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Transmission History Based Distributed Adaptive Contention Window Adjustment Algorithm Cooperating with Automatic Rate Fallback for Wireless LANs

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SUMMARY This paper proposes and investigates a distributed adaptive contention window adjustment algorithm based on the transmission history for wireless LANs called the transmission-history-based distributed adaptive contention window adjustment (THAW) algorithm. The objective of this paper is to reduce the transmission delay and improve the channel throughput compared to conventional algorithms. The feature of THAW is that it adaptively adjusts the initial contention window (CW_{init}) size in the binary exponential backoff (BEB) algorithm used in the IEEE 802.11 standard according to the transmission history and the automatic rate fallback (ARF) algorithm, which is the most basic algorithm in automatic rate controls. This effect is to keep CW_{init} at a high value in a congested state. Simulation results show that the THAW algorithm outperforms the conventional algorithms in terms of the channel throughput and delay, even if the timer in the ARF is changed.

key words: *IEEE 802.11, CSMA/CA, contention window adjustment, automatic rate fallback*

1. Introduction

For the past decade, wireless LAN devices have been installed in many types of electronic equipment such as personal computers, portable gaming machines, and portable music players. It is expected that those devices will be introduced into every type of electronic equipment in the years to come. The IEEE 802.11 Working Group established the standard for wireless LANs [1]. IEEE 802.11 wireless LAN networks can be categorized as an infrastructure network that uses an access point (AP) or ad hoc network that does not use an AP. In terms of the wireless LAN market, the shipment volume of portable gaming machines has increased significantly. Hence, this paper focuses on the ad hoc network for single hop communications in which any station can communicate directly with other stations in an environment in which the distance among stations is relatively short, as shown in Fig. 1.

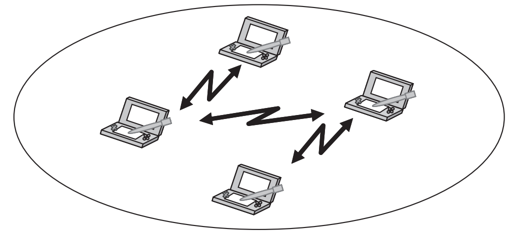

Fig. 1 Ad hoc networks.

Two types of media access control (MAC) protocols, the distributed coordination function (DCF) and point coordination function (PCF), are defined in the IEEE 802.11 standard. However, the PCF cannot be applied to the ad hoc network due to the absence of an AP. The IEEE 802.11 DCF mechanism employs the binary exponential backoff (BEB) algorithm, which is simple and robust, but cannot efficiently utilize the limited wireless channel bandwidth due to packet collisions in a congested state. The major problem is that the initial contention window (CW_{init}) size, which also means the contention window after successful transmission, is kept fixed regardless of the traffic conditions. In other words, the initial size of the backoff range, i.e., [0, CW_{init}], is small under all conditions. In [2], it was reported that the channel throughput is improved with an increase in CW_{init}, but how to adjust CW_{init} to suit the traffic conditions was not addressed.

Some algorithms such as the exponential increase exponential decrease (EIED) backoff algorithm, the determinist contention window algorithm (DCWA), and the fast collision resolution (FCR) algorithm were proposed to address this issue [3]–[8]. In terms of both channel throughput and delay, the performance of these algorithms is not fully efficient. More specifically, CW_{init} in the EIED tends to become small because CW_{init} is decreased once the packet transmission is successful. Hence, many packet collisions occur in the congested state. Additionally, the DCWA changes both backoff range bounds, i.e., upper and lower bounds, according to the channel utilization rate. Not only does the lower bound of the backoff range retain a high value when the channel utilization rate is high but also the initial size of the backoff range is small. The probability of packet collisions is continually high once a packet collision occurs. In addition, the DCWA requires an extra function to measure the channel utilization rate. Furthermore, the FCR greatly modifies the BEB to avoid future potential packet colli-

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sions in advance. Although this FCR achieves good channel throughput, the delay increases in the uncongested state because future potential packet collisions are considered.

The IEEE 802.11 standard supports multiple transmission rates and the rate is selected in an adaptive manner using an automatic rate control algorithm. As the most basic algorithm, there is the automatic rate fallback (ARF) algorithm based on the number of consecutively successful or failed transmissions and the timer control to automatically select the transmission rate that suits the link conditions [9]. Since it is assumed that the distance among stations is relatively short in this paper, the problem is the failed transmissions caused by packet collisions. The reason for this is that the probability of failed transmissions caused by received power degradation is low in that situation. Hence, it is a problem if the transmission rate decreases due to packet collisions. To improve the channel throughput, it is necessary to avoid packet collisions. Therefore, this paper focuses on packet collisions.

As stated above, packet collisions cause channel throughput degradation, and delay increases due to both the retry transmissions caused by packet collisions and the packet collision avoidance. Therefore, this paper proposes a distributed adaptive contention window adjustment algorithm based on the transmission history for wireless LANs, called the transmission-history-based distributed adaptive contention window adjustment (THAW) algorithm, in order to reduce the transmission delay and improve the channel throughput compared to the IEEE 802.11 DCF, EIED, DCWA, and FCR. To keep CW_{init} at a high value in the congested state, THAW is the adjustment of CW_{init} in the BEB based on both the number of consecutively successful or failed transmissions and the timer control, which are employed in the ARF.

The rest of the paper is organized as follows. Section 2 reviews the related studies. Section 3 describes the proposed algorithm, THAW. Section 4 presents the simulation results, and compares the performance of the THAW to that of the conventional algorithms. Finally, Sect. 5 concludes the paper.

2. Related Studies

The IEEE 802.11 DCF is the fundamental access mechanism in the IEEE 802.11 MAC. The IEEE 802.11 DCF is based on carrier sense multiple access with collision avoidance (CSMA/CA). To avoid packet collisions among stations, the IEEE 802.11 DCF mechanism employs the BEB. At each packet transmission, the backoff time is uniformly chosen in the range [0, cw]. The cw is called the contention window, and depends on the number of failed transmissions for the packet. At the first transmission attempt, cw is set equal to the value of $\mathrm{CW_{min}}$, which is called the minimum contention window and is also CW_{init}. If the channel is determined to be busy, the station defers until the channel is determined to be idle for a distributed inter-frame space (DIFS) and the backoff procedure concerning the decrement of backoff time is invoked. If the channel is determined to be idle during a particular backoff slot, then the backoff procedure decrements its backoff time by a slot time. If the backoff time reaches zero, the station transmits the packet. After each unsuccessful transmission, cw is increased, e.g., in the case of IEEE 802.11b, cw is $2^{(i+5)} - 1$, where i is the retry count, up to the maximum value, $\mathrm{CW_{max}}$. After a successful transmission, cw is reset to $\mathrm{CW_{min}}$ for the next packet. This algorithm is represented as follows.

$$\left.\begin{array}{l} cw = \min[2 \cdot (cw+1) - 1, \mathrm{CW_{max}}] \\ \quad \text{when a transmission fails,} \\ cw\ (= CW_{init}) = \mathrm{CW_{min}} \\ \quad \text{when a transmission succeeds.} \end{array}\right\} \tag{1}$$

The values of $\mathrm{CW_{min}}$ and $\mathrm{CW_{max}}$ are 31 and 1023, respectively, in the IEEE 802.11b [1].

In [2], it was reported that the maximum channel throughput is improved with an increase in CW_{init}. In other words, the maximum channel throughput depends on CW_{init}. Hence, CW_{init} must be changed adaptively according to the traffic conditions to improve the channel throughput.

Several dynamic adaptive contention window adjustment algorithms were studied [3]–[7]. However, these algorithms have several problems. The algorithm in [3] requires accurate on-line measurement of the number of active stations. Therefore, it requires an AP to measure the number of active stations.

In [4]–[6], CW_{init} for the next packet is changed according to cw of the ongoing transmission. As in the case of the IEEE802.11 DCF, if the channel is determined to be idle during a particular backoff slot, then its backoff time is decremented by a slot time. This algorithm is represented as follows.

$$\left.\begin{array}{l} cw = \min[r_I \cdot (cw+1) - 1, \mathrm{CW_{max}}] \\ \quad \text{when a transmission fails,} \\ cw\ (= CW_{init}) = \max[(cw+1)/r_D - 1, \mathrm{CW_{min}}] \\ \quad \text{when a transmission succeeds.} \end{array}\right\} \tag{2}$$

Here, [4] and [5] use only $r_I = r_D = 2$. Although the EIED in [6] uses other parameters including $r_I = r_D = 2$, it is assumed that the EIED uses $r_I = r_D = 2$ in this paper. This algorithm changes CW_{init} whenever the ongoing packet transmission is successful. On the other hand, cw is increased after each unsuccessful transmission, similar to the case of the IEEE 802.11 DCF. Since CW_{init} is decreased after a packet transmission succeeds, the value of CW_{init} tends to become small. Therefore, the algorithm is more effective than the IEEE 802.11 DCF, only when packet collisions occur continually.

The DCWA changes both backoff range bounds according to the channel utilization rate, and it defines CW_{lb} and CW_{ub} as the lower and upper bounds, respectively [7]. The backoff range [CW_{lb}, CW_{ub}] in the DCWA differs from that for the IEEE 802.11 DCF; however, it is similar to the IEEE802.11 DCF in that if the channel is determined to be idle during a particular backoff slot, then its backoff time is decremented by a slot time. Furthermore, both CW_{lb} and

CW_{ub} are changed according to retry count i, as shown in the following equations[†].

$$\left.\begin{array}{l} size(i) = 32 \cdot (i+1) \\ CW_{lb}(i) = CW_{ub}(i-1) \\ CW_{ub}(i) = CW_{ub}(i-1) + size(i) \\ backoff\_time = \text{random}[CW_{lb}(i), CW_{ub}(i)] \end{array}\right\} \quad (3)$$

Here, $size(i)$ is the parameter related to the size of the backoff range. When two or more stations transmit a packet with a different retry count, there is no overlapping of the backoff ranges among them. On the other hand, after each successful transmission, both CW_{ub} and CW_{lb} are changed according to the instantaneous channel utilization rate, $B(t)$, as shown in the following equations.

$$\left.\begin{array}{l} CW_{ub}(0) = CW_{ub}(i) \cdot B(t) + \mathrm{CW}_{\min} \cdot (1 - B(t)) \\ CW_{lb}(0) = CW_{ub}(0) - size(0) \end{array}\right\} \quad (4)$$

Here, $\mathrm{CW}_{\min}$ is the minimum contention window size defined by the IEEE 802.11 standard. More specifically, if the channel utilization rate is high, the new $CW_{ub}(0)$ for the next transmission is close to the previous $CW_{ub}(i)$. On the other hand, if the channel utilization rate is low, the new $CW_{ub}(0)$ is close to $\mathrm{CW}_{\min}$. However, since $CW_{lb}(i)$ retains a high value when the channel utilization rate is high and the initial size of the backoff range ($size(0)$) is small, the probability of packet collisions remains high once packet collisions occur. In addition, the DCWA requires an extra function to measure the channel utilization rate.

The FCR greatly modifies the BEB to avoid future potential packet collisions in advance [8]. The FCR has four states: a packet successful state, a packet failure state, a deferring state, and a backoff state. The difference between the FCR and IEEE 802.11 DCF is in the deferring state and backoff state. The deferring state means a situation where the station detects the start of a new busy period, which indicates either a packet collision or a packet transmission. The backoff state represents a situation where the station detects that the channel is idle and the backoff time is reduced. When the station detects the start of a new busy period, which indicates either a packet collision or a packet transmission, the station increases its contention window and selects a new backoff time as well as the transmission failure. Hence, the FCR increases the contention window size when a station is in either the packet failure state or deferring state. In addition, the FCR reduces the backoff time exponentially fast when a pre-fixed number of consecutive idle slots are detected. That is, the feature of the FCR is that the backoff time is reduced when a station is in the backoff state, and the backoff time is increased when the station is in the deferring state. Hence, although the probability of packet collisions is reduced in the congested state because the backoff time is distributed among stations, the delay and the probability of packet collisions increase in the uncongested state because the backoff time is reduced rapidly.

The IEEE 802.11 standard supports multiple transmission rates and the rate is selected in an adaptive manner using an automatic rate control algorithm. As representative algorithms, the ARF and Receiver-Based AutoRate (RBAR) were proposed [9], [10]. The ARF changes the transmission rate according to the transmission status, successful or failed transmission, at the transmitter. The RBAR changes the transmission rate according to the channel quality information. To obtain the channel quality information at the transmitter, the transmitter and receiver must use the RTS/CTS handshake and modify the packet construction. Since it is easy for the contention window algorithm to cooperate with the ARF, this paper focuses on the ARF algorithm. The ARF algorithm, which is used in Lucent Technologies WaveLAN II networking devices, is one of the few algorithms that are available to the public as an automatic rate control [9]. Using the ARF, the transmission rate is switched according to the value of a counter, which counts the number of consecutively successful or failed transmissions. Additionally, the ARF has a timer control to increase to a higher transmission rate. More specifically, the ARF decreases the current transmission rate and starts a timer when two consecutive transmissions fail in a row. When either the timer expires or the number of successfully received acknowledgements (ACKs) reaches ten, the transmission rate is increased to a higher rate and the timer is reset. When the transmission rate is increased, the first transmission after the rate increase must succeed or the rate is immediately decreased and the timer is restarted rather than trying the higher rate a second time. As mentioned above, the ARF has three parameters[††], the number of consecutively successful transmissions, the number of consecutively failed transmissions and the timer value. In [9], two of the three parameters, the number of consecutively successful transmissions and the number of consecutively failed transmissions, are specified, but the timer value is not specified.

This paper focuses on the collaboration between the contention window adjustment algorithm and ARF. Additionally, the proposed algorithm reuses the existing functions in the wireless LANs.

3. THAW Algorithm

3.1 Basic Idea

The THAW algorithm is an amendment to the IEEE802.11 DCF using the ARF. Basically, the THAW algorithm obeys the BEB used in the IEEE 802.11 DCF. The contribution point here is how to change the range of the contention window, and the proposed algorithm focuses on CW_{init}. To keep CW_{init} at a high value in the congested state, THAW adjusts CW_{init} by employing the number of consecutively successful or failed transmissions and the timer control used in ARF. In addition, the THAW considers network conditions based on

[†]In [7] the equations for the DCWA are different from the descriptions. This paper derives the equations for the DCWA from the descriptions in [7], as shown in Eq. (3). Hence, the equations in this paper are different from in those in [7].

[††]These parameters are different in wireless LAN products.

transmission contentions, as well as the conventional algorithms. The point of the proposal is to adjust CW_{init} without changing the parameters in the IEEE 802.11 or ARF. That is, the proposed algorithm must handle any timer value since the timer value is not specified in [9], as mentioned above. There are two objectives in using the ARF. One is the change of CW_{init} employing the number of consecutively successful or failed transmissions, which is the count function in the ARF. The other is that parameter N_{suc}, which is defined later, is optimized corresponding to the timer used in the ARF, and it cooperates with the behavior of the ARF. If the timer value is pre-fixed, the latter objective is not required because the optimal value of parameter N_{suc} is determined by the timer value. In addition, the THAW does not apply RTS/CTS handshake for hidden terminal problems because the failed transmissions due to packet collisions, which mean the ACK is not received after a packet is transmitted, rarely occur if RTS/CTS handshake is used.

3.2 THAW Algorithm

In the THAW algorithm, a station increases CW_{init} once a transmission fails. In other words, the station increases CW_{init} whenever the number of failed transmissions reaches one. On the contrary, whenever the number of consecutively successful transmissions reaches N_{suc}, a station decreases CW_{init}. By employing the number of consecutively successful or failed transmissions used in the ARF, CW_{init} is dynamically changed. The THAW algorithm can be represented as follows.

$$\left.\begin{array}{l} cw = \min[2\cdot(cw+1)-1, \mathrm{CW}_{\max}] \\ \quad \text{when a transmission fails,} \\ cw\ (= CW_{init}) = \max[(cw+1)/2-1, \mathrm{CW}_{\min}] \\ \quad \text{when the number of consecutively successful} \\ \quad \text{transmissions reaches } N_{suc}. \end{array}\right\} \quad (5)$$

Similar to the BEB, cw for the retry transmission is increased after an unsuccessful transmission. That is to say, the new CW_{init} for the next packet is set to cw of the previous transmission until the number of consecutively successful transmissions reaches N_{suc}. The feature of the algorithm is that CW_{init} for THAW is kept at a high value and this is a point of difference from the algorithms in [4]–[6]. However, the THAW algorithm is similar to the IEEE802.11 DCF, EIED, and DCWA in that if the channel is determined to be idle during a particular backoff slot, then its backoff time is decreased by a slot time. If N_{suc} is a large value and CW_{init} is also a large value, delays may increase because the backoff time is always large. Since the ARF has the timer control to increase to a high transmission rate, the fall of the transmission rate caused by unsuccessful transmissions may be recovered by the timer. The merit of a high transmission rate is that the channel occupancy time is short, and a higher channel throughput is achieved. Hence N_{suc} must be changed according to any value of the timer because the ARF does not define the value of the timer.

When the timer control is invoked before the number of consecutively successful transmission reaches 10 in the ARF, N_{suc} must be a small value, to reduce the channel wasted time. However, since CW_{init} tends to be small when N_{suc} is a small value, the packet collision probability will become high. Hence, N_{suc} must be a large value after unsuccessful transmission. Considering that, the value of N_{suc} is changed according to the following equations.

$$\left.\begin{array}{l} N_{suc} = N_{suc} - Dp \\ \quad \text{when the transmission rate is increased} \\ \quad \text{by the timer control,} \\ N_{suc} = N_{suc} + 1 \\ \quad \text{when the first transmission after decreasing} \\ \quad CW_{init}\text{, is unsuccessful.} \end{array}\right\} \quad (6)$$

Here, Dp is a parameter to decrease N_{suc}. The condition of the former equation means that the THAW judges the wasted time related to backoff by comparing the behaviors of two controls for transmission rates, which are the timer and number of consecutively successful transmission. The condition of the latter equation means that the THAW judges the congestion state because packet collisions occur immediately after decreasing CW_{init}. Here, the initial value of N_{suc} is set to the maximum value within the range of N_{suc}. After the N_{suc} is tuned using the timer by the former equation, it is fine-tuned in accordance with packet collisions by the latter equation. That is, the reason why the parameter is used in the subtraction is that Dp makes N_{suc} a small value actively. Moreover, the multi-rate is supported by fine-tuning even if the value of Dp is fixed. In the next section, Dp and the range of N_{suc} are discussed.

As mentioned above, the THAW algorithm obeys the BEB used in the IEEE 802.11 DCF and reuses the counter to measure the number of consecutively successful or unsuccessful transmissions used in the ARF. Therefore, the proposed algorithm is simple and compatible with both the IEEE 802.11 DCF and ARF.

4. Performance Evaluation

4.1 Simulation Conditions

To clarify the performance, the simulation conditions are defined. As common parameters, the simulation parameters given in Table 1† are used. The other parameters for the conventional algorithms are the same as those given in the respective references for the algorithms. In addition,

†Since each station has a transmission queue, the packet generation interval is shorter than the packet transmission interval when the offered traffic is high. Hence, the behavior of the packet transmission is bursty, and this behavior corresponds to a general traffic case such as UDP traffic in this situation. On the other hand, the characteristics we evaluate do not depend on the traffic generation conditions because the packet collision probability is low when the offered traffic is low. Therefore, the simulation results show valid characteristics even if the Poisson process for packet generation is used. Although the Poisson process is also used in conventional studies such as [11], the evaluation which actual application traffic is applied to is the challenging issue.

Table 1 Simulation parameters.

Parameter	Value
Transmission Rate	11.0 / 5.5 / 2.0 / 1.0 Mbps
Packet Length	1500 bytes
Queue Length	100 packets
Packet Generation	Poisson Process
Automatic Rate Control	ARF
Timer in ARF	50 msec / 5 sec

the parameters for THAW are the same values as those for IEEE 802.11b. This paper considers only the unsuccessful transmissions caused by a packet collision because it is assumed that the distance among stations is relatively short, as mentioned above. That is, the maximum transition rate is 11 Mbps. In those evaluations, access method is basic access without RTS/CTS handshake for hidden terminal problems. Moreover, a source station randomly selects the destination station.

There are two traffic models for the simulations.

Model 1) The offered traffic for each station is 400 kbps and the number of stations is variable.

Model 2) The number of stations is 10, and the offered traffic for each station is the same and it is variable.

The other case is evaluated in the Appendix. Additionally, the simulations are carried out ten times and the average values in the simulation results over the ten trials are shown.

4.2 Evaluation Criteria

To evaluate the performance, the following criteria are defined.

- Channel throughput: This represents the amount of traffic successfully transmitted between stations.
- Delay†: This is the access delay which is from the time a station starts to transmit a packet until the packet is completely sent out, i.e., the ACK is successfully received.
- Collision probability: This represents the ratio of the number of retry transmissions to the total number of transmissions.

4.3 Determination for Values of N_{suc} and Dp

First of all, to determine the values of N_{suc} and Dp, which are the parameters for THAW, the proposed THAW algorithm is evaluated in Model 1. Initially, to determine the range of N_{suc}, the value of N_{suc} is set to be fixed. The upper and lower thresholds of N_{suc} are determined by the values of N_{suc} which is calculated to achieve the best performance when the timer is 50 msec and 5 sec. Figure 2 shows the channel throughput and delay when the number of stations is 18, i.e., this is a congested state. The other cases are given in the Appendix. When the number of stations is small, i.e., an uncongested state, the probability of packet collision is low. Hence, the results in which the number of stations is 18 are used to determine the range. The timer in the ARF has the following features. The time to recover the transmission rate is short when the timer is short. On the other hand, the time to recover the transmission rate is long when the timer is long. Hence, N_{suc} must be set to a small value when the timer is short because the fall of the transmission rate is recovered within a short period. On the contrary, N_{suc} must be set to a large value when the timer is long. In Fig. 2, the delay is long when N_{suc} is small, i.e., N_{suc}=2, because many packet collisions occur. Furthermore, the delay is also long when N_{suc} is large, i.e., N_{suc}=12, because the waiting time for transmission is long. When the timer is 50 msec, Fig. 2 shows the best performance achieved in the case where N_{suc} is set to four. Additionally, when the timer is 5 sec, Fig. 2 shows that the best performance is achieved in the case where N_{suc} is set to ten. Therefore, it is verified that N_{suc} must be changed according to the value of the timer to achieve the best performance. From the results, the range for N_{suc} is set from four to ten, i.e., $4 \leq N_{suc} \leq 10$.

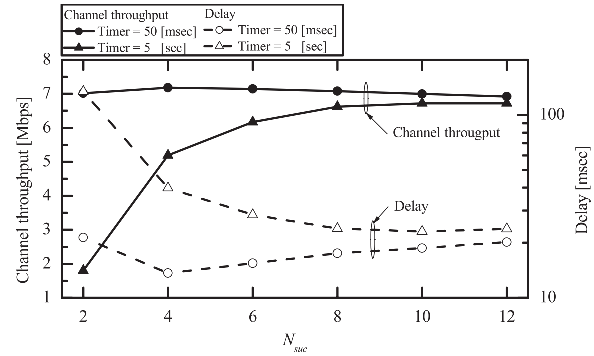


Fig. 2 Average channel throughput and delay when N_{suc} is fixed.

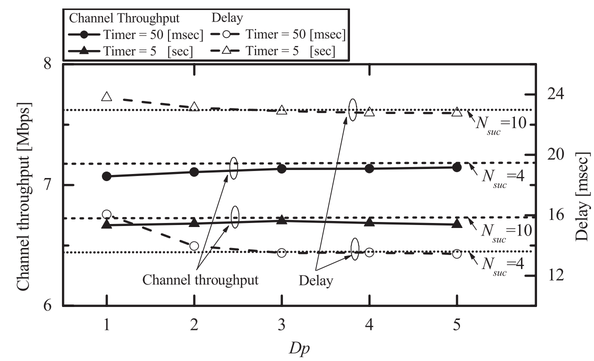
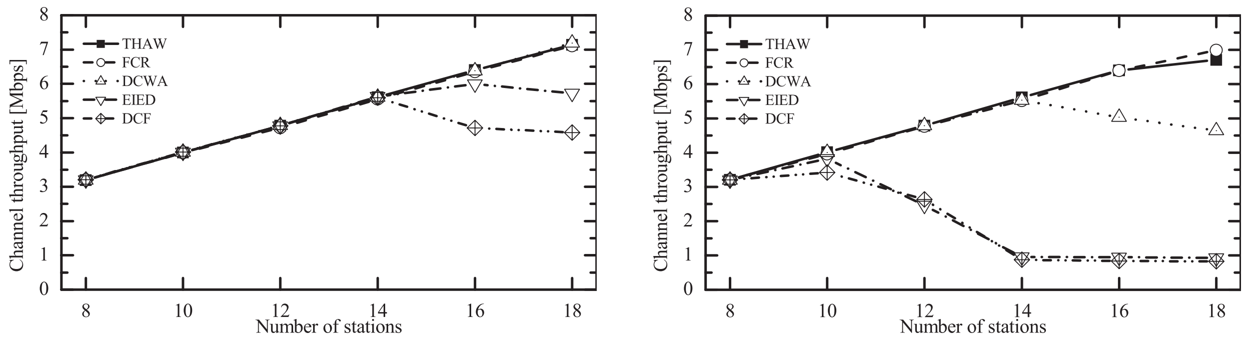


Fig. 3 Average channel throughput and delay when Dp is changed.

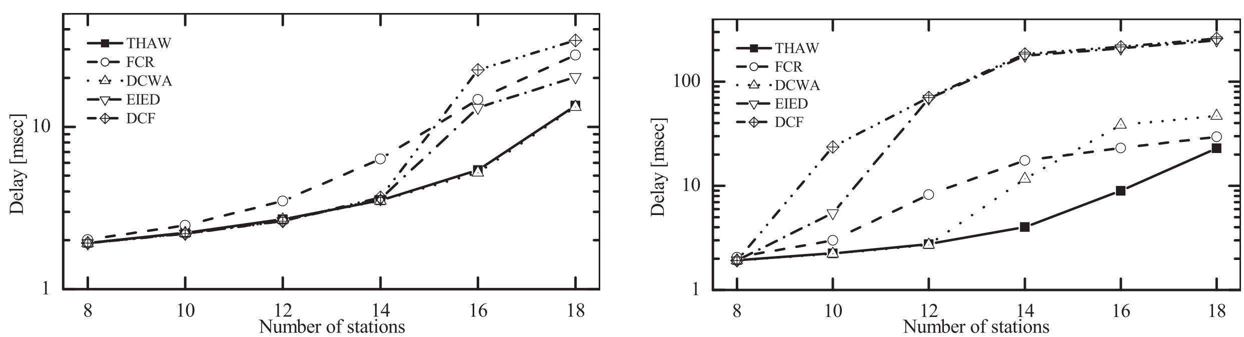
Next, to determine the value of Dp, the performance is evaluated when Dp is changed and the number of stations is 18 as with the above evaluations. The value of Dp is determined by that of Dp when the performance is the almost same as the best performance of above evaluations. Figure 3 shows the channel throughput and delay in this case. To compare the results for a fixed value of N_{suc}, Fig. 3 also

†To clarify the difference in the delay compared with conventional algorithms, this paper focuses on the access delay because the difference between those algorithms is media access control not including queuing control.



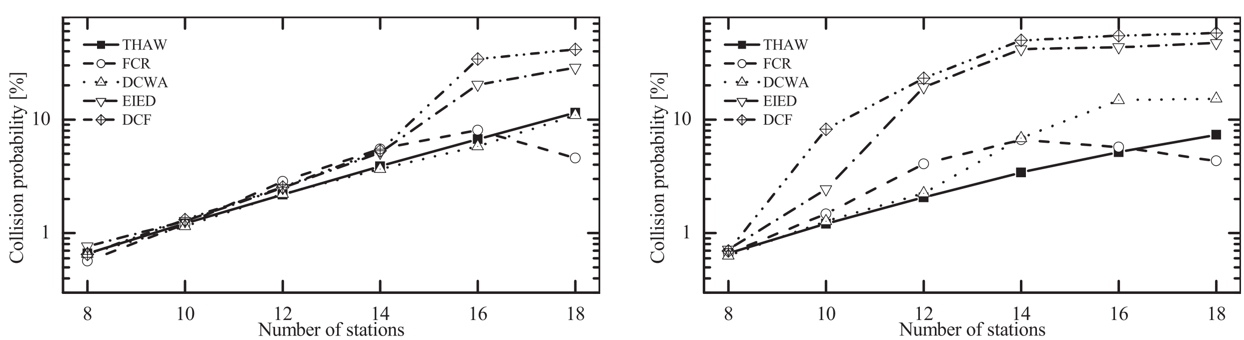
(a) Timer = 50 msec (b) Timer = 5 sec

Fig. 4 Comparison of average channel throughput in Model 1.



(a) Timer = 50 msec (b) Timer = 5 sec

Fig. 5 Comparison of average delay in Model 1.



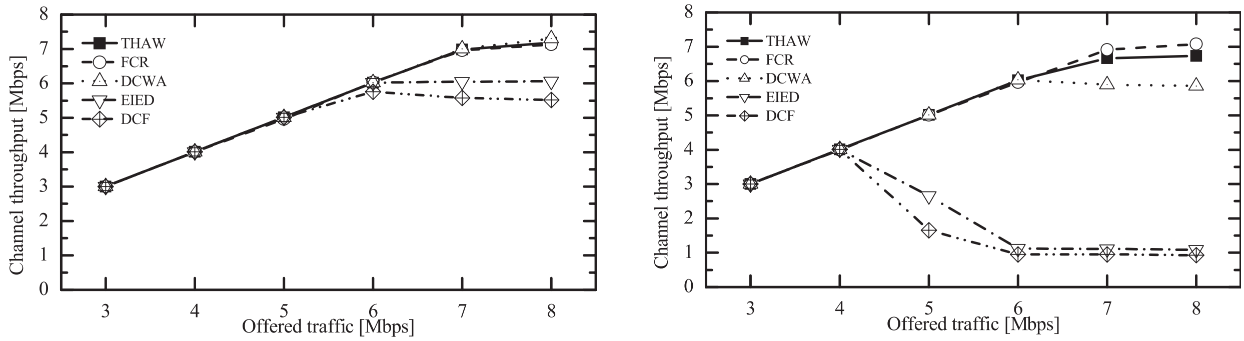
(a) Timer = 50 msec (b) Timer = 5 sec

Fig. 6 Comparison of average collision probability in Model 1.

shows the channel throughput and delay for N_{suc}=4 when the timer is set to 50 msec and those for N_{suc}=10 when the timer is set to 5 sec. These values are indicated by the straight line. As mentioned above, if the timer is a small value, N_{suc} must be a small value because the fall of the transmission rate is recovered within a short period. To reduce N_{suc}, parameter *Dp* is used. In fact, Fig. 3 shows that the delay decreases with an increase in *Dp*. More specifically, when the timer is 50 msec, the performance when *Dp* is equal to 3 or greater is almost the same as that for the fixed value of N_{suc}=4. When the timer is 5 sec, the performance when *Dp* is equal to 3 or greater is almost the same as that for the fixed value of N_{suc}=10. Therefore, *Dp* is set to 3 since the performance is saturated when the *Dp* is equal to 3 or greater. Here, the fluctuation of delay is large only when the timer is 50 msec. The reason for this is that the initial value of N_{suc} is set to the maximum value within the range of N_{suc} and the value of N_{suc} is prone to change dynamically since the timer control is invoked before the number of consecutively successful transmission reaches 10 in the ARF. On the other hand, the reason why the fluctuation of throughput when the timer is 50 msec is small is that the fluctuation of throughput is small when the N_{suc} is changed as shown in Fig. 2.

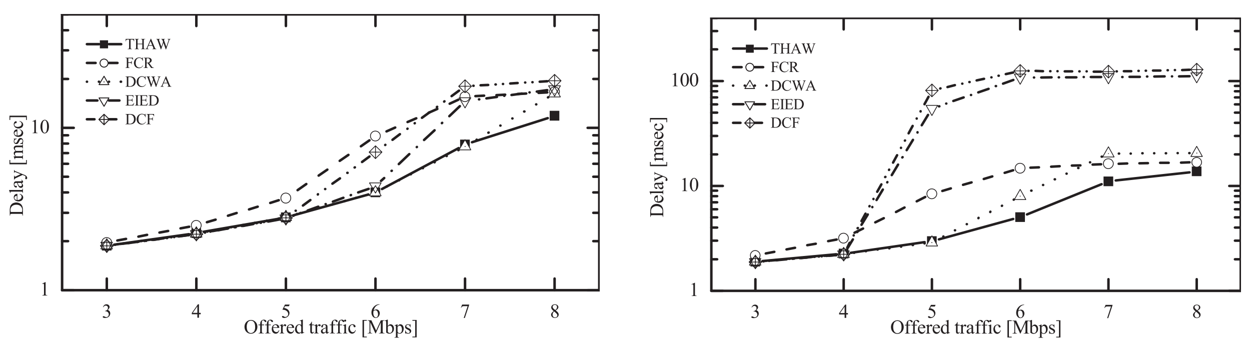
4.4 Comparison between Proposed and Conventional Algorithms

The THAW algorithm is compared to the IEEE 802.11 DCF, EIED, DCWA, and FCR. Figures 4–6 show the channel throughput, delay, and collision probability, respectively. First of all, the performance in the case where the timer



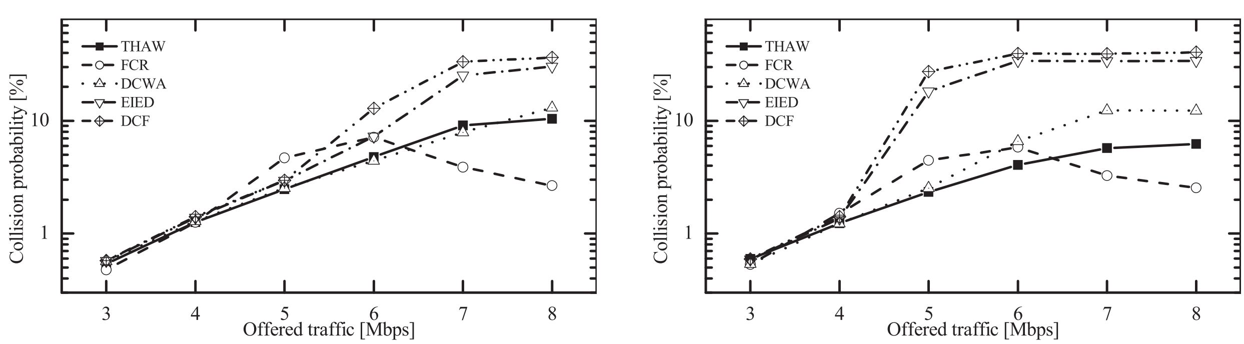
(a) Timer = 50 msec (b) Timer = 5 sec

Fig. 7 Comparison of average channel throughput in Model 2.



(a) Timer = 50 msec (b) Timer = 5 sec

Fig. 8 Comparison of average delay in Model 2.



(a) Timer = 50 msec (b) Timer = 5 sec

Fig. 9 Comparison of average collision probability in Model 2.

is 50 msec is discussed. The DCF yields the worst performance in the congested state in terms of the channel throughput and delay because CW_{init} is reset to $\mathrm{CW_{min}}$ after a successful transmission. The EIED is the second worst because CW_{init} is reduced at once after a successful transmission. Since the EIED does not immediately reset CW_{init} to $\mathrm{CW_{min}}$, the collision probability for EIED is lower than that for DCF. Hence, the performance for EIED is better than that for DCF. The FCR achieves the best channel throughput but tends to increase the delay in the uncongested state because the FCR avoids future potential collisions in advance. In addition, since the FCR reduces the backoff time exponentially fast when a pre-fixed number of consecutive idle slots is detected, the backoff time is reduced rapidly in the uncongested state. Therefore, the collision probability increases in the uncongested state because the channel tends to be idle. On the other hand, the collision probability is the lowest in the congested state, i.e., the number of stations is 18. In fact, the FCR has the ability to avoid packet collisions in mainly the congested state.

The characteristics of DCF, EIED, and FCR in the case where the timer is 5 sec are the same as those in the case where the timer is 50 msec. Although the channel throughput for THAW is slightly lower than that for FCR in the congested state when the timer is 5 sec, the delay for FCR is higher than that for THAW in the uncongested state. Considering the cases when the timer is 50 msec, the degradation for FCR is high in terms of the delay. On the other hand, the performance for THAW is almost the same as that for DCWA when the timer is 50 msec. However, the performance for DCWA is worse than that for THAW when the timer is 5 sec. In the congested state, the channel utilization rate increases because the transmission rate is decreased due to packet collisions. In addition, the time to recover a transmission rate in the case where the timer is 5 sec is longer than that in the case where the timer is 50 msec. Hence, since the lower bound of the contention window remains at a high value and the initial range of the contention window is small, the performance is degraded when the timer is 5 sec. In addition, the THAW is simple just reusing the existing functions of the wireless LAN, while the FCR and DCWA require greater modifications from the existing functions as stated above. Therefore, it is concluded that the proposed THAW algorithm achieves better performance compared to the conventional algorithms, even if the timer in ARF is changed. Specifically, the contribution of THAW is to reuse the existing functions without requiring the extra functions. This point is greatly different from conventional algorithms. Those figures turn out to show the good results even if the number of stations exceeds 14. Thus, this technique can be utilized at where many people gather, to concurrently achieve all of their usage scenarios such as exchanging information, games with local ad hoc play, and so on with small electronic equipments such as smartphones, gaming machines and electronic books.

Finally, Figs. 7–9 show the simulation results in Model 2. As in the previous evaluation, N_{suc} is set from four to ten, and Dp is set to three. These results show that the characteristics are the same as those in Model 1. Hence, the results verify that the proposed THAW algorithm achieves better performance compared to the conventional algorithm.

5. Conclusion

This paper proposed a distributed adaptive contention window adjustment algorithm based on the transmission history for wireless LANs, called the THAW algorithm. The proposed algorithm has the functions to adjust adaptively the initial contention window according to the transmission history and the timer in the ARF, to keep CW_{init} at a high value in the congested state. Simulation results showed that THAW yields better performance in terms of channel throughput and delay compared to the conventional algorithms, even if the timer in the ARF is changed. Finally, discussion of an algorithm related to the cross-layer control between the physical layer and MAC layer including failed transmissions caused by received power degradation in an environment where the distance among stations is long is a challenging issue to be addressed in the future.

References

[1] IEEE, "Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications," IEEE Std. 802.11, 2007.

[2] G. Bianchi, "Performance analysis of the IEEE 802.11 distributed coordination function," IEEE J. Sel. Areas Commun., vol.18, no.3, pp.535–547, March 2000.

[3] Y. Peng, H. Wu, S. Cheng, and K. Long, "A new self-adapt DCF algorithm," Proc. IEEE GLOBECOM 2002, vol.1, pp.87–91, Taipei, Taiwan, Nov. 2002.

[4] H. Wu, S. Cheng, Y. Peng, K. Long, and J. Ma, "IEEE 802.11 distributed coordination function (DCF): Analysis and enhancement," Proc. IEEE ICC 2002, vol.1, pp.605–609, New York, NY USA, April-May 2002.

[5] W.-K. Kuo and C.-C. Jay Kuo, "Enhanced backoff scheme in CSMA/CA for IEEE 802.11," Proc. IEEE VTC 2003 Fall, vol.5, pp.2809–2813, Lake Buena Vista, FL USA, Oct. 2003.

[6] N. Song, B. Kwak, J. Song, and L.E. Miller, "Enhancement of IEEE 802.11 distributed coordination function with exponential increase exponential decrease backoff algorithm," Proc. IEEE VTC 2003 Spring, vol.4, pp.2775–2778, Seoul, Korea, April 2003.

[7] A. Ksentini, A. Nafaa, A Gueroui, and M. Naimi, "Determinist contention window algorithm for IEEE 802.11," Proc. IEEE PIMRC 2005, vol.4, pp.2712–2716, Berlin, Germany, Sept. 2005.

[8] Y. Kwon, Y. Fang, and H. Latchman, "Design of MAC protocols with fast collision resolution for wireless local area networks," IEEE Trans. Wireless Commun., vol.3, no.3, pp.793–807, May 2004.

[9] A. Kamerman and L. Monteban, "WaveLAN-II: A high performance wireless LAN for the unlicensed band," Bell Labs Tech. J., vol.2, no.3, pp.118–133, Aug. 1997.

[10] G. Holland, N. Vaidya, and P. Bahl, "A rate-adaptive MAC protocol for multi-hop wireless networks," Proc. ACM MOBICOM'01, pp.236–251, Rome, Italy, July 2001.

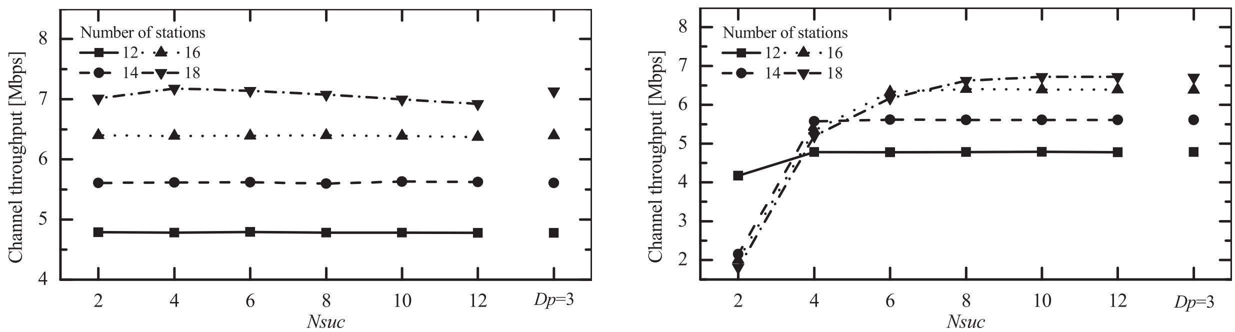
[11] C. Liu and M. Rong, "Throughput analysis of the IEEE 802.11 DCF under both saturated and non-saturated conditions," IEICE Trans. Commun., vol.E92-B, no.6, pp.2168–2174, June 2009.

Appendix

In the Appendix, two additional evaluations are given. The first is an additional evaluation where the value of N_{suc} is fixed. Since the characteristics only when the number of stations is 18 are evaluated in Fig. 2, the characteristics in the other cases are evaluated in the Appendix. The second is the evaluation based on the additional traffic models. In both Models 1 and 2 defined in Sect. 4.1, the offered traffic for each station is the same. In this Appendix, the performance is evaluated in the situation where the offered traffic for each station is different. Specifically, the offered traffic for half of the stations is 50 kbps and the other stations is 750 kbps, and the number of stations is variable. This traffic model is called Model A.

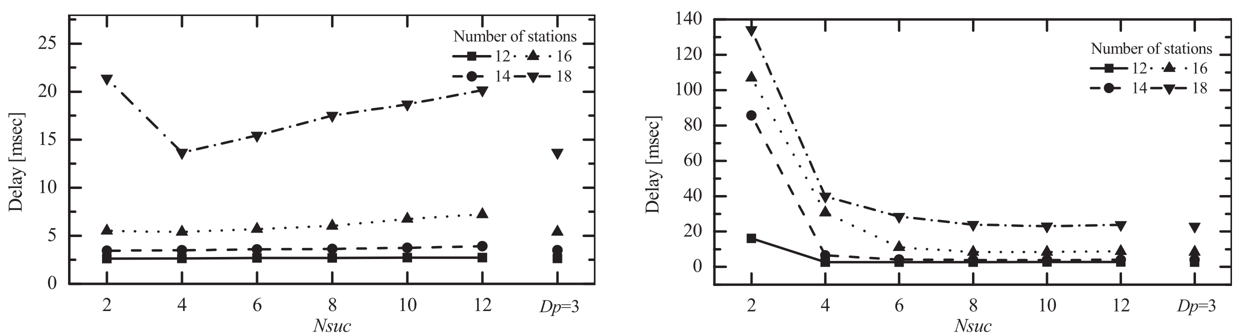
Figures A· 1 and A· 2 show the channel throughput and delay when the value of N_{suc} is fixed. In these figures, "Dp=3" represents the characteristics when the parameter N_{suc} is optimized. When the timer is 50 msec and the number of stations is small, the characteristics are almost flat because the number of opportunities to recover the transmission rate is large. On the other hand, when the timer is 5 sec, the characteristics vary irrespective of the number of stations because of the packet collisions. Since the number of opportunities to recover the transmission rate is small when the timer is a large value, the characteristics are degraded once packet collisions occur. Therefore, the value of N_{suc} in which the best performance is accomplished is changed according to the timer value, as mentioned in Sect. 4.3.

Next, Figs. A· 3 and A· 4 show the simulation results in



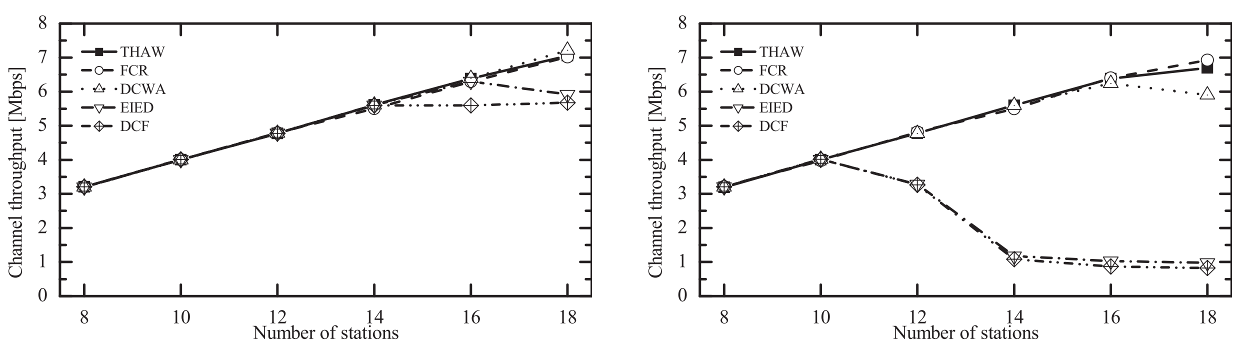
(a) Timer = 50 msec (b) Timer = 5 sec

Fig. A·1 Average channel throughput when N_{suc} is fixed.



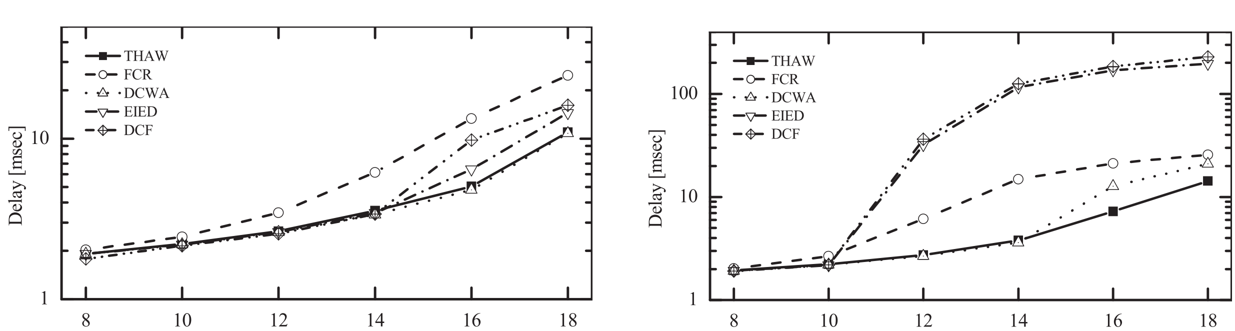
(a) Timer = 50 msec (b) Timer = 5 sec

Fig. A·2 Average delay when N_{suc} is fixed.



(a) Timer = 50 msec (b) Timer = 5 sec

Fig. A·3 Comparison of average channel throughput in Model A.



(a) Timer = 50 msec (b) Timer = 5 sec

Fig. A·4 Comparison of average delay in Model A.

Model A. In this evaluation, N_{suc} is set from four to ten, and Dp is set to three, which is the same as the evaluation described in Sect. 4.4. Although the characteristics in Model A are the same as that in Model 1, the performance is slightly improved overall. The reason for this is that the possibility of packet collisions is reduced compared to the situation in Model 1 because the number of transmission opportunity is small for half of the stations. Hence, we verified that the proposed THAW algorithm achieves better performance compared to the conventional algorithm even if the offered traffic for each station is different.

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