ab} \cdot \lambda_{bd}}{\lambda_{ba} \cdot \lambda_{db} + 2\lambda_{ab} \cdot \lambda_{db} + \lambda_{ab} \cdot \lambda_{bd}}. \tag{4.10}$$

Since the process of each conversation pair is independent and identical, the probability of the integrated source of the conversation pairs staying in the four states, denoted by N_a, N_b, N_c, and N_d, respectively, can be obtained by using multinomial distribution

$$\begin{aligned} &P[(N_a, N_b, N_c, N_d)] \\ &= \tfrac{N_V!}{N_a!N_b!N_c!N_d!} p_a^{N_a} p_b^{N_b} p_c^{N_c} p_d^{N_d}, \end{aligned} \tag{4.11}$$

where $N_a+N_b+N_c+N_d = N_V$. Also, the time durations for packet exchange in the four states, denoted byT_a,T_b,T_c, and T_d, are calculated by

$$T_a = 0 \tag{4.12}$$

$$T_b = T_{PIFS} + (T_{PHY} + \frac{H_{MAC} + H_{UP} + Rs \cdot T_{SF}}{Rc}) \tag{4.13}$$

$$T_c = T_{SIFS} + (T_{PHY} + \frac{H_{MAC}}{Rc}) + T_b \tag{4.14}$$

$$T_d = T_{PIFS} + T_{SIFS} + 2(T_{PHY} + \frac{H_{MAC} + H_{UP} + Rs \cdot T_{SF}}{Rc}) \tag{4.15}$$

From (4.12) to (4.14), the system parameters are listed in Table I. Denote $E[T_{CAP}]$ the mean duration of CAP in a superframe. By the properties of multinomial distribution, the mean duration of CAP is expressed by

$$\begin{aligned} E[T_{CAP}] &= \sum_{i\in(a,b,c,d)} E[N_i] \cdot T_i \\ &= N_V \cdot \sum_{i\in(a,b,c,d)} p_i \cdot T_i \end{aligned} . \tag{4.16}$$

4.4.2 Mean Contention Time of Voice QSTA

During a contention period, due to the behavior of intermittent transmission, the mean number of contending voice QSTAs is smaller than 2 in our analysis. Thus, $CW_{\min\_RT} = CW_{\max\_RT}$=2. Firstly, we derive the successful contention probability during one contention period provided that there are N_C contending voice QSTAs, denoted by $P_s(N_C)$.

Denote N_1 as the number of contending QSTAs with counter set to 1, and N_2 as the number of contending QSTAs with counter set to 2. The successful contention event during a contention period is just either $\{N_1 = 1\}$ or $\{N_2 = 1\}$. Equivalently, it can be written by

$$\begin{array}{l} P_S\left(N_C\right) = \Pr\left[\{N_1 = 1\} \cup \{N_2 = 1\} | N_1 + N_2 = N_C\right] \\ = \Pr\left[N_1 = 1 | N_C\right] + \\ \sum\limits_{\substack{k=0 \\ k \neq 1}}^{N_C} \Pr\left[\{N_2 = 1\} | N_2 = N_C - k\right] \cdot \Pr\left[N_1 = k | N_C\right], \end{array} \tag{4.17}$$

where $\Pr\left[\{N_2 = 1\} | N_2 = N_C - k\right] = \begin{cases} 1, & k = N_C - 1 \\ 0, & otherwise \end{cases}$.

For N_C=1, $P_S(N_C) = 1$; for N_C=2, $P_S(N_C) = 0.5$;, and for $N_C \geq 3$, $P_S\left(N_C\right) = N_C \cdot \left(\frac{1}{2}\right)^{N_C - 1}$. Since successful contention in either time slot 1 or time slot 2 are also equally probable for $\mathrm{N_C} \geq 3$ cases, the average time spent in one successful contention period is given by

$$\bar{T_S} = \begin{array}{l} L_S + \frac{3}{2}\sigma \mathrm{for} \mathrm{N_C} = 1 \\ 2L_S + 2\sigma \mathrm{for} \mathrm{N_C} = 2 \\ L_S + \frac{3}{2}\sigma \mathrm{for} \mathrm{N_C} \geq 3 \end{array} . \tag{4.18}$$

For the time spent in collision period given N_C contention users, we enumerate all the situations how the collision happens as:$(N_1 = N_C,\ N_2 = 0)$, $(N_1 = 0,\ N_2 = N_C)$, and $(N_1 \geq 2, N_2 \geq 2, N_1 + N_2 = N_C)$. The first two events are equally probable and can be expressed by

$$\begin{aligned}
&\mathrm{P}_{1,\mathrm{coll}} \\
&{=}\mathrm{Pr}\{\mathrm{allcountersaresetto1}|\mathrm{failurecontentionperiod}\} \\
&= \Pr\{N_1 = N_C|N_C,\ failure\} = \left(\tfrac{1}{2}\right)^{N_C}\Big/(1 - P_S\left(N_C\right)) \qquad (4.19) \\
&\mathrm{orequivalently,} \\
&= \Pr\{N_2 = N_C|N_C,\ failure\} = \left(\tfrac{1}{2}\right)^{N_C}\Big/(1 - P_S\left(N_C\right))\cdot
\end{aligned}$$

For $N_C \geq 2$,

$$\begin{aligned}
&\mathrm{P}_{2,\mathrm{coll}} \\
&{=}\mathrm{Pr}\{\mathrm{collisionsoccurinboth}|\mathrm{failurecontentionperiod}\} \\
&= \Pr\{N_1 \geq 2 \cap N_2 \geq 2|N_C,\ failure\} \\
&= \left(1 - P_S\left(N_C\right) - 2P_{1,coll}\left(N_C\right)\right)\!\Big/\!\left(1 - P_S\left(N_C\right)\right)\cdot
\end{aligned} \qquad (4.20)$$

The average time spent in one contention period that collision occurs is given by

$$\bar{T}_C\left(N_C\right) = 2L_C + 2\sigma - \frac{\left(\frac{1}{2}\right)^{N_C}}{1 - P_S\left(N_C\right)}\left(2L_C + \sigma\right). \qquad (4.21)$$

Define random variable K_{N_c} as the number of contention periods such that first one of N_C users successfully contends. We further define T_{CT} as the random variable for the contention time. In general, the average contention time can be recursively given by

$$\begin{aligned}
&E[T_{CT}|N_C] \\
&= \sum_{k=1}^{\infty}\left[(k-1)\,\bar{T}_C\left(N_C\right) + \bar{T}_S\right]\cdot\ \Pr\{K_{N_c} = k\} + E[T_{CT}|(N_C - 1)] \\
&= \bar{T}_S\left(N_C\right) + \bar{T}_C\left(N_C\right)\cdot\tfrac{1 - P_S(N_C)}{P_S(N_C)} + E[T_{CT}|(N_C - 1)].
\end{aligned} \qquad (4.22)$$

Then, we can get the approximation form as follow

$$\begin{aligned}
&E[T_{CT}|N_C] = E[T_{CT}|(N_C - 1)] \\
&+\bar{T}_C\left(N_C\right)\cdot\tfrac{1 - P_S(N_C)}{P_S(N_C)},\mathrm{forN_C} \geq 3
\end{aligned}\ \cdot \qquad (4.23)$$

Also, for $N_C = 1$, $E[T_{CT}|N_C = 1] = \frac{3}{2}\sigma + L_S$, and, for $N_C = 2$, $E[T_{CT}|N_C = 2] = 2L_S + L_C + \frac{7}{2}\sigma$.

Finally, the mean duration of contention time, $E[T_{CT}]$, can be written as follow

$$
\begin{aligned}
&E[T_{CT}] = \\
&\sum_{(N_a+N_b)=0}^{N_V} \sum_{N_C=0}^{N_a+N_b} E[T_{CT}|N_C]P[N_C|N_a, N_b]P[N_a, N_b] \cdot
\end{aligned}
\tag{4.24}
$$

4.4.3 Power Consumption

Here, the power consumption of a voice QSTA is defined as the ratio of power-on period over superframe duration. For the ODP scheme, the power consumption factor η_{ODP} is equivalently expressed by

$$\eta_{ODP} = \frac{E[T_{CAP}] + E[T_{CT}]}{N_V \times T_{SF}} \tag{4.25}$$

4.4.4 Aggregate Throughput of Data QSTA

To calculate the aggregate throughput, the frame transmission probability,τ, is obtained firstly. The frame transmission probability here is derived by two-dimensional Markov chain queuing model. After some derivations, the probability can be expressed by

$$
\begin{aligned}
\tau &= \frac{2 \times \sum_{i=0}^{M} P^i}{\sum_{i=0}^{M} P^i \times (2^i * W_0 + 1)} \\
&= \frac{2(1 - 2P)}{W_0(1 - P)(1 - (2p)^{M+1})/(1 - P^{M+1}) + (1 - 2P)},
\end{aligned}
\tag{4.26}
$$

where P is the collision probability of data QSTAs. Then, we can derive the P based on its definition. P is the probability that a ready-to-transmit QSTA collides with any other QSTAs, which yields

$$P = 1 - (1 - \tau)^{N_D - 1}. \qquad (4.27)$$

Eq. (4.26) and (4.27) represent a nonlinear system in the two unknowns τ and P, which can be solved using numerical method.

DefineP_{idle} the probability that none of data QSTAs transmit in a randomly chosen time slot. P_{idle} is given by Also, denote P_{one} the probability that only one data QSTA transmits in a randomly chosen time slot. P_{one} is given by Then, the throughput of a single data QSTA, S', can be obtained by

$$\begin{aligned} S' &= \frac{\text{E[Payload Information in a slot time]}}{\text{E[length of a slot time]}} \\ &= \frac{\tau(1-\tau)^{N_D-1}E[P]}{P_{idle}\cdot\sigma + P_{one}{\cdot}T_S + (1 - P_{idle} - P_{one})T_C} \end{aligned} \qquad (4.28)$$

where T_S (T_C) is the average time that channel is sensed busy because of a successful (collided) transmission, and $E[L_{PKT}]$ is the average frame length. Since the transmitted frames of data QSTAs cannot preempt those of voice QSTAs, the aggregate throughput of data QSTAs, denote as S, can be written as

$$S = (\frac{T_{SF} - T_B - E[T_{CAP}] - E[T_{CT}]}{T_{SF}}){\cdot}S' \qquad (4.29)$$

where T_{SF} is maximum duration of the superframe, T_B is the time to send a Beacon. Due to the chapter limitation, the detailed derivations of frame

Table 4.1: SYSTEM PARAMETERS

Parameter	Symbol	Value
Duration of the superframe	T_{SF}	20ms
Voice coding rate in bps	R_S	8K
Transmission rate in bit/sec	R_C	11M
MAC header (QoS data type) in bits	H_{MAC}	30*8
Header overhead (IP+UDP+RTP) in bits	H_{UP}	40*8
Physical overhead in sec (including preamble length and header length)	T_{PHY}	192μs
Beacon size in bit	B	40*8
SIFS	T_{SIFS}	10μs
PIFS	T_{PIFS}	30μs
Slot time	σ	20μs
Time to send a beacon	T_B	*Computed*
Time to successfully transmit a contention packet	L_S	*Computed*
Time to transmit a collided packet	L_C	*Computed*
Time to send an ACK frame (14bytes)	T_{ACK}	*Computed*

transmission probability and aggregate throughput are not shown here. Please refer to our previous work [6].

4.5 Simulation Result

The parameters used for the following numerical analysis and simulations are summarized in Table 1, where the values of PHY-related parameters are referred to IEEE 802.11b [1]. Unless otherwise specified, a constant frame payload size of 1028 bytes, which includes 1000 bytes application data payload, 20 bytes IP header, and 8 bytes UDP header, is used in the simulations. The maximum backoff stage M was set to be equal to 5. The power consumption of voice QSTAs and the aggregate throughput for data QSTAs are investigated by using the proposed ODP scheme, a round robin mechanism with Brady's model (RR-Brady) and a round robin [2] mechanism with constant bit rate (RR-CBR) [2], respectively. The coding rate of CBR source model is 8Kbps. Since the voice QSTAs will not contend in CP within RR-Brady and RR-CBR schemes, $AIFS_{NRT}$ is set to be DIFS.

Fig. 2 shows the normalized power consumption versus the number of voice pairs. The normalized power consumption is defined as the percentage of a voice QSTA operating in active mode within a superframe. From the figure, it can be found that the power consumption of ODP scheme increases with the number of voice pairs, which is resulted from the increased mean contention time. However, the ODP scheme consumes the least power while the RR-CBR consumes the most. In the ODP scheme the voice QSTA will be removed from the polling list when there is no packet to transmit. It is beneficial to reduce the power consumption. When the number of voice pair is 25, the ODP scheme outperforms the RR-Brady and RR-CBR schemes by an amount of 12.5% and 32.4%, respectively.

Fig. 3 shows the aggregate throughput of data QSTAs versus the number of voice pairs. The number of data QSTA is 15 and the minimum contention window is 32. The number of voice QSTAs is ranging from 0 to 25. The curves of simulation results and numerical results of ODP are very close, which justifies the accuracy of our analysis. Also, comparing the three schemes, it can be found that the aggregate throughput of ODP scheme is the highest among the three schemes. The throughput enhancement increases as the number of voice QSTAs increases. When the number of voice QSTAs is 25, the ODP scheme outperforms the RR-CBR (RR-Brady) scheme by an amount of 90.6%

(56%). The aggregate throughput of ODP scheme is smaller than that of RR-Brady and RR-CBR schemes for the cases that accommodated voice pairs is smaller than two. The reason is that the $AIFS_{NRT}$ of RR-Brady and RR-CBR schemes is smaller than that of ODP, which contributes to a higher aggregate throughput.

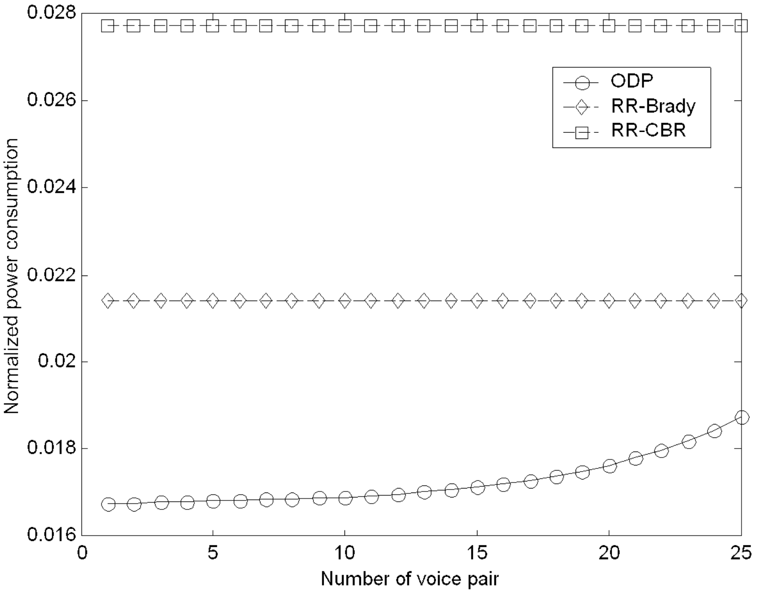

Figure 4.2: Normalized power consumption for voice QSTA

4.6 Concluding Remarks

We propose a power-efficient MAC protocol, named ODP, for WLAN supporting integrated voice and data service. In the ODP scheme, the voice QSTAs are polled by the QAP at the CAP while the data QSTAs contend for transmission at the CP. When a voice QSTA is back from silence mode, it uses EDCF to request for including itself to the polling list. Simulation results show the accuracy of our analysis and the performance of the ODP scheme. It is found that the ODP scheme consumes less power and achieves higher

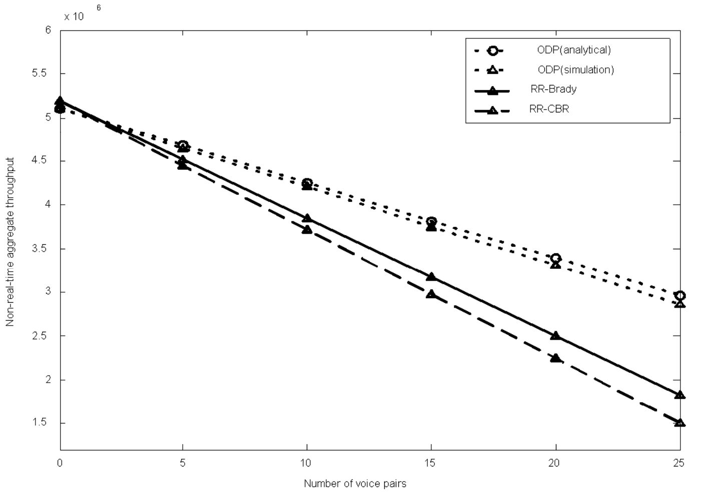

Figure 4.3: Aggregate throughput for data QSTAs

data throughput compared with the RR/Brady and RR/CBR schemes.

Reference

[1] IEEE Std 802.11e/D3.0, “Wireless medium access control (MAC) and physical layer (PHY) specifications: Medium access control (MAC) enhancements for quality of service (QoS),” May 2002.

[2] M. A. Visser and M. E. Zarki, “Voice and data transmission over an 802.11 wireless network,” *IEEE PIMRC*, Sept. 1995.

[3] B. P. Crow, I. Widjaja, J. G. Kim, P. Sakai, ”Investigate of IEEE 802.11 medium access control (MAC) sublayer functions,” *IEEE INFOCOM 1997*, vol.1, pp. 126.-133, 1997.

[4] M. Veeraraghavan, N. Cocker, T. Moors “Support of voice services in IEEE 802.11 wireless LANs,” *IEEE INFOCOM 2001*, vol. pp. 488-497, 2001.

[5] P. T. Brady, “A model for on-off speech patterns in two-way conversation,”*Bell Syst.Tech.J*, vol. 48, pp. 2445-2472, Jan. 1969.

[6] C. Y. Shih, R. G. Cheng, C. J. Chang, and Y. S. Chen, “Achieved weighted fairness for wireless multimedia Services” *Proc. Of VTC-Fall 2003.*

Chapter 5

Concluding Remarks and Future Works

In the second year of the project, we study the the radio resource management techniques for the B3G networks. Extending the research results of the first year, we propose a utility-based scheduling algorithm with differentiated QoS provisioning, a situation-aware data access manager using fuzzy Q-learning technique (FQ-SDAM) for WCDMA systems, and a on-demand polling scheme for WLAN in chapter 2, 3, and 4, respectively.

In chapter 2, we propose a utility-based scheduling algorithm assuming three design aspects of radio resource efficiency, QoS requirement achievement, and weighted fairness for multimedia CDMA networks. The utility function of each connection is the radio resource function weighted by both the QoS requirement achievement function and weighted fairness function. The system resource is scheduled via maximizing the overall system utility while the weighted fairness and the QoS requirements are kept. The optimal scheduling solution is derived as a resource assignment vector for active connections by solving the optimization of overall utility function via a polymatroid structure. To investigate the performance of the proposed utility-based scheduling algorithm, we compare its average throughput with the *Exponential Rule* scheduling algorithm. Simulation results show that the utility-based scheduling scheme has higher throughput than that of exponential rule scheme when both dedicated and shared channels are used in the system. Therefore, the utility-based scheduling algorithm is effective for multimedia CDMA systems with diverse of QoS requirements and with dedicated and shared channels support. In our future work, more complicated simulation scenarios with variant service types will be carried out to further investigate the performance and the features of the utility-based scheduling algorithm.

Chapter 3 proposes a novel situation-aware data access manager FQ-SDAM for multi-cell WCDMA systems, which contains a fuzzy Q-learning-based residual capacity estimator (FQ-RCE) and a data rate scheduler (DRS). By applying the perceptual coordination method (PCM), the FQ-RCE treats the received home-cell interference and adjacent-cell interference as two separate linguistic variables such that it can adaptively determine the residual capacity according to the current loadings in the home cell and the radio resource allocation in the adjacent cells. For the DRS, a location-dependent scheduling algorithm, modified exponential rule, is also proposed. Simulation results show that, the proposed FQ-SDAM can effectively reduce the packet error probability. In our future work, we will continue to evaluate the aggregate data throughput. Also, the performance of FQ-SDMA under non-homogeneous will be evaluated.

In chapter 4, we propose a power-efficient MAC protocol, named ODP, for WLAN supporting integrated voice and data service. The main design concept of the ODP scheme is to exploit the nature of two-way voice communication; that is, the ODP scheme will adjust the polling list according to the communication state. In the ODP scheme, the voice QSTAs are polled by the QAP at the CAP while the data QSTAs contend for transmission at the CP. When a voice QSTA is back from silence mode, it uses EDCF to request for including itself to the polling list. Simulation results show the accuracy of our analysis and the performance of the ODP scheme. It is found that the ODP scheme consumes less power and achieves higher data throughput compared with the RR/Brady and RR/CBR schemes. In our future work, we will study the techniques for power saving by considering the effect of silence mode and other physical layer behaviors of the WLAN.

Perspectively, we will integrate all the developed RRM technologies for WCDMA and WLAN network in the first and second years of the research and apply them to the B3G network in the third year.