By Use of Frequency Diversity and High Priority in Wireless Packet Retransmissions

Xiaoyan Liu, and Huiling Zhu
School of Engineering and Digital Arts, University of Kent, Canterbury, Kent, CT2 7NT, United Kingdom
{xl36, h.zhu}@kent.ac.uk

Abstract – In orthogonal frequency division multiple access (OFDMA) systems, the delay of packets has a great impact on quality of service (QoS), especially for real-time transmission. The basic concept of resource allocation in an OFDMA system is to allocate a subcarrier to the user with the best channel condition in that subcarrier. However, when a retransmission technique is used, the packet delay of successfully transmitting failed packets may be quite large in conventional retransmission schemes. In this paper, the packet combining technique in the downlink OFDMA system is introduced in conjunction with frequency diversity to enhance the reliability of retransmissions and thus reduce the large time delay caused by retransmissions. The novel retransmission technique aims to reduce the maximum number of retransmissions for failed packets, while maximizing the throughput. Bit error rate (BER) and transmit power are constraints in the throughput maximization formulation. A suboptimal algorithm is proposed, in which the retransmitted packets have higher priorities to be given resources than new packets. At the receiver, failed and retransmitted packets are combined before detection by using maximal ratio combining (MRC). It is shown that the proposed retransmission scheme can reduce the maximum packet delay significantly, while maintaining the maximum throughput of the conventional retransmission schemes.

Index Terms **– Orthogonal frequency division multiple access (OFDMA), real-time transmission, resource allocation, retransmission techniques, maximal ratio combining (MRC).**

I. INTRODUCTION

Future wireless communication services need to be provided with high data rate as well as guaranteed quality of service (QoS) such as delay and packet error rate (PER). Orthogonal frequency division multiple access (OFDMA) is a promising technique achieving high data rates for the next generation wireless systems. Adaptive resource allocation, which allocates resources such as modulation, power and number of subcarriers etc., adaptively to different users according to channel conditions, can utilize the radio resource efficiently due to the time-varying nature of the wireless channel. The nature of multiple subcarriers in OFDMA system leads to the possibility of adaptively choosing which subcarriers to be used among users, and at what rate and power to transmit on each subcarrier in each time slot.

Unlike packets in non-real-time applications (e.g. file transfer), which is considered delay-tolerant, packets in real-time applications are required to be constantly and correctly received by an end user. In real-time services, packets are often generated at regular intervals and required to be received with a delay constraint. Each packet has a delay deadline by which it must reach its destination; otherwise, it will be discarded. For example, in a full- motion video application [1], 30 frames are generated every second, and each frame must be delivered to the destination within a time delay to avoid any observable jitter. Therefore, transmission delay is an important problem in high data rate wireless communications especially for real-time traffic. When retransmission techniques are used, the maximum number of retransmissions for failed packets determines transmission delay. As known, automatic repeat request (ARQ) schemes are effective to recover non-real-time data corrupted by channel errors. However, with the high data rate in modern wireless networks, ARQ schemes are also favoured in real-time traffic [2]. Due to the QoS requirement of maximum packet delay caused by retransmissions of failed packets, in this paper a novel retransmission mechanism is proposed to reduce the delay introduced by retransmissions by allocating resources to retransmitted packets with higher priority than new packets.

In conventional resource allocation [3]–[5] in the OFDMA, retransmission techniques have not been considered for real-time traffic. Even if all retransmitted packets are guaranteed to be allocated subcarriers not being the best subcarriers, there is still a possibility that the retransmitted packets are received incorrectly. Thus, the maximum transmission delay in traditional resource allocation is long. In this paper, effective resource allocation is investigated when retransmission techniques are considered. The aim is to reduce the maximum number of retransmissions, so that the maximum transmission delay will be reduced. In other words, a new optimization problem is formulated to reduce the maximum packet delay by giving resources with a priority to retransmitted packets for transmission while maximizing the throughput in the downlink OFDMA system. In order to achieve high retransmission reliability, low density parity check (LDPC) [6]–[7], an advanced channel coding, is adopted since the constituent codes in the LDPC have parity check relationships and extremely good performance. Packet combining technique using maximal ratio combining (MRC) has also been adopted. In addition, the retransmitted packet will be transmitted through a subchannel which is different from the failed transmission to achieve frequency diversity.

The rest of the paper is organized as follows. The next section represents system model and is followed by Section III about the description of problem formulation. In Section IV, a suboptimal solution is proposed. Simulation results for different retransmission schemes and comparison among different allocation algorithms are given in Section V. Finally, this paper is concluded in Section VI.

ISBN:978-1-61208-113-7

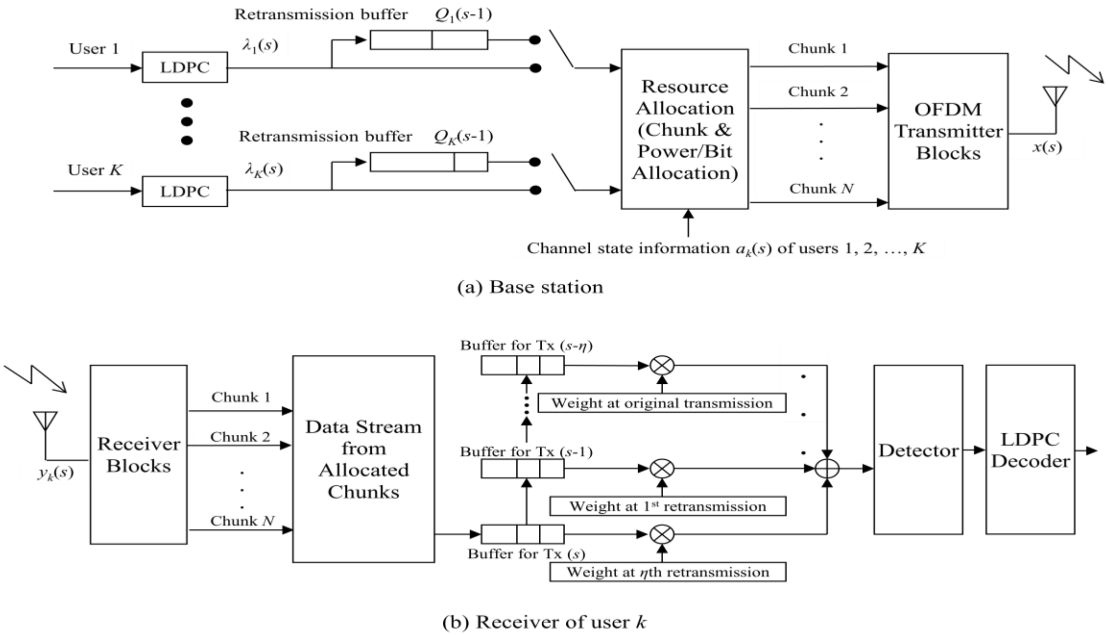

Fig. 1 Downlink multiuser OFDMA block diagram

II. SYSTEM MODEL

The downlink OFDMA system with N_{total} subcarriers and K users is shown in Fig. 1. In the base station (BS), resource is allocated periodically. An allocation period, denoted by τ, is equivalent to a downlink frame which includes several OFDM symbols. It is assumed that the channel condition is unchanged during one allocation period, i.e. one frame, so that the channel condition is the same for all the τ/T_s OFDM symbols in the frame, where T_s is the symbol duration and τ/T_s is assumed to be an integer. The time interval $(s-1, s]\tau$, where s is an integer, denotes the sth allocation period.

At the receiver of user k, the received signal in period s is expressed as

$$y_k(s) = \alpha_k(s) x_k(s) + v_k(s), \quad k = 1, 2, \cdots, K \tag{1}$$

where $\alpha_k(s)$, called channel fading factor, is the magnitude of the frequency response of the channel fading of user k, which is independent from that of other users, $x_k(s)$ is the transmitted signal of user k, and $v_k(s)$ is a zero-mean additive white Gaussian noise (AWGN) of user k with bandwidth B and double-sided power spectral density $N_0/2$.

II A.Transmitter

The dynamic channel allocation is carried out on a packet-by-packet basis. As shown in Fig. 1, when a packet is ready for downlink transmission, it is numbered and stored in the input queue of the transmitter. After its transmission, the packet is saved in the retransmission buffer of the transmitter. When a positive acknowledgement (ACK) for a packet in the retransmission buffer is received, the packet is released from the retransmission buffer. If the receiver detects any error and feedbacks a negative acknowledgement (NAK), the transmitter will resend the failed packet. Any new packets from all users are encoded by the LDPC encoders, are allocated subcarriers and power, and transmitted together with packets to be retransmitted. The process is repeated until the packets are correctly received.

In the BS, the traffic source is considered greedy, which means that the system transmits as many packets as it can and there are always packets waiting to be transmitted. The BS allocates subcarriers and power to packets from all users according to their channel qualities, which may vary from frame to frame.

With the objective of minimizing the delay of retransmitted packets, all the retransmitted packets in queue, which are received incorrectly, will be guaranteed to be sent out during each allocation period (i.e. a frame). If a newly arrived packet is allocated with resources for its transmission, the packet will be encoded by the LDPC encoder before being sent to the transmitter block. By denoting the number of retransmitted packets failed in the $(s-1)$th period from user k as $Q_k(s-1)$, the number of newly arrived packets to be transmitted in the sth period from user k as $\lambda_k(s)$ and the number of acknowledged packets from user k in the sth period as $q_k(s)$, at the end of the sth period, the number of failed packets to be retransmitted for user k in the next transmission is given by

$$Q_k(s) = Q_k(s-1) + \lambda_k(s) - q_k(s), \quad k = 1, 2, \cdots, K. \tag{2}$$

1) Subchannel Grouping

In this paper, the chunk allocation method[1] is used in allocating resources, where the number of bits and power are assumed to be the same for all the subcarriers within one chunk. In each allocation period, if a chunk is allocated to a

[1] In order to reduce the overhead and complexity of subcarrier allocation, the correlation between adjacent subchannels in the OFDMA system can be considered. Since the condition of a subchannel and its adjacent subchannels are quite similar, properly grouping a set of contiguous OFDM subchannels into one chunk and allocating the downlink resource chunk by chunk to users can make the subcarrier allocation simple while approaching the downlink throughput close to that of subcarrier-based allocation which allocates spectrum to users subcarrier by subcarrier [8].

 ISBN:978-1-61208-113-7

user to transmit its newly arrived data, channel coding is exploited across the subcarriers in each chunk and the length of the new packet to be transmitted is adapted in order to be filled into one entire chunk. It is assumed that every m neighbouring subcarriers are grouped into one chunk. By grouping all subcarriers into N chunks, the computational complexity of resource allocation can be reduced approximately by $m = N_{total}/N$ times. To maintain transmission reliability while achieving high system throughput, the bit error rate (BER) constraint is given. Adaptive multi-level quadrature amplitude modulation (M−QAM) is adopted under this BER constraint. The adaptive modulation level M takes values from the set $\mathbf{M}$ = {0, 4, 16, 64, 256} and the corresponding data rate (bits per symbol) b takes values from the set $\mathbf{b}$ = {0, 2, 4, 6, 8}. If the allocated bits of user k on each subcarrier over chunk n is $b_{k,n}$, the total bits for user k transmitted on this chunk is $mb_{k,n}$, and the bits of the new packet equals

$$L_{k,n} = mb_{k,n} \frac{\tau}{T_s}. \qquad (3)$$

On the other hand, for a failed packet, the retransmitted signal will be combined at the receiver with that in the first transmission to achieve frequency diversity. In order to explore the frequency diversity, the retransmitted packet will be allocated by a chunk which is different from the previous one for transmission, and the bits of the retransmitted packet is the same as the original transmitted packet. Therefore, the bits allocated for retransmitted packets are set by the bits of the original transmission, and accordingly the power is allocated adaptively for the retransmission to guarantee the bits allocation.

2) Rate-Adaptive LDPC-Coded Modulation

In this system, similar to [9]−[12], the traffic source is encoded with $|\mathbf{M}|$ LDPC codes, where $|\mathbf{M}|$ denotes the size of $\mathbf{M}$, followed by Gray code mappings to $|\mathbf{M}|$ M−QAM constellations. When the M−QAM constellation is derived by the resource allocation in a frame, an LDPC code is constructed such that the bits of all coded packets can be fitted into τ/T_s OFDM symbols in the corresponding constellation. The length is calculated according to (3). The encoder and decoder hence have a given set of $|\mathbf{M}|$ LDPC generator matrices and corresponding parity check matrices, respectively. If chunk n is allocated to user k with code rate $R_{k,n}$, a source vector consisting of $R_{k,n}L_{k,n}$ information bits and $(1-R_{k,n})L_{k,n}$ parity check bits are encoded into the binary codeword of size $L_{k,n}$. The number of parity check bits to be transmitted in one of OFDM symbols in the frame is then equal to $(1-R_{k,n})mb_{k,n}$, whereas the number of information bits in each OFDM symbol is $R_{k,n}mb_{k,n}$. The number of information bits in all subchannels, which equals the system throughput (bits per OFDM symbol), is $R_{k,n}mb_{k,n}N$. This is carried out by letting the QAM modulator map to a transmitted signal vector of length τ/T_s, by taking τ/T_s distinct sets of $mb_{k,n}$ bits and mapping each such set into a complex OFDM symbol.

As greedy traffic is assumed, one packet occupies one chunk for its transmission and from (3), the packet length is determined by bits allocated to that chunk. In another word, the number of bits carried on the chunk is determined by the modulation level employed for that chunk. For simple implementation, chase combining technique [13] is used, in which exactly the same information bits and parity bits are contained in every retransmission. Because all transmissions are identical, chase combining can be seen as additional repetition coding. One retransmitted packet is only allowed to be transmitted over one chunk rather than over many. Thus, in order to fill a retransmitted packet within a chunk, the number of bits on the chunk is predetermined by the failed packet's length or its previous modulation level, only if it is known that this chunk is assigned to retransmitted packets. Except for chunks allocated to retransmitted packets, the sizes of the other chunks are still determined by their bits allocation as in (3). In another word, same modulation level is employed for each retransmitted packet as in its previous transmission since they have the same packet length.

II B.Receiver

The modulated M−QAM symbol vector is then transmitted through a slowly varying flat fading channel. Such a vector will then be subjected to (almost) constant fading and the added channel noise v. As shown in Fig. 1, the data stream from the allocated chunks of user k is split through a chunk selection block. The demodulator uses maximum likelihood soft decision detection on the received data stream, while the subsequent decoder uses an iterative belief propagation decoding algorithm [14].

In the recovered data stream, if a packet is received incorrectly, it will be stored in the receiver buffer, and this packet must be retransmitted immediately in the next frame to guarantee the delay constraint of real-time services. Along with the previous received incorrect packet stored in the receiver buffer, the retransmitted signals could be combined by implementing MRC. One major issue is that sufficient chunks are assumed to guarantee the transmission of failed packets, and retransmitted packets may be transmitted in different chunks from previous transmission, which depends on the chunk condition. In our proposed allocation scheme, failed packets have higher priority to be allocated the best chunks of their users. By implementing this sort of frequency diversity technique, high transmission success could be achieved.

The buffers in the receiver are used to save failed packets together with the channel state information from the user. The received signals from different transmission slots of the same packet are saved separately in buffers for the corresponding transmission slot, as in Fig. 1. Suppose a packet is received correctly after η retransmissions, then at the receiver side η buffers are used to save this packet from allocated chunks of user k in transmission slot $s-\eta$, $s-1$, …, and s. Along with the weights, proportional to channel state information, of these transmissions, the receiver uses the MRC to combine the

 ISBN:978-1-61208-113-7

received signal with the signals received from previous transmissions. The detector is the demodulator with log-likelihood ratio (LLR) output and is followed by a LDPC decoder with hard decision, where the number of iterations to be performed for decoding one codeword is given.

III. PROBLEM FORMULATION FOR RESOURCE ALLOCATION

Here, optimizing the chunk and power allocation is considered under the power constraint with rate discretization, so as to reduce the maximum packet delay caused by retransmission while maximizing the throughput of new packets of all users. The retransmitted packets and new packets are treated differently in resource allocation by taking into account of the trade-off between maximizing system throughput and reducing packet delay. The optimal chunk assignment is considered for a given set of total available chunks defined as $\mathbf{I} = \{1, 2, \cdots, N\}$. The current allocation period index s is omitted for some variables in the following for simple notation. The optimization problem can be mathematically formulated as

$$\max_{\{\Omega_k, p_{k,n}\}} \sum_{k=1}^{K} \sum_{n \in \Omega_k} b_{k,n} \tag{4}$$

subject to
$$\sum_{k=1}^{K} \sum_{n \in \Omega_k} m p_{k,n} \le P_T - \sum_{k=1}^{K} \sum_{n' \in \Omega'_k} m p_{k,n'} \tag{4.1}$$

$$p_{k,n} \ge 0 \text{ for all } k, n \tag{4.2}$$

$$b_{k,n'} = b_{k,n}(s-1), n' \in \mathbf{\Omega}'_k, n \in \mathbf{A}(s-1) \text{ for all } k \tag{4.3}$$

$$\left(\bigcup_{k=1}^{K} \mathbf{\Omega}_k \right) \cup \left(\bigcup_{k=1}^{K} \mathbf{\Omega}'_k \right) \subseteq \mathbf{I} \tag{4.4}$$

$$|\mathbf{\Omega}'_k| = Q_k(s-1) \text{ for all } k \tag{4.5}$$

where the prime, ('), denotes notation on retransmissions. $p_{k,n}$ is the power allocated to user k on each subcarrier in chunk n, and it is constant across the subcarriers in each chunk. Constraints (4.1) and (4.2) guarantee that allocated power on all chunks is non-negative and the total power does not exceed the total power constraint P_T. $\mathbf{\Omega}_k$ and $\mathbf{\Omega}'_k$ are the sets of chunks assigned to user k for its packets on first-time transmissions and retransmissions, respectively. In constraint (4.3), the assigned transmit bits per symbol of allocating one retransmitted packet of user k, equals its previous allocated bits $b_{k,n}(s-1)$, which is an integer, if the allocated chunk in previous allocation period is $n \in \mathbf{A}(s-1)$, where $\mathbf{A}(s-1)$ is the set of previous chunk allocation.[2] Its assigned transmit bits per symbol in this allocation period is $b_{k,n' \in \mathbf{\Omega}'_k}$. (4.3) also shows that the modulation level of the retransmitted packet must be the same as that in the previous transmission in order to combine retransmitted packet with previous failed transmitted packet(s) to achieve frequency diversity. Both new and retransmitted packets for all users are allocated by chunks from the set of total available chunks, $\mathbf{I}$, as in (4.4). (4.5) assures that all retransmitted packets $Q_k(s-1)$ can be sent out by assigning sufficient number of chunks, $|\mathbf{\Omega}'_k|$ is the number of chunks for the retransmitted packets of user k.

Assuming that the retransmitted packets from every user have high priority to be transmitted, they are allocated sufficient chunks within the allocation period in order to guarantee the transmission of all the retransmitted packets. From (4.1), the remaining power from the consumption by retransmitted packets is allocated to new packets with the objective of maximizing the throughput. Thus, the optimization problem in (4) can be split into 2 steps. For the retransmitted packets of all users, since the bits to be transmitted are given, the problem turns to minimize the overall power consumption on retransmission.

After solving the above problem with all these constraints, a set of chunks occupied by retransmitted packets $\mathbf{\Omega}'_k$ for all k, $\mathbf{\Omega}'_k = \{n' | \rho'_{k,n'} = 1\}$ can be obtained. The set of chunks available for new packets $\mathbf{\Omega}_k$ is thus obtained as the rest of $\mathbf{I}$. By assuming all new packets have the same BER requirement, $b_{k,n \in \mathbf{\Omega}_k}$ could be chosen for any user k according to the required received signal to noise ratio (SNR) and the SNR gap Γ [9], which is a constant related to this given BER requirement if the same modulation scheme is used.[3] The value of Γ depends on the capacity difference between practical system using adaptive modulation and Shannon capacity (Γ=1 (0dB)). $b_{k,n}$ can be expressed approximately as

$$b_{k,n} = \log_2 \left(1 + \frac{p_{k,n} |\alpha_{k,n}|^2}{\Gamma N_0 B / N} \right), \text{ integer for all } k, n \tag{5}$$

where $\alpha_{k,n}$ is the channel fading factor for user k in chunk n. $b_{k,n}$ can be determined through the received SNR region given in Table I, conditioned on different values of BER constraint.

IV. PROPOSED ALLOCATION ALGORITHM

Firstly, let A_n, b_n and p_n denote the user index, number of bits and power allocated on chunk n, respectively. $H_{k,n} = |\alpha_{k,n}|^2 / N_0(B/N)$ is defined as the channel to noise ratio for user k in chunk n. With the bits allocation $\{b_n(s-1)\}$ and chunk allocation $\{A_n(s-1)\}$ of retransmitted packets in previous resource allocation $n \in \mathbf{A}(s-1)$, which are registered in two separate registers embedded in the BS's resource allocation block. In order to guarantee the success of retransmitted packets, each packet on retransmission has higher priority to choose its best chunk. The proposed allocation determines the number of bits and chooses 'better' chunks, which has higher channel to noise ratio, for retransmitted packets according to the algorithm shown below. Given the BER constraint, the number of bits $\{b_{n'}\}$ and chunk conditions $\{\alpha_{n'}\}$on allocated

[2] The value of each element A_n in $\mathbf{A}$ is the user index on subcarrier n.

[3] If no coding is considered, for M−QAM $\Gamma = -\ln(5BER)/1.6$, where BER is the BER requirement [11].

chunks for retransmitted packets, the power distribution $\{p_{n'}\}$ thus can be obtained from the transformation of (5). The rest of available chunks will be allocated to new packets according to the *Maximum-Capacity* allocation algorithm which is explained in the following paragraph.

In order to maximize system throughput for new packets, multiuser diversity is explored by allocating chunks to the user with the best channel condition on that chunk. In this algorithm, the transmit power is firstly assumed to be the same for subcarriers in one chunk and then an optimal power allocation algorithm is carried out among chunks to maximize throughput. The optimal transmit power adaptation is the well-known water-filling procedure [15]. In water-filling, more power is allocated to "better" chunks with higher SNR, so as to maximize the sum of data rates in all chunks.

The new packets are firstly allocated with chunks within the rest of available chunks according to their channel to noise ratio. The initial power distribution is then obtained by water-filling algorithm, so that corresponding initial bit rate can be calculated according to (5), which can be a continuous value. Because of the integer nature of bits per subcarrier per symbol, data rate calibration (as described in step 3-c below) has to be carried out to fit the existing modulation scheme. Finally, the transmit power allocated over its assigned chunk is recalculated accordingly. The algorithm can be described as:

1) *Initialization*
   a) Set $b_n = 0$, $A_n = 0$, $p_n = 0$ for $n = 1, 2, \cdots; N$.
2) *For Retransmitted Packets:*
   for i = 1 to number of retransmitted packets
      a) find n' satisfying $|H_{k,n'}| \geq |H_{k,n}|$ for all n;
      b) let $b_{n'} = b_n(s-1)$, $A_{n'} = A_n(s-1)$;
      c) calculate $p_{n'}$ according to (5).
   end
3) *For New Packets on* $n \in \{A_n = 0\}$*:*
   for i = 1 to N – number of retransmitted packets
      a) find k' satisfying $|H_{k',n}| \geq |H_{k,n}|$ for all k;
      b) let $A_n = k'$, use water-filling algorithm to obtain $p_{k',n}$ among the rest power $P_T - \Sigma\, p_{n'}$;
      c) calculate $b_{k',n}$ according to (5), let $b_n = \lfloor b_{k',n} \rfloor_{\mathbf{M}}$,
      where $\lfloor x \rfloor_{\mathbf{M}}$ denotes the largest number in the set of **M** less than or equal to x.
      d) calculate p_n according to (5).
   end

Then, $\mathbf{A} = \{A_{n'}\} \cup \{A_n\}$ is the chunk allocation, $\mathbf{B} = \{b_{n'}\} \cup \{b_n\}$ is the bit allocation, $\mathbf{P} = \{p_{n'}\} \cup \{p_n\}$ is the power allocation.

V. SIMULATION RESULTS

In this section, the performance of the OFDMA system with the proposed resource allocation algorithm is investigated. It is assumed that an OFDMA system has 512 subcarriers (N_{total} = 512), the number of subcarriers per chunk is 16 (m = 16), the number of users is eight (K = 8) and symbol duration T_s = 20μs. A frequency-selective Rayleigh fading channel is assumed, which is a 17-path channel with exponential power delay profile and root mean square (RMS) delay spread of 0.5μs for each path. Doppler frequency is 1Hz. The channel fading factor $\{\alpha^2_{k,n}\}_{n=1,2,\ldots,N}$ of each chunk is Rayleigh distributed with $E\{|\alpha_{k,n}|^2\} = 1$. The BER constraint is 10^{-3} with 1/2 rate LDPC coding, where the parity check matrix is even in both rows and columns, length-four cycle is eliminated, and number of ones per column is 3. Greedy traffic source is assumed. That is, the traffic source of user k generates packets at the rate depending on allocated bits per symbol $\{b_{k,n}\}$; and each packet contains flexible length of information bits accordingly. Each frame consists of 6 OFDM symbols. The number of packets the system could allocate within one frame is N = 32, since each chunk carries only one packet. 100 frames are chosen for each simulation run.

Given the target BER, two allocation schemes are investigated by taking packet retransmissions into account. One is the conventional *Maximum-Capacity* scheme, by allocating each chunk to the user with the best chunk condition for that chunk. The other one is our proposed allocation scheme with packet combining, where all the packets on retransmissions are allocated with their best chunks, and assigned same modulation as in original transmission; new packets are allocated among the remaining chunks based on the conventional *Maximum-Capacity* allocation scheme. Packet combining is also adopted at the receiver to improve the performance. Each retransmitted packet is sent out immediately for both schemes as long as its user is allocated sufficient chunks, otherwise it will wait in the retransmission buffer until the next frame. The following metrics are used to evaluate the performance through the simulation:

- *Packet Error Rate*, defined as the total number of error packets divided by the total number of packets sent.
- *Average Packet Delay*, defined as the average number of retransmissions of each successful packet.
- *Cumulative Distribution Function of Number of Retransmissions*, where the probability distribution of each number of retransmissions is defined as the ratio of amount of each number of retransmissions to total number of successful packets within the simulation run.

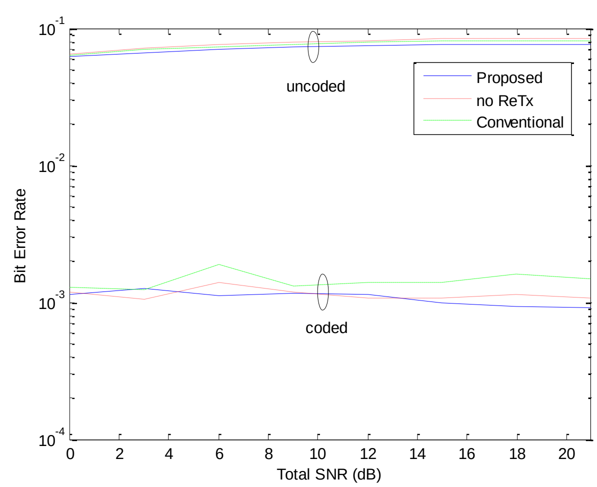

Fig. 2 BER versus total transmit SNR

Fig. 2 shows the BER performance for both the uncoded and coded systems when SNR takes values from 0dB to 25dB. In this figure, *Proposed* denotes the proposed allocation scheme, *Conventional* denotes the conventional *Maximum-Capacity* allocation scheme, and *noReTx* denotes the conventional allocation scheme without retransmission. The notations are also used in the following simulation results. Given the target BER of the LDPC coded to be 10^{-3}, due to the discrete rate adaptation and constant power restriction, the instantaneous BER fluctuates very slightly around the target BER, i.e. 10^{-3}, as SNR varies. The average BER ranges from $0.9594*10^{-3}$ to $1.913*10^{-3}$ at the target BER 10^{-3}. The BER performance is almost the same for the proposed scheme and conventional one when the same BER constraint is given. The corresponding uncoded average BER increases monotonically as the SNR increases and is within an order of magnitude for all the SNR values. The uncoded BER results of the proposed and the conventional schemes are quite close, while both go up slightly as SNR increases due to the impact of the PER. It shows BER constraint is guaranteed under both schemes.

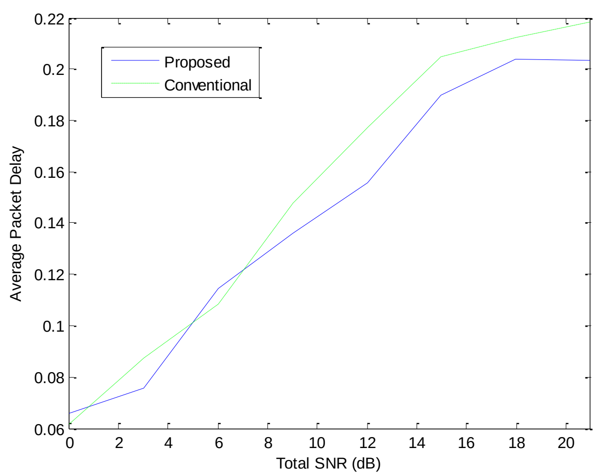

Fig. 3 Average packet delay versus total transmit SNR

The average packet delay is shown for the two allocation schemes in Fig. 3. It is assumed that the transmission of a packet will not be completed until the end of a frame, i.e. each transmission period is a frame. Therefore, the time delay equals the value of delay multiplied by the frame period τ. The number of retransmissions equals packet errors because once a packet is received in error, a retransmission will be initiated; and the total number of packets sent approximates to that of the successful packets when PER is low. Thus, average packet delay has the similar trend as PER according to their definitions.

VI. CONCLUSION

In this paper, a novel adaptive chunk and power allocation scheme with packet retransmissions in the downlink OFDMA system is proposed, which supports ARQ. In order to reduce the maximum number of retransmissions for failed packets while maximizing the throughput of new packets in the system, a new optimization is formulated under a target BER and a total transmission power constraint. Based on that the retransmitted packets have higher priorities to be allocated with resources, the proposed allocation scheme with the use of packet combining technique has better performances than the conventional scheme in terms of the maximum packet delay.

REFERENCES

[1] B. Melamed, "Modeling compressed full-motion video", *Proceedings of the 29th Conference on Winter Simulation*, pp. 1368−1374, 1997.

[2] J. Neckebroek, H. Bruneel, and M. Moeneclaey, "Application layer ARQ for protecting video packets over an indoor MIMO-OFDM link with correlated block fading," *IEEE J. Sel. Areas Commun.*, vol. 28, no. 3, pp. 467–475, Apr. 2010.

[3] C. Y. Wong, R. S. Cheng, K. B. Letaief, and R. D. Murch, "Multiuser OFDM with adaptive subcarrier, bit, and power allocation," *IEEE J. Sel. Areas Commun.*, vol. 17, no. 10, pp. 1747–1758, Oct. 1999.

[4] Z. Shen, J. G. Andrews, and B. L. Evans, "Adaptive resource allocation in multiuser OFDM systems with proportional rate constraints," *IEEE Trans. Wireless Commun.*, vol. 4, no. 6, pp. 2726–2737, Nov. 2005.

[5] Y. J. Zhang, and K. B. Letaief, "Cross-layer adaptive resource management for wirless packet networks with OFDM signalling," *IEEE Trans. Wireless Commun.*, vol. 5, no. 11, pp. 3244–3254, Nov. 2006.

[6] S. Lin and D. J. Costello, Jr., *Error Control Coding: Fundamentals and Applications*, 2nd ed., Prentice Hall: Englewood Cliffs, NJ, 2004, Ch. 17, Ch. 22.

[7] M. Yang, W. E. Ryan, and Y. Li, "Design of efficiently encodable moderate-length high-rate irregular LDPC codes," *IEEE Trans. Commun.*, vol. 52, no. 4, pp. 564–571, Apr. 2004.

[8] H. Zhu, and J. Wang, "Chunk-based subcarrier allocation in OFDMA systems: part I: chunk allocation," *IEEE Trans. Commun.*, vol. 57, no. 9, pp. 2734–2744, Sept. 2009.

[9] J. G. Proakis, *Digital Communications*, 4th ed., New York: McGraw-Hill, 2001.

[10] M.-S. Alouini, and A. J. Goldsmith, "Adaptive modulation over Nakagami fading channels," *Kluwer J. Wireless Commun.* vol. 13, pp. 119−143, May 2000.

[11] S. T. Chung, and A. J. Goldsmith, "Degrees of Freedom in Adaptive Modulation: A Unified View," *IEEE Trans. Wireless Commun.*, vol. 49, no. 9, pp. 1561–1571, Sept. 2001.

[12] B. Xia and J. Wang, "Effect of channel estimation error on QAM systems with antenna diversity," *IEEE Trans. Commun.*, vol. 53, no. 3, pp. 481–488, Mar. 2005.

[13] N. Miki, H. Atarashi, K. Hiquchi, S. Abeta, and M. Sawahashi, "Experimental evaluation on effect of hybrid ARQ with packet combining in forward link for VSF-OFCDM broadband wireless access," *IEEE Proceedings on Personal, Indoor and Mobile Radio Commun.*, PIMRC 2003, vol. 1, pp. 360–365, Sept. 2003.

[14] R. Lucas, M. Fossorier, Y. Kou, and S. Lin, "Iterative Decoding of One-Step Majority Logic Decodable Codes Based on Belief Propagation," *IEEE Trans. Commun.*, 48 (6): 931–37, Jun. 2000.

[15] W. Yu, and J. M. Cioffi, "On constant power water-filling," *IEEE International Conference on Commun.*, ICC 2001, vol. 6, pp. 1665–1669, Jun. 2001.

 ISBN:978-1-61208-113-7