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# Adaptive Subcarrier Allocation and Bit Loading for Multiuser OFDM Systems

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**Abstract**—This paper proposes a new algorithm of adaptive subcarrier allocation and bit loading for simultaneous voice and data transmission in multiuser OFDM systems. The algorithm dynamically assigns the number of subcarriers and bits/per symbol on each subcarrier for each user in a single cell. To cope with the stringent delay requirement of voice service, the subcarriers with low channel gains are assigned for voice transmission with a small number of bits per symbol to achieve the required bit error rate and transmission rate. With the remaining subcarriers, that generally have higher channel gains, and the transmission power, the throughput of data transmission is maximized by loading as many bits as possible on each subcarrier to achieve the required transmission rate and quality. Theoretical analysis and simulations on the proposed algorithm show that better performances are achieved compared to previously reported schemes.

## I. INTRODUCTION

Orthogonal frequency division multiplexing (OFDM) is attractive for combating the effects of delay spread in high-speed wireless data transmission. One way to improve the performance of reliable transmission uses adaptive resource allocation and adaptive modulation (or bit loading), where the modulation scheme used for each subcarrier is selected based on the corresponding multipath channel response. In general, the modulation order (or the number of bits per symbol) used for a particular subcarrier is increased with the corresponding channel gain. With the known instantaneous channel transfer functions of all users, significant performance improvement in terms of channel utilization, bit-error-rate (BER) and the transmission power could be achieved by using adaptive bit loading algorithms [1–4], [16].

Although multiservice transmission is becoming increasingly important to various applications, however, most reported work for OFDM systems ([1–4], for example) focused only on the resource allocation and bit loading without sufficiently considering efficiently supporting different services such as simultaneous transmission of voice and data, which generally have different requirements on the BER and delay tolerance. In [6], algorithms were reported for two services with different requirements of quality-of-service (QoS) in a single user OFDM system. However, it is not appropriate to use the QoS alone as the optimization parameters for the voice/data systems. It is known that the QoS of data service generally requires higher data rates and lower BERs than the voice

service and the delay requirement of voice service is much more critical than the data service. Based on single carrier systems, a scheme of adaptive hybrid binary phase shift keying (BPSK)/ $M$ -ary amplitude modulation ( $M$ -AM) was proposed for voice and data transmission [5]. Since  $M$ -AM is spectrally less efficient than the  $M$ -ary quadrature amplitude modulation ( $M$ -QAM), an adaptive technique employing variable rate uniform  $M$ -QAM (U-MQAM) was reported for integrated voice/data by changing the constellation size dynamically to take advantage of time varying nature of channel fading [7]. Due to the use of uniform  $M$ -QAM constellation, however, the voice transmission achieves unnecessary extra protection at the expense of spectral efficiency and outage probability that means only for data transmission. An adaptive scheme for multi-rate services (A-MRS) [8] was presented for OFDM systems in which the low-rate service for voice transmission was supported by loading one or two bits per subcarrier and high rate data transmission was loaded with more bits, but the subcarriers and transmission power are assigned to voice and data with low efficiency. Consequently, higher speed data transmission cannot be efficiently supported.

Since data transmission generally requires higher data rate and better quality than the voice transmission, it is very important to make the best use of the subcarriers and transmission power for two different kinds of services by analyzing their respective requirements. On the condition of the limited channel bandwidth and transmission power, the adaptive subcarrier allocation and bit loading algorithm proposed in this paper first deals with the efficient subcarrier allocation for voice transmission and then maximize the data transmission rate. An uplink multiuser OFDM/CDMA system is considered for the simultaneous transmission of voice and data, in which any user can share all available subcarriers by using CDMA technique [9], [10]. In general, voice transmission does not need rigid requirements on transmission rate and BER, but has stringent real-time constraints, which are just opposite to the requirements for most data services. This suggests that fixed-rate transmission combined with adaptive bit loading on low quality subcarriers is well suited to voice transmission, while variable-rate transmission, which maximizes the total throughput and obtain low BER, is best suited to data communications. Since only one or two bits are loaded for low-rate voice transmission with relaxed BER requirement,

the selected subcarriers generally have low channel gains. After the requirements of both BER and transmission rate of voice service are satisfied, the remaining subcarriers are loaded with as many bits as possible to achieve the maximum throughput for data transmission. The proposed algorithm applies a threshold on the transmission power of each user, i.e., 23 dBm<sup>1</sup>, which effectively limits the transmission power of each user as well as the interference to other users. The proposed algorithm guarantees a low power consumption on voice transmission [17]. Consequently, most transmission power and subcarriers with better quality, which are generally employed a higher order modulation, can be used for the maximization of data throughput. Our theoretical analysis and experimental simulations show that this adaptive subcarrier allocation and bit loading algorithm brings high utilization of channel and power, or low outage probability and high data rate.

## II. SYSTEM MODEL

The block diagram of adaptive voice/data transmission in a multiuser OFDM/CDMA system is shown in Fig. 1, in which each user uses all  $N$  subcarriers, and  $\{a\}_i$  and  $\{b\}_i$  represent the sets of voice and data signals of user  $i$ , respectively. Based on the estimates of channel gains, the process is adaptively performed to assign a suitable number of bits on the allocated subcarriers for voice and data transmission. After subcarrier allocation and bit loading, spreading codes are added to signals on each user's subcarriers to distinguish different users with decorrelation operations carried out at the receiver. The main task is to determine  $\{N_i^v\}_{i=1}^K$ , which is the number of subcarriers assigned for voice transmission by considering the subcarrier conditions of user  $i$  and the characteristics of voice/data transmission. With the available information on subcarrier allocation and bit loading from the transmitter, the receiver demodulates the voice and/or data signals to  $\{\hat{a}\}_i$  and  $\{\hat{b}\}_i$  on each subcarrier of each user.

At the transmitter, subcarrier allocation and bit loading are performed based on the knowledge of instantaneous subcarriers gains [11]. The fading characteristics of subcarriers are assumed to be constant over one OFDM symbol duration, but vary from symbol to symbol. On the  $n$ th subcarrier of user  $i$ , the BER expression for the MQAM signal with Gray bit mapping in additive white Gaussian noise (AWGN) is approximately a function of received signal-to-noise ratio (SNR),  $\gamma_{i,n}$ , and constellation size,  $M = 2^{c_{i,n}}$ , as [13]

$$\text{BER}_{i,n} \approx \frac{1}{5} \exp\left(\frac{-1.6\gamma_{i,n}}{2^{c_{i,n}} - 1}\right) \quad (1)$$

for  $c_{i,n} \geq 2$ , which is the number of loaded bits on the  $n$ th subcarrier for user  $i$ , and  $0 \leq \gamma_{i,n} \leq 30$  dB. Since the total bandwidth  $B$  is equally divided into  $N$  subbands and signals on these subcarriers are transmitted in parallel, the bandwidth of the subchannels becomes  $B_n = B/N$  for all values of  $n$ . Therefore, the noise variance  $\sigma^2$  can be

rewritten as  $\sigma^2 = N_0 B_n = N_0 B/N$ , where  $N_0$  is the noise power spectral density. Consequently, the received SNR from subchannel  $n$  of user  $i$  can be written as

$$\gamma_{i,n} = \frac{\alpha_{i,n}^2 P_{i,n}}{\sum_{k \neq i}^K \alpha_{k,n}^2 \rho_{i,k}^2(\tau_k) P_{k,n} + \sigma^2}, \quad (2)$$

where  $\sum_{k \neq i}^K \alpha_{k,n}^2 \rho_{i,k}^2(\tau_k) P_{k,n}$  is the interference term due to the sharing a subcarrier by  $K$  users, and  $P_{k,n}$  is the signal power to be transmitted for subchannel  $n$  of user  $k$ . It is clear that the total interference power suffered by a particular user increases with the average service rates of the other users in the system. These factors are effectively considered in the denominator of the received SNR in (2), where  $\rho_{i,k}$ , described below, is the cross-correlation between the spreading codes  $G^i(t)$  and  $G^k(t)$  for user  $i$  and  $k$ ,

$$\rho_{i,k}(\tau_k) = \frac{1}{T_s} \int_0^{T_s} G^i(t) G^k(t - \tau_k) dt \quad (3)$$

where  $\tau_k$  is the time shift for user  $k$  and  $T_s$  is the length of the spreading code.

For a given BER <sub>$i,n$</sub> , the number of bits loaded on the  $n$ th subcarrier of user  $i$  is obtained from (1) as

$$c_{i,n} = \log_2 \left( 1 + \frac{\gamma_{i,n}}{\Gamma_{i,n}} \right), \quad (4)$$

where  $n = 1, 2, \dots, N$ ;  $i = 1, 2, \dots, K$ , and  $\Gamma_{i,n} = -\ln(5 * \text{BER}_{i,n})/1.6$ . Because the transmission rate is the sum of the rates supported by all subcarriers, the total transmission rate of user  $i$  is given in terms of bits/OFDM symbol [2],

$$R_i = \sum_{n=1}^N c_{i,n} = \sum_{n=1}^N \log_2 \left( 1 + \frac{\gamma_{i,n}}{\Gamma_{i,n}} \right). \quad (5)$$

The corresponding power,  $\tilde{P}_{i,n}$ , needed for demodulation at the receiver becomes

$$\tilde{P}_{i,n}(c_{i,n}) = \frac{N_0}{3} \left[ Q^{-1} \left( \frac{\text{BER}_{i,n}}{4} \right) \right]^2 (2^{c_{i,n}} - 1), \quad (6)$$

where  $Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^\infty e^{-t^2/2} dt$ . The derivation of (6) is based on the approximated BER for  $M$ -ary QAM [12]. Therefore, the power needed at the transmitter of user  $i$  is

$$\begin{aligned} P_{i,n}(c_{i,n}) &= \frac{\tilde{P}_{i,n}}{\alpha_{i,n}^2} \\ &= \frac{N_0}{3\alpha_{i,n}^2} \left[ Q^{-1} \left( \frac{\text{BER}_{i,n}}{4} \right) \right]^2 (2^{c_{i,n}} - 1) \end{aligned} \quad (7)$$

where  $\alpha_{i,n}$  is the gain of the  $n$ th subchannel for user  $i$ . To meet the requirements of the transmission rate,  $R_T^v$ , for voice service, we first calculate  $c_{i,n}^v$ , which is the number of bits to be loaded on the  $n$ th subcarrier, and  $N_i^v$ , which is the number of subcarriers used. Then, the total power at the transmitter of

<sup>1</sup>In the absence of any justification, the transmitted power would be 200 mW ( $10 \lg 200 = 23$  dBm), the maximum level for many CDMA mobile handsets.

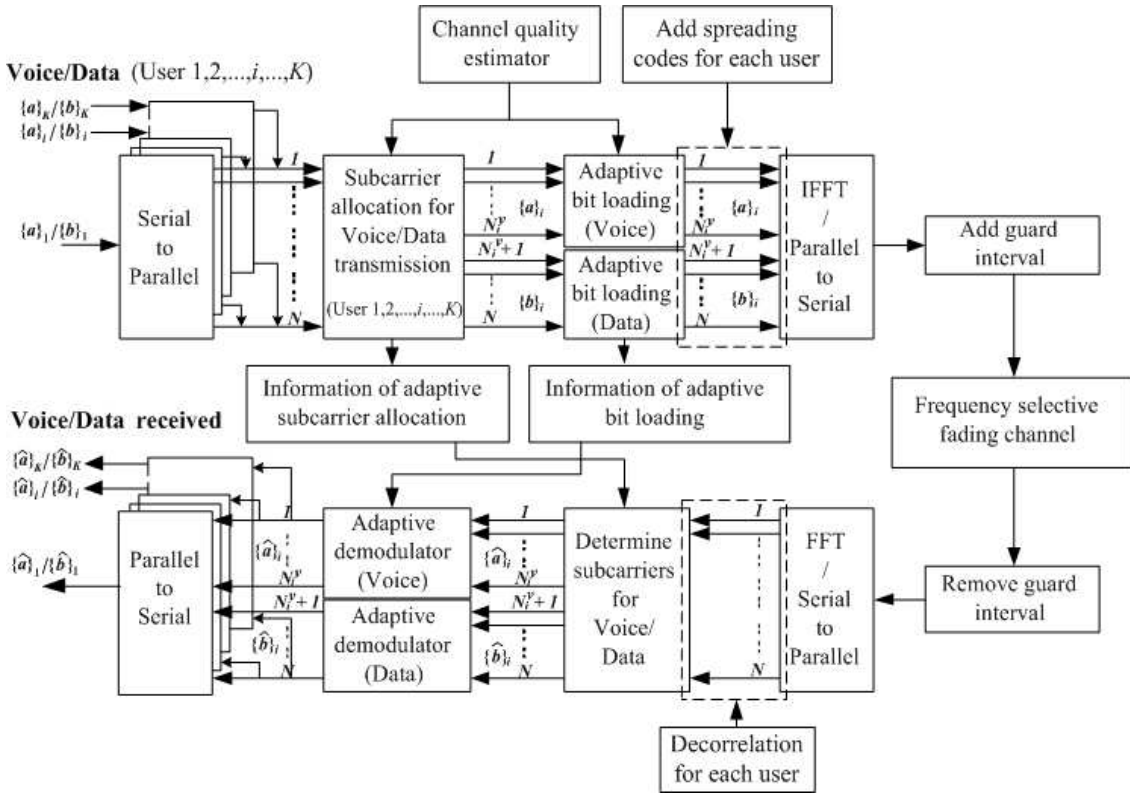


Fig. 1. Block diagram of adaptive voice/data transmission in a multiuser OFDM/CDMA system.

user  $i$  for voice service is

$$\begin{aligned}
 P_i^v &= \sum_{n=1}^{N_i^v} P_{i,n}^v(c_{i,n}^v) \\
 &= \sum_{n=1}^{N_i^v} \frac{N_0}{3\alpha_{i,n}^2} \left[ Q^{-1} \left( \frac{\text{BER}_{i,n}^v}{4} \right) \right]^2 (2^{c_{i,n}^v} - 1). \quad (8)
 \end{aligned}$$

Once subcarrier allocation and bit loading for voice transmission are completed, the maximization of data transmission rate,  $R^d$ , is performed on the condition of allowable transmission power,  $P_i^d = P_T / \bar{\alpha}_i^2 - P_i^v$ , where  $\bar{\alpha}_i$  is the average amplitude of the subcarrier from user  $i$  to the base station, and  $P_T$  is the received power of each user at the base station. According to the dynamic power control theory of CDMA system, the received power from each user, say  $P_T$ , at the base station should be as the same as possible [14].

### III. SUBCARRIER ALLOCATION AND BIT LOADING ALGORITHM

This section presents the adaptive subcarrier allocation and bit loading algorithm (A-SABL) to support simultaneous voice and data transmission in the OFDM multiuser system. The criteria for the adaptive process is to maximize the data throughput,  $R^d$ , under the condition of achieving the required transmission rate  $R_T^d$  with the expected BER for both data and voice services. The first step of the A-SABL algorithm is to assign subcarriers and the number of bits loaded for

voice transmission. With the expected BER, the second step makes the best use of the remaining transmission power and subcarriers to maximize the data throughput. It should be pointed out that the A-SABL algorithm is to achieve the subcarrier allocation and bit loading for both voice and data services under the condition of given transmission power.

#### A. Voice Transmission

Because voice service typically requires low transmission rates with high BERs and small delays, it has a higher priority of the subcarrier allocation to meet the stringent delay requirement. The number of bits,  $c_{i,n}$ , to be loaded on a subcarrier is increased with the quality of the subchannel, i.e., the higher the SNR of the subchannel, the more bits are loaded on the subcarrier. The power to be consumed on the subcarrier is also increased exponentially with the number of loaded bits.

There exist several ways of assigning the voice traffic to the subcarriers. The first one is to arrange the available subchannels into an ascending order based on the gains of subchannels, such as  $\alpha_{i,1} \leq \alpha_{i,2} \cdots \alpha_{i,N_i^v} \leq \cdots \leq \alpha_{i,N}$ , for  $i = 1, 2, \dots, K$ . Consequently, the subcarrier allocation process begins from the worst subchannel. Each subcarrier with a poor channel gain is loaded with a few bits only, such as  $c_{i,n} \leq 2$ . Alternatively, the allocation process can be also performed from the best channel to load a relatively large number of bits, and the subcarriers are arranged with a descending order of the subchannel gains. For the fixed throughput of voice services, the former allocation method

requires many low quality subchannels and the latter approach uses only a few high quality subchannels. The other possible approach to assign subcarriers for voice is the blind allocation without considering the quality of the subchannels.

If the subcarrier arrangement in the descending order is chosen, the subcarriers with better quality are assigned with many bits at the beginning of the allocation process. The required transmission power is increased significantly since the power consumption increases with the number of loaded bits exponentially. However, much less transmission power is used if the subcarrier allocation is performed in an ascending order since the subcarriers for voice are loaded with only one or two bits. Therefore, the subcarrier allocation of our algorithm is based on the ascending order of the subchannel gains. Such an arrangement is not only to keep more transmission power for data transmission, but also substantially reduce the multiuser interference as shown in (2).

Table I presents the algorithm of subcarrier allocation and bit loading for voice transmission, where  $U$  and  $W$  are the sets containing the indexes of subcarriers that are to be discarded and to be assigned, respectively. We first exclude the subcarriers in deep fade, as seen by the user, which can not even support BPSK modulation for the BER requirement of voice transmission. Because these subchannels generally have low channel gains, the number of bits,  $\hat{c}_{i,n}^v$ , loaded on these selected subcarriers are limited within 0, 1 and 2. Subject to the fixed rate  $R_T^v$ , the total number of subcarriers,  $N_i^v$ , and the corresponding number of loaded bits,  $\hat{c}_{i,n}^v$ , are determined and stored in the set  $W$ . Voice outage is declared when the required power for voice transmission,  $P_i^v$ , exceeds a preset threshold, for example,  $P_T/\bar{\alpha}_i^2$ .

### B. Data Transmission

Once the number of subcarriers, i.e.,  $N_i^d = N - N_i^v$ , and their indexes for data transmission are determined, the remaining task is to maximize the total throughput by loading a suitable number of bits onto the available subcarriers. The indexes of the subcarriers usable for data transmission of the  $i$ th user are ranked from  $N_i^v + 1$  to  $N$ . From (5), the total throughput for data transmission is

$$\begin{aligned} R_i^d &= \sum_{n=N_i^v+1}^N c_{i,n} = \sum_{n=N_i^v+1}^N \log_2 \left( 1 + \frac{\gamma_{i,n}}{\Gamma_{i,n}} \right) \\ &= \sum_{n=N_i^v+1}^N \log_2 \left[ 1 + \frac{\alpha_{i,n}^2 P_{i,n} / \Gamma_{i,n}}{\sum_{k \neq i}^{K=1} \alpha_{k,n}^2 \rho_{i,k}^2 (\tau_k) P_{k,n} + \sigma^2} \right], \end{aligned}$$

where  $i = 1, 2, \dots, K$ . Then the maximization of data throughput,  $\max_{c_{i,n}^d} R_i^d$ , is subject to the power limit,

$$\sum_{n=N_i^v+1}^N P_{i,n}^d = P_T / \bar{\alpha}_i^2 - P_i^v \quad (9)$$

which is the constraint on the available power for data transmission. To achieve the maximum data throughput, the transmitted power allocation for data can be performed based

on the number of bits assigned to each available subcarrier. The method of power allocation to maximize the total data rate can be found by using the standard Lagrange multiplier technique [15]. Let us define the Lagrangian function as,

$$\begin{aligned} T_i^d &= \sum_{n=N_i^v+1}^N \\ &\log_2 \left\{ 1 + \frac{\alpha_{i,n}^2 P_{i,n}^d}{[\sigma^2 + \sum_{k \neq i}^{K=1} \alpha_{k,n}^2 \rho_{i,k}^2 (\tau_k) P_{k,n}] \Gamma_{i,n}} \right\} \\ &- \lambda_i \left[ \sum_{n=N_i^v+1}^N P_{i,n}^d - (P_T / \bar{\alpha}_i^2 - P_i^v) \right], \quad (10) \end{aligned}$$

where  $\lambda_i$  is a Lagrange multiplier. By solving  $\partial T_i^d / \partial P_{i,n}^d = 0$  to maximize the total rate of the data transmission, the transmitted power allocated to the  $n$ th subcarrier of the  $i$ th user is

$$\begin{aligned} \hat{P}_{i,n}^d &= \frac{1}{\beta_i} - \frac{[\sigma^2 + \sum_{k \neq i}^{K=1} \alpha_{k,n}^2 \rho_{i,k}^2 (\tau_k) P_{k,n}] \Gamma_{i,n}}{\alpha_{i,n}^2}, \\ n &= N_i^v + 1, N_i^v + 2, \dots, N; \quad i = 1, 2, \dots, K \quad (11) \end{aligned}$$

where  $\beta_i = \lambda_i \ln 2$ . From  $\sum_{n=N_i^v+1}^N \hat{P}_{i,n}^d = P_T / \bar{\alpha}_i^2 - P_i^v = P_i^d$ ,  $\beta_i$  for  $i = 1, 2, \dots, K$  is derived to be,

$$\beta_i = \frac{N^d}{P_i^d + [\sigma^2 + \sum_{k \neq i}^{K=1} \alpha_{k,n}^2 \rho_{i,k}^2 (\tau_k) P_{k,n}] \Gamma_{i,n} \sum_{n'=N_i^v+1}^N \frac{1}{\alpha_{i,n'}^2}}. \quad (12)$$

Substituting (12) into (11),  $\hat{P}_{i,n}^d$ , which is the transmission power for the  $n$ th subcarrier of user  $i$ , becomes,

$$\begin{aligned} \hat{P}_{i,n}^d &= \frac{P_i^d + [\sigma^2 + \sum_{k \neq i}^{K=1} \alpha_{k,n}^2 \rho_{i,k}^2 (\tau_k) P_{k,n}] \Gamma_{i,n} \sum_{n'=N_i^v+1}^N \frac{1}{\alpha_{i,n'}^2}}{N_i^d} \\ &= \frac{[\sigma^2 + \sum_{k \neq i}^{K=1} \alpha_{k,n}^2 \rho_{i,k}^2 (\tau_k) P_{k,n}] \Gamma_{i,n}}{\alpha_{i,n}^2}. \quad (13) \end{aligned}$$

It is possible that  $\hat{P}_{i,n}^d$  is negative for certain subcarriers since some subcarriers are not good enough to be used for data transmission. This results in that the total number of subcarriers,  $N_i^d$ , is reduced and the actual total power,  $\sum_{\{N_i^d\}} \hat{P}_{i,n}^d$ , used for data transmission becomes larger than  $P_i^d$ . In such a case, the transmitted power for each subcarrier of data transmission has to be recalculated based on (13). This process may be performed for several times till  $\{\hat{P}_{i,n}^d\} \geq 0$  for each available subcarrier. In this way, a new group of the available subcarriers is formed, as described in Table II.

From (7), the number of bits loaded to the  $n$ th subcarrier for data transmission of user  $i$  can be calculated by

$$\hat{c}_{i,n}^d = \log_2 \left\{ 1 + \frac{3 \hat{P}_{i,n}^d}{N_0 \left[ Q^{-1} \left( \frac{BER_{i,n}^d}{4} \right) \right]^2} \right\}$$

TABLE I  
ALGORITHMS OF SUBCARRIER ALLOCATION AND BIT LOADING FOR VOICE TRANSMISSION OF USER  $i$

<b>1</b>	<b>Initialization</b> For user $i$ , $\alpha_{i,1} \leq \alpha_{i,2} \leq \dots \leq \alpha_{i,N}$ , $i = 1, 2, \dots, K$ Let $N_i^v = 0$ ; $R_i^v = 0$ ; $W = \{\phi\}$ ; <span style="float: right;">/* <math>W</math> includes used subcarriers for Voice */</span> $U = \{\phi\}$ ; <span style="float: right;">/* <math>U</math> includes discarded subcarriers */</span> $U \cup V = \{1, \dots, N\}$ ; $U \cap V = \{\phi\}$ .
<b>2</b>	for $n = 1$ to $N$ $\{ c_{i,n}^v = 1$ ; <span style="float: right;">/* BPSK Modulation */</span> $BER_{i,n}^v = \frac{1}{2} \text{erfc}(\sqrt{\gamma_{i,n}})$ [5]; <span style="float: right;">/* According to <math>\gamma_{i,n}</math> in (2) */</span> if $BER_{i,n}^v > BER_{i,n}^v$ $\{ U = U \cup n$ ; <span style="float: right;">Continue }</span> $\}$ Obtain $V$ based on $U$ . <span style="float: right;">/* <math>V</math> includes available subcarriers */</span>
<b>3</b>	for $n = V(1)$ to $N$ <span style="float: right;">/* Start Loading from subcarrier <math>V(1)</math> */</span> $\{ c_{i,n}^v = \log_2 \left( 1 + \frac{\gamma_{i,n}}{\Gamma_{i,n}} \right)$ ; if $c_{i,n}^v > 3$ $\{ c_{i,n}^v = 3$ ; $R_i^v = R_i^v + c_{i,n}^v \}$ if $R_i^v \leq R_T^v$ $\{ W = W \cup n$ ; $N_i^v = N_i^v + 1$ } else Break } $P_i^v = \sum_{n=1}^{N_i^v} P_{i,n}^v (c_{i,n}^v)$ ; <span style="float: right;">/* Transmitted power for Voice */</span> if $P_i^v > P_T / \alpha_i^2$ { Voice outage occurs } else $\{ P_i^d = P_T / \alpha_i^2 - P_i^v \}$ . <span style="float: right;">/* Transmitted power for Data */</span>

TABLE II  
CALCULATION OF THE TRANSMITTED POWER FOR DATA TRANSMISSION OF USER  $i$

<b>1</b>	Obtain $\{\hat{P}_{i,n}^d\}$ with $N_i^d$ by (13),
<b>2</b>	Loop begins with $N_i^d$ Updating $N_i^d$ by discarding subcarriers with $\hat{P}_{i,n}^d < 0$ , Calculating $\{\hat{P}_{i,n}^d\}$ by (13), Loop ends till all $\{\hat{P}_{i,n}^d\} \geq 0$ and $\sum_{\{N_i^d\}} \hat{P}_{i,n}^d = P_i^d$ .
<b>3</b>	Obtain updated $N_i^d$ and $\{\hat{P}_{i,n}^d\}$ .

which maximizes the total throughput of data transmission. In practice,  $\hat{c}_{i,n}^d$  should be rounded to the nearest integers for the corresponding modulation schemes. The subcarriers with  $\hat{c}_{i,n}^d = 0$  are to be discarded. If the total calculated data rate  $R_i^d = \sum_{n=N_i^v+1}^N \hat{c}_{i,n}^d$  is smaller than  $R_T^d$ , it means that the transmission power for data service is not enough to support the data rate  $R_T^d$  and then the data outage of user  $i$  is declared.

According to A-SABL algorithm, each user can share all subchannels. The ascending arrangement of subcarrier allocation for voice transmission makes the best use of subcarriers with low channel gains and consumes less power. Since the subchannel fading varies among different users, it is likely that on the same subchannel, one user is transmitting voice while the other user is transmitting data. Each user finds its own suitable subcarrier for voice and data transmission dynamically based on the fading of all subchannels. Consequently, this avoids the situation that some subcarriers are always assigned to one user regardless the change of channel fading.

#### IV. PERFORMANCE COMPARISON

We now evaluate the performance of the proposed subcarrier allocation and bit loading algorithm given in Table I, Table II and (14) by computer simulations. The channel is modeled by a finite impulse response filter with time-varying coefficients

[4]. The impulse response for the experiments was generated based on the symbol-spaced impulse response by fading each of the impulses with the Rayleigh distribution of a normalized maximal Doppler frequency of  $f_d' = 1.235 \times 10^{-5}$ , where the normalization time duration is the length of the OFDM symbol [4].

Let us consider the voice/data OFDM/CDMA system that has 256 subcarriers. Subcarriers are assigned to support voice service first and the remaining subcarriers are used for data transmission. With the A-SABL algorithm, studies are to be made on the performance of voice/data transmission in multiuser systems. Simulation results of the average BERs, service rates and outage probability obtained by the A-SABL method are presented for various average channel SNRs and numbers of users. These results are compared with those of the adaptive multi-rate services (A-MRS) [8] and the uniform  $M$ -QAM (U-MQAM) [7], which are extended to be used in the same multiuser OFDM/CDMA system. We assume that the total transmitted power is 20 dBm and the required BERs for voice,  $BER^v$ , and for data,  $BER^d$ , are  $10^{-2}$  and  $10^{-5}$ , respectively. The voice rate requirement  $R_T^v$  for the first simulation is assumed to be 50 bits/symbol and  $R_T^d$  is 400 bits/symbol.

In terms of the average service rate, Fig. 2 shows the voice and data rates supported by the three algorithms with different average channel SNRs in a three-user system. The required voice rate of 50 bits/symbol is guaranteed by A-SABL, A-MRS and U-MQAM. For the data rate, the proposed algorithm can support 70 bits/symbol more than U-MQAM and nearly 26 bits/symbol more than A-MRS. On the same condition with the same data rate of 400 bits/symbol, the average BER of three users are shown in Fig. 3. The voice BERs obtained by the three schemes have similar changes and the BER requirement of  $10^{-2}$  is satisfied when the average channel SNR is more than 16 dB. To achieve  $BER = 10^{-5}$  requirement

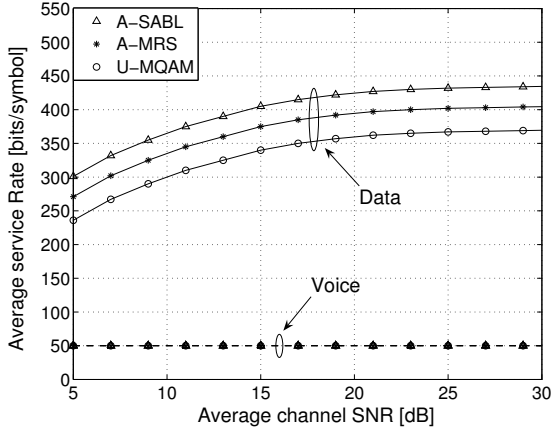


Fig. 2. Comparison of average voice/data rates in a three-user system.

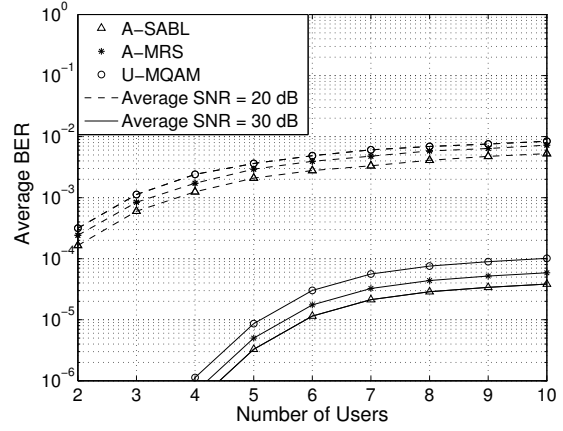


Fig. 4. Comparison of BERs of voice service with different numbers of users.

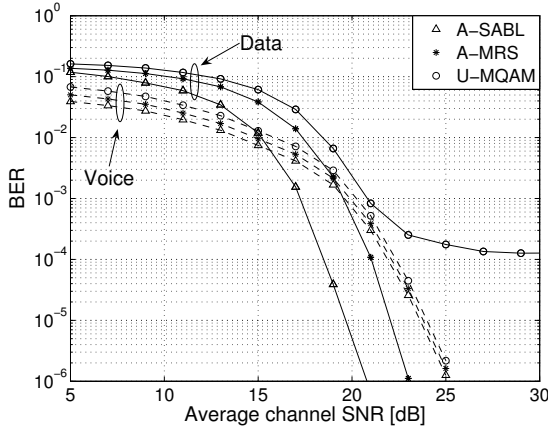


Fig. 3. Comparison of BERs obtained from simulations of a three-user system.

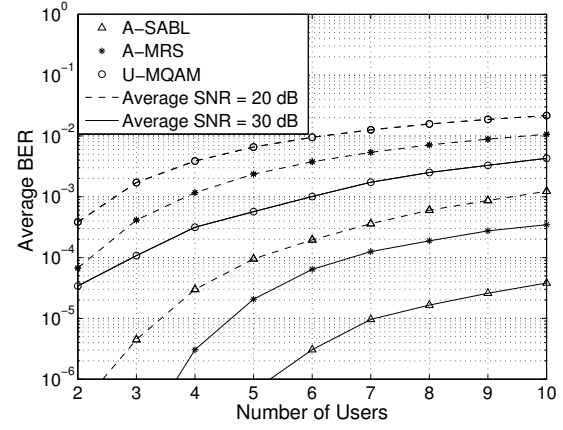


Fig. 5. Comparison of BERs of data service with different numbers of users.

for data service, the A-SABL must have an average SNR of 19 dB, which is about 2 dB lower than the A-MRS. For the U-MQAM scheme, 400 bits/symbol cannot be possibly supported to achieve the average BER of  $10^{-5}$ .

Since users in multiuser systems share all the subchannels, the interference among different users has significant effects on the BERs. Fig. 4 presents the performance of voice service when the average channel SNRs are 20 and 30 dB, respectively. For A-SABL, A-MRS and U-MQAM, the achieved average BERs for voice service have a similar tendency as the numbers of users in the system is increased. It is seen that when 10 users are in the system, the BER requirement of  $10^{-2}$  can be still achieved when the average SNR is 20 dB and the voice rate  $R_T^v = 50$  bits/symbol. For data service with BER requirement of  $10^{-5}$  and rate requirement of 400 bits/symbol, Fig. 5 shows that the A-SABL algorithm can support at least 2 and 6 more users than the A-MRS and U-MQAM, respectively, as the average channel SNR is 30 dB. For BER requirement of data transmission to be  $10^{-4}$ , the proposed algorithm supports about 3 more users than the A-MRS and U-MQAM when the average channel SNR is 20 dB. In general,

the BER performance of the A-SABL is degraded when the number of users in the system is increased due to the limited channel bandwidth. Alternatively, the effect of increasing the number of users can be compensated by reducing the data rates supported by the subcarriers. Fig. 6 shows that, at the average channel SNR of 20 dB, the data rate in the system with A-SABL changes from 427 to 365 bits/symbol when the number of users increases from 2 to 10, while the data rates supported by A-MRS and U-MQAM decrease from 395 to 336 bits/symbol and 365 to 297 bits/symbol, respectively. However, the voice rates are guaranteed at 50 bits/symbol by all the three schemes. Practically, different users have the different SNRs, which makes it more difficult to observe the effects of the interference among users when the number of users is increased. It is possible that the interference from two users with some average SNRs is the same as that from one user with the average SNR of 30 dB, since this user can carry more bits with more power. Therefore, in order to show the effect of interference among users when the number of users changes, we assume that all the users are in the similar environment around the base station in this simulation.

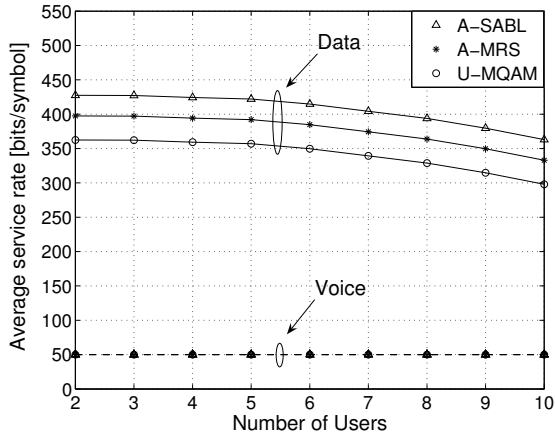


Fig. 6. Comparison of average voice/data rates with different numbers of users (average SNR=20 dB).

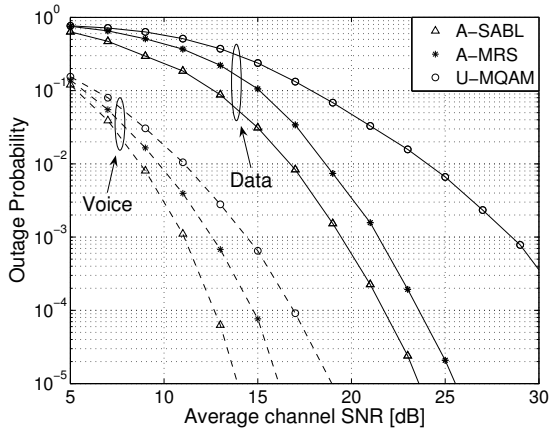


Fig. 7. Comparison of outage probabilities for voice/data with  $R_T^v = 50$  bits/symbol,  $R_T^d = 400$  bits/symbol and  $BER^v = 10^{-2}$ ,  $BER^d = 10^{-5}$ .

Based on the target BER performance and rate requirements, We also consider the outage probability of data service. Since the voice outage is declared when the transmitted power for voice exceeds the preset power threshold, the outage probability of voice service can be obtained from calculating the transmitted power for voice directly. Fig. 7 shows the comparison of outage probability for voice/data versus average SNR in a three-user system. With the outage probability of  $10^{-3}$ , the A-SABL achieves, respectively, 1.5 dB and 4 dB SNR reduction compared with A-MRS and U-MQAM for voice transmission, and with the outage probability of  $10^{-5}$ , nearly 2 dB SNR reduction is achieved compared with the A-MRS for data service. Due to the poor BER performance of U-MQAM, as shown in Fig. 3, it has a much higher outage probability when the data rate requirement is 400 bits/symbol.

## V. CONCLUSION

This paper proposes an adaptive subcarrier allocation and bit loading algorithm for integrated voice/data transmission in a multiuser OFDM system. The objective is to meet the

realtime requirement and the QoS for voice transmission, and maximize the transmission throughput for data service and to achieve the target BER with given transmission power. The improved performance obtained by the A-SABL scheme results from more efficient utilization of the total transmitted power to support much higher data rate. The simulation results show that the proposed algorithm achieves significantly better performance, in terms of maximizing data rate, BER and outage probabilities, than that achieved by the A-MRS and U-MQAM methods.

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