Abstract—The data rates of wireless networks are getting faster and faster. With this trend, IEEE 802.11a, published in 1999 to provide faster data transmission rates, is one of the strongly considered next generation wireless LAN standards. It uses CSMA/CA as the medium access control protocol and OFDM as the physical layer technology. In this paper, to enhance the throughput performance of CSMA/CA protocol in the OFDM based wireless LAN like IEEE 802.11a, we propose a new contention algorithm called parallel contention algorithm that divides the subcarriers into multiple groups to reduce the contention time. We analyze our proposed scheme by extending the Markov chain model and verify the accuracy of the analysis through the simulations. Our protocol performs well especially when the transmission speed and the number of users are getting higher, thereby achieving a better performance improvement ratio than the original IEEE 802.11a standard.

Keywords-802.11; wireless LAN; CSMA/CA; contention window

I. INTRODUCTION

In these days, multiple wireless networking technologies have been proposed for the next generation wireless networks. They include Bluetooth and Ultrawideband for wireless Personal Area Networks (PANs), IEEE 802.11, Home Radio Frequency (HomeRF) and High-Performance Radio LAN (HIPERLAN) for wireless Local Area Networks (LANs), and IEEE 802.16 for wireless Metropolitan Area Networks (MANs). Among these, IEEE 802.11a and HIPERLAN II are well focused due to its support of high data rates, and they use Orthogonal Frequency Division Multiplexing (OFDM) as a powerful candidate technology for the physical layer.

The IEEE 802.11 wireless LAN standard [1] is the most competitive wireless data communication solutions. It uses Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) as the basic algorithm for Medium Access Control (MAC). For the Physical Layer (PHY), it adopts the Complementary Code Keying (CCK) and Packet Binary Convolutional Coding (PBCC) for IEEE 802.11b, and OFDM for IEEE 802.11a [2], [3].

The IEEE 802.11a operates in the unlicensed 5 GHz radio band, whereas the IEEE 802.11b in the unlicensed 2.4 GHz band. Each is with its own modulation technique. The 802.11a system provides the data rate up to 54 Mbps, which is much higher than that of the 802.11b system which shows the maximum rate of 11Mbps.

In this paper, we propose a new contention algorithm for the MAC protocol in OFDM based networks. This algorithm improves the throughput performance and channel utilization by lowering the contention time. This can be made possible by grouping subcarriers of OFDM into smaller subgroups.

The paper is outlined as follows. In Section II, we briefly review the CSMA/CA mechanism and OFDM of the IEEE 802.11a system, and subcarrier grouping. Section III proposes our parallel contention algorithm, and Section IV deals with the analysis of our method by extending the Markov chain model. The analysis results are confirmed through the simulation results and the performance improvement over the current standard is provided especially for high-speed data rates. Finally, Section V concludes our paper.

II. BACKGROUND

A. IEEE 802.11 Distributed Coordination Function

The Distributed Coordination Function (DCF) defines how the medium is shared among members on a wireless network. Since it is difficult to detect and manage the transmission collision on the wireless media, the DCF uses the CSMA/CA method that does not require any collision detection capability at the hardware level. In the CSMA/CA, the station senses the medium before transmission to check whether it is in use or not. If the medium is in use, the station waits for some amount of time before access trial according to its own rules. If the station senses the medium idle for some time equal to a DCF interframe space (DIFS), it will have the right to transmit a frame over the channel.

After the busy period, the station with a frame to transmit generates a random backoff time that is uniformly distributed in the range of $[0, CW - 1]$. This value of Contention Window (CW) is doubled every transmission failure until the contention resolves. The backoff time counter decreases by one every slot time, $\sigma$, as long as the channel is sensed idle, pauses when the channel is detected busy, and reactivated when idle again for a DIFS interval. The station transmits the frame when the backoff time counter reaches zero.
The basic access mechanism uses the two-way handshaking technique. So if the destination station receives the data frame successfully, it replies with an ACK after a short period of time called short interframe space (SIFS) which is shorter than a DIFS. The DCF also defines a four-way handshaking technique for transmission that can be optionally used. This mechanism is shown in Fig. 1.

![Figure 1. RTS/CTS/data/ACK and NAV setting. (RTS/CTS access mechanism)](image)

The four-way handshaking technique is a virtual carrier sensing mechanism, which uses Request to Send (RTS) and Clear to Send (CTS) frames. Once a station hears the RTS or CTS frame that is destined for some other station, it sets the Network Allocation Vector (NAV). The NAV is a timer that indicates the amount of time during which the station remains waiting before the medium access trial. When it reaches zero, it is understood that the medium is free and available for contention. That is, new contention resolution period begins again. The four-way mechanism has the advantage of naturally solving the hidden terminal problem at the cost of sending the two additional short frames of RTS and CTS.

**B. Physical Layer of 802.11a: OFDM and Subcarrier grouping**

IEEE 802.11a uses OFDM as its transmission technology and has several versions of quadrature amplitude modulation (QAM) to support up to 54 Mbps of the transmission rate. OFDM is based on the Fast Fourier Transform (FFT) which enables several subcarriers to be overlapped without losing their orthogonality.

The purpose of overlapping is to use the limited channel resource maximally in an efficient manner. The high data rate transmission is made possible by combining many lower-speed subcarriers to create one high-speed channel. The wireless channel is divided into 52 subcarriers among which forty-eight subcarriers are for data and four for pilot signals.

The subcarrier grouping method is originally introduced in IEEE 802.16 and named Orthogonal Frequency Division Multiple Access (OFDMA). The OFDMA method uses a set of grouped subcarriers as a subchannel. So several stations can simultaneously use one OFDM channel which is divided into subchannels [7].

Three kinds of subcarrier grouping methods are introduced in [9]. The first one is “grouped subcarriers.” This simplest method is to slice up the OFDM channel into several subcarrier groups to minimize the inter-user interference. However, grouping the subcarriers makes the transmission susceptible to fading since the whole group of subcarriers can be lost in a null in the spectrum.

The second one is “adaptive frequency hopping.” It assumes that all users have different channel characteristics according to their locations. Each station selects the most effective subcarrier with few collisions. However, in the fast fading channel environment, we cannot use this method because the channel estimation is so complex.

The last method is “comb spread subcarriers” which example is shown in Fig. 2. Subcarriers are allocated in a comb pattern and spread over the entire system bandwidth. Each channel can work properly even though there exists the null of frequency selective fading channel. We can assume that the synchronization for the contention resolution period is intrinsic because the end of data transmission can be used as a synchronization beacon in this method. The station broadcasts the end of data by transmitting an ack frame. Therefore the inter-user interference in the “comb spread subcarriers” can be minimized, which is the primary reason that we adopt this method for our algorithm. From now on we call each group of “comb spread subcarriers” “a subcarrier group.”

### III. PARALLEL CONTENTION ALGORITHM

Now we propose a new CSMA/CA contention algorithm named the parallel contention algorithm. The purpose of this algorithm is to resolve contention among the competing stations faster than the original standard.

The existing CSMA/CA backoff mechanism decreases the backoff counter by one every slot time, and the station with the backoff counter zero transmits the data frame (see Fig. 3(a)). Differently from the original method, our parallel contention algorithm decreases the backoff counter by the group size of \( G \) not by one (see Fig 3(b)).

![Figure 2. Subcarrier grouping (comb spread subcarriers)](image)

![Figure 3. Examples of backoff count. (a) Standard (b) Parallel contention algorithm (\( G=3 \))](image)

Our algorithm uses the procedures of RTS/CTS dialogue. When a station transmits an RTS frame, it uses one of the subcarrier groups with the randomly selected backoff counter. The purpose of using the subcarrier selection mechanism is to
avoid the collision among the stations that have different backoff counters.

The example in Fig. 3(b) shows that every station decreases the backoff counter by 3, the size of $G$, every slot time. When the backoff counter reaches 0, 1 or 2, the station transmits an RTS frame over the corresponding subcarrier group 0, group 1, or group 2, respectively (see Fig. 2). The destination station replies with a CTS frame over the same subcarrier group after a SIFS interval. If the source station receives the CTS frame successfully, it transmits the data frame by using all the subcarrier groups after a SIFS. That is, the station uses the allocated whole OFDM channel.

Since each station is assumed to hear all the signals from the subcarrier groups, it can receive all the CTS frames that are transmitted at the same slot. Therefore the source station knows the order of transmission, that is, which station is first and which is last at the same slot. For example, the station with the subcarrier group 0, which receives the CTS with the backoff counter zero, transmits the data first and so on. The detailed example is shown in Figs. 4 and 5.

In Fig. 4, stations 1 and 2 initially set the backoff counters at 25 and 20, respectively. Station 2 terminates the backoff process before station 1, and sets new backoff counter at 15. After that, station 1 finishes the remaining backoff process. Fig. 5 shows that since stations 2 and 3 collide in the RTS frame transmissions, they fail to receive the CTS frames. Therefore, station 1 receiving the CTS frame transmits the data packet. Stations 2 and 3 try retransmission later without any loss of channel utilization due to collision.

In this section, we analyze the saturation throughput of parallel contention algorithm by extending the Markov chain model introduced in [5]. We use the two same assumptions as used in [5] for the saturation throughput analysis. One is the ideal channel condition to guarantee no hidden terminal and capture effect. The other is the assumption of saturation condition to make data frames always ready for transmission at the queue of each station. For convenience, we use the number $G$ as a divisor of minimum contention window. That is, $CW_{min} = aG$ ($a$ is an integer).

![Markov chain model (Some flows are simplified.)](image)

**Figure 6.** Markov chain model (Some flows are simplified.)

### A. Markov Chain Model

To model the parallel contention algorithm, we use the discrete time Markov chain. Let the stochastic processes $s(t)$ and $b(t)$ represent backoff stage and backoff counter at time $t$, respectively. By using the bidimensional process $\{s(t), b(t)\}$, we can depict the discrete-time Markov chain as shown in Fig. 6. $b(t)$ decreases by one at the beginning of slot. Since the backoff counter decrement stops when the channel is sensed busy, the time interval between two consecutive decrements can be much longer than the slot time size $\sigma$. This means that the interval may include the data transmission time.

We use the short notation of $P\{i, k | i_s, k_s\}$ to represent $P\{s(t') = i_s, b(t') = k_s | s(t) = i, b(t) = k\}$, where $t'$ denotes the next slot beginning time. As in [5], we approximate that each transmission attempt experiences a constant and independent collision probability $p$ regardless of the number of retransmissions suffered. Additionally, we define $W_i$ as the contention window size at backoff stage $i$ which is equal to $2CW_{min}$, and $m$ as the allowed maximum number of backoff stages. If a station experiences more than $m$ consecutive collisions, it will give up transmission. Then, in the Markov chain of Fig. 6, the only non-null one-step transition probabilities are given as follows.

\[
\begin{align*}
P\{i, k | i, k + G\} &= 1 & i \in [0, m], k \in [0, W_i - 1] \\
P\{0, k | i, g\} &= (1 - p)/W_i & i \in [0, m], k \in [0, W_i - 1] \\
P\{i, k | i - 1, g\} &= p/W_i & i \in [1, m], k \in [0, W_i - 1] \\
P\{m, k | m, g\} &= p/W_i & k \in [0, W_m - 1].
\end{align*}
\]

where the number $g$ is an element from the set $\{0, 1, 2, \cdots, G - 1\}$.

**IV. THROUGHPUT ANALYSIS AND SIMULATIONS**

In this section, we analyze the saturation throughput of parallel contention algorithm by extending the Markov chain model introduced in [5]. We use the two same assumptions as used in [5] for the saturation throughput analysis. One is the ideal channel condition to guarantee no hidden terminal and capture effect. The other is the assumption of saturation condition to make data frames always ready for transmission at the queue of each station. For convenience, we use the number $G$ as a divisor of minimum contention window. That is, $CW_{min} = aG$ ($a$ is an integer).
Since the Markov chain model for the parallel contention algorithm is too complex to be represented in a figure, we omitted some flows for backoff counter decrement and successful transmission. Successful transmission flows are simply represented as the “Success” flow.

B. Analysis of the Markov Chain Model

In this subsection, we analyze the stationary distribution of the Markov chain model in Fig. 6. Let $b_{i,k} = \lim_{i \to +\infty} P\{i,k\}$, $i \in [0,m]$, $k \in [0,W_i-1]$ be the stationary distribution of the chain. Since $b_{i,k} = b_{i,j}$ for all $i \in [0,m]$, $k \in [0,G-1]$, and $l \in [0,G-1]$, we can obtain $\sum_{i=0}^{m} b_{i,k} = Gb_{i,g}$.

Since $G$ is the divisor of $W_i$, we have

$$b_{i,g} = p^{i}b_{i,0}, \quad (0 \leq i < m)$$

and for $k \in [G,W_i-1]$, we obtain

$$b_{i,k} = G \left\{ \frac{W_i-k}{G} \right\} \left\{ \begin{array}{ll}
(1-p) \sum_{j=0}^{m} b_{j,g} & (i = 0) \\
p^{i}b_{i-1,k-g} + b_{i,g} & (0 < i < m) \\
p^{i}(b_{i+1,g} + b_{i,g}) & (i = m)
\end{array} \right. \quad (3)$$

From $\sum_{i=0}^{m} b_{i,k} = \frac{b_{i,k}}{1-p}$, we can rewrite (3) as

$$b_{i,k} = G \left\{ \frac{W_i-k}{G} \right\} b_{i,g}, \quad i \in [0,m], k \in [0,W_i-1]. \quad (4)$$

Thus, by using the relation of (2) and (4), all the $b_{i,k}$’s are expressed as a function of $b_{i,g}$ and the conditional collision probability $p$. From $\sum_{i=0}^{m} \sum_{k=0}^{W_i-1} b_{i,k} = 1$, we can write

$$1 = \sum_{i=0}^{m} \sum_{k=0}^{W_i-1} b_{i,k} = \sum_{i=0}^{m} \sum_{k=0}^{W_i-1} G \left\{ \frac{W_i-k}{G} \right\} = \sum_{i=0}^{m} b_{i,g} \frac{W_i+G}{2}$$

$$= \sum_{i=0}^{m} p^{i}b_{i,g} \frac{2W_i+G}{2} + \frac{p^{m}b_{i,g}}{1-p} \frac{2W_i+G}{2} \quad (5)$$

$$= \frac{b_{i,g}}{2} \left[ W_i \left( \sum_{i=0}^{m} \left( \frac{2p}{1-p} \right)^{i} + \frac{2p^{m}}{1-p} \right) + G \frac{1}{1-p} \right]$$

where

$$b_{i,g} = \frac{2(1-2p)(1-p)}{(1-2p)(W_i+G) + pW_i(1-(2p)^m)}. \quad (6)$$

Now, we define $\tau$ as the probability that a station transmits in a randomly chosen slot. Since any transmission occurs when the backoff counter reaches $g (< G)$, regardless of the backoff stage, we can express $\tau$ as

$$\tau = \sum_{i=0}^{\infty} b_{i,g} = G \sum_{i=0}^{\infty} b_{i,g} = G \frac{b_{i,g}}{1-p} = \frac{2G}{(1-2p)(W_i+G) + pW_i(1-(2p)^m)}. \quad (7)$$

In (7), we can obtain the same result as in [5] which just deals with the case of $G = 1$.

In some other way, we can obtain the different relationship between $p$ and $\tau$. Since the probability of transmission over a subcarrier group that is randomly selected by the random backoff counter is $\tau / G$, we have

$$p = 1 - \left(1 - \frac{\tau}{G}\right)^{n-1}. \quad (8)$$

From (7) and (8) we can obtain the unique solutions for the two unknowns $\tau$ and $p$ through the numerical method. The proof of the uniqueness is the same as in [5].

C. Saturation Throughput Analysis and Simulations

Let $S$ be the normalized system throughput that represents the fraction of time the channel is used to transmit the payload successfully. Let $P_s$ be the probability that there is at least one transmission in the considered slot time and $n$ be the number of stations. Then we simply have the following.

$$P_s = 1 - (1-\tau)^n. \quad (9)$$

Let $P_{sk}$ be the probability that there are $k$ successful transmissions, given that there is at least one transmission in the considered slot time and $0 < k \leq G$. Then $P_{sk}$ is given by

$$P_{sk} = \binom{G}{k}(a_{0,0}S_0 + a_{0,1}S_1 + \cdots + (-1)^{k-1}a_{0,G-1,k-1}S_k)/P_s, \quad (10)$$

where $S$ is the probability that the $i$ transmission over the consecutive subcarrier groups from the first to the $i$th are all successful, and $a_{i,j}$ is the entry of the $i$th row and $j$th column of the Pascal’s triangle. Since the calculation process for the $P_{sk}$ is too cumbersome, it is omitted in this paper. Using these, we can express $S$ as

$$S = \frac{\sum_{k=1}^{G} P_s P_{sk} kE[P]}{(1-P_s)\sigma + \sum_{k=1}^{G} P_s P_{sk} T_{sk} + P_s (1-\sum_{k=1}^{G} P_{sk}) T_{C}}, \quad (11)$$

where $E[P]$ is the average payload length, $T_{sk}$ is the time spent by $k$ stations for transmitting data successfully at the same slot, and $T_{C}$ is the time wasted due to collision.

Fig. 7 plots the numerical results obtained through the analysis and simulation. The lines represent the analytical results and the symbols represent the simulation results. Fig. 7 verifies that our model is accurate because the analytical results are the same as the simulation results. The parameter values used are shown in Table I.
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>MAC header</td>
<td>272 bits</td>
</tr>
<tr>
<td>PHY header</td>
<td>128 bits</td>
</tr>
<tr>
<td>ACK</td>
<td>112 bits + PHY header</td>
</tr>
<tr>
<td>RTS</td>
<td>160 bits + PHY header</td>
</tr>
<tr>
<td>CTS</td>
<td>112 bits + PHY header</td>
</tr>
</tbody>
</table>

**Figure 7.** Numerical and Simulation Results (24Mbps).

**Figure 8.** Saturation Throughput Improvement ratio (G = 4).

Table 1: IEEE 802.11a System Parameters Used.

- **Channel Bit Rate:** 24 Mbps
- **Slot Time:** 9 μs
- **SIFS:** 16 μs
- **DIFS:** 34 μs

The parallel contention algorithm uses RTS and CTS frames prior to data transmission. If we would use proper RTS threshold, it would be possible not to use RTS and CTS frames. Therefore, we need to make an experiment to find the proper value of RTS threshold.

**V. Future Work**

The parallel contention algorithm uses RTS and CTS frames prior to data transmission. If we would use proper RTS threshold, it would be possible not to use RTS and CTS frames. Therefore, we need to make an experiment to find the proper value of RTS threshold.

**VI. Conclusion**

In this paper, we proposed a new contention algorithm for CSMA/CA in OFDM based high-speed wireless LANs. We used the generic characteristics of OFDM which uses the subcarrier groups. By dividing the subcarriers into several groups and allocating a group to each station, we could reduce the contention resolution time.

We analyzed our proposed algorithm by extending the Markov chain model in [5]. The numerical results of our model are compared with the simulation results. The Markov chain analysis and simulations proved that our system model is very accurate. Through the numerical analysis we also verified that our algorithm shows the considerably improved performance. The performance has been remarkably improved when the data rates are high. This is because the ratio of the contention resolution time to the data transmission time is relatively high in the high rate transmission compared with the low rate transmission. It also runs well when the number of competing users is large.

**References**


