IEEE 802.16 환경의 레인징 부채널에서 랜덤액세스 프로토콜의 Backoff 알고리즘 성능 향상 기법

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An Improved Backoff Algorithm for the Random Access Protocol for the Ranging Subchannel of IEEE 802.16 Networks

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요 약

본 논문은 IEEE 802.16 환경의 레인징 부채널에서 사용된 OFDMA/CDMA slotted ALOHA의 랜덤 재전송의 개선된 backoff 알고리즘을 제안한다. IEEE 802.16 환경에서 기지국은 UL-/DL-MAP을 이용하여 채널 접속을 조정한다는 사실을 활용하여, 본 논문은 처리량 증가, 지연의 편차 감소 그리고 랜덤액세스 프로토콜의 과부하 조건에서 성능 감소율 향상을 위해 현재의 IEEE 802.16 환경을 약간 수정하는 방안을 제안한다. 이 알고리즘은 기본적으로 BS가 충돌이 발생한 단말들의 재전송 확률을 계산하기 위해 backlogged 사용자 수와 arrival rate를 예측한다. 컴퓨터 시뮬레이션은 제안한 알고리즘의 효과를 증명하고 Binary Exponential Backoff 알고리즘과 성능 비교를 위해 수행되었다.

Key Words : Random Access, BEB, Retransmission Probability, IEEE 802.16

ABSTRACT

An improved backoff algorithm for retransmission randomization for OFDMA/CDMA/slotted ALOHA used in the ranging subchannel of IEEE 802.16 network is proposed. Exploiting the fact that a base station coordinates channel access using UL-/DL-MAP in the IEEE 802.16 networks, we propose a minor modification of the existing IEEE 802.16 in order to increase throughput, decrease delay variation and achieve a graceful performance degradation in case of overload channel condition of the random access protocol. The algorithm basically estimates the number of backlogged users and arrival rate using which, the BS calculates retransmission probability for the subscriber stations involved in a collision. Computer simulation is performed to demonstrate the effectiveness of the proposed algorithm and to compare the performance with existing binary exponential backoff algorithm.

I. Introduction

As a promising solution of fixed broadband wireless system (BWA) providing broadband data to business or homes and an alternative to the wired "last-mile" access links using fiber, cable,

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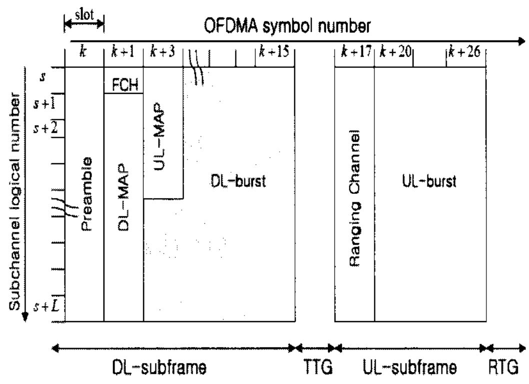

Fig. 1 OFDMA/TDD Frame Structure

or telephone lines, IEEE 802.16 wireless metropolitan area networks (MAN) has been standardized for bands from 10 to 66 GHz. It mainly assumes a point-to-multipoint (PMP) topology with a controlling base station (BS) that connects subscriber stations (SSs) to various public networks linked to the BS.

Among the various features of the physical layer in IEEE 802.16a/b/c/d/e, we focus on orthogonal frequency-division-multiplexing (OFDM) with TDD mode[1,2,3]. As shown in Fig.1, a frame of 5 msec consists of downlink (DL)- and uplink (UL)- subframe with Tx/Rx transition gap (TTG) and Rx/Tx transition gap (RTG). In the horizontal axis, each slot of DL- and UL- subframe, which respectively consists of two and three OFDM symbols with one or more subchannels is depicted, while the subchannel logical number, each of which contains 48 data subcarriers, is shown in the vertical axis. In DL- subframe, the preamble of a distinct pseudo-noise (PN) code for each BS is transmitted first and frame control header (FCH) follows for specifying the information regarding the current frame and its DL-MAP. The Dl- and UL-MAP are respectively used for delivering the down- and uplink control information for each SS. Referring to these MAPs and the following downlink channel descriptor (DCD)/uplink channel descriptor (UCD) messages, SS can respectively receive and transmit the information through DL- and UL-burst regions. For obtaining the transmission channel, i.e., the number of slots by the number of subchannels, in UL-subframe, each SS performs the bandwidth request ranging based on OFDMA-CDMA in the ranging channel. We define the slot-subchannel as one slot by six subchannels within the ranging channel in Fig.1. When a ranging channel consists of N_s slots and L subchannels, there can be total of $L_t(=N_s(L/6))$ slot-subchannels within the ranging channel. Each SS transmits a bandwidth request code on a slot-subchannel randomly selected in the ranging channel, which is called bandwidth request (BWR) ranging. The bandwidth request code of 144 bits long is a subsequence of a PN sequence of 144×256 bits long, which distinct for each BS. Among 256 subsequences, each BS uses a group of N_c codes for BWR ranging. We define a code-slot as a pair of a slot-subchannel and a code. Note that there are four code groups are used for synchronization. In the frame shown in Fig.1, the length of one slot corresponds to three OFDM symbols.

First two symbol times of a slot are used for initial- and handover ranging for an SS to synchronize the system with coarse tuning. After these procedures, an SS can tune its timing periodically with the last symbol time, in which BWR ranging is also performed.

SSs are allowed to collide on this ranging channel. To effect a ranging transmission, each SS randomly choses one ranging code from a bank of specified binary codes. In a random access protocol collisions occur when more than one SS access one code-slot at the same time. In order to avoid repeated collisions, SSs use retransmission randomization algorithm. Generally binary exponential backoff (BEB) algorithm is used for this purpose. Using the BEB algorithm an SS calculates backoff interval in such a way that system adapts to traffic load. However, the BEB algorithm has 'fairness problem' and 'fall into unstable problem' and 'waste of resource'[4], because it accesses independently of previous backoff intervals after a successful transmission. Since, there is a BS which has a global view of network traffic condition, it is desirable for the BS to estimate retransmission probability and to broadcast it to all the SSs to adjust the retransmission randomization parameter.

This paper is organized as follows. In section 2, a related research of associated retransmission probability control parameter estimation is described. We then present our proposed retransmission probability control (RTPC) algorithm in section 3. In section 4, computer simulation examples are presented and the results are compared with those of the binary exponential backoff (BEB) algorithm. Concluding remarks are given in section 5.

Ⅱ. Related Work and Our Motivation

In [5], a new recursive tracking algorithm is proposed for retransmission randomization in a slotted ALOHA system. Stability of the system is achieved by dynamically adjusting control parameters to determine the retransmission probability. This retransmission probability calculation depends on the estimated number of backlogged users which in turn is adjusted using control parameters. The algorithm is based on the observation that the optimum retransmission is achieved when the expected number of SSs attempting to access a slot (new arrival plus retransmissions) must be one. In such a case, the maximum throughput is $e^{-1} \approx 0.368$.

The retransmission probability is updated in each time slot using a combination of control parameters, using on the feedback information. To update the retransmission probability, the joint drift equation, the arrival rate, the offered load, the number of backlogged and control parameters are used in a recursive manner. Their recursive tracking algorithm uses the offered load and the control parameters in the arrival rate estimation. On the assumption that the BS has a ternary feedback on the previous slot transkission. That is, the outcome as to whether there is a success, collision or empty slot is available for control parameter optimization.

However, the BS in IEEE 802.16 network in which the random access uses CDMA code cannot distinguish whether there is a collision or there is no transmission in a code-slot. Therefore, only binary feedback is available for retransmission parameter determination. That is, information as to whether a code-slot is successful or not is available for theBS.

In order to circumvent the difficulty in estimating the number of backlogged users and the offered load, we propose to modify the IEEE 802.16 specification such that whenever, a SS is given a positive ACK after its random access is successful, it reports the number of collisions it experienced and the access delay. These are then used by the BS for retransmission probability calculation.

Ⅲ. Retransmission Probability Control (RTPC) Algorithm

3.1 Assumption for Design of RTPC Algorithm

In the ranging channel of the IEEE 802.16, random access protocol is based on OFDMA-CDMA. The BS can identify the SSs that have succeeded in ranging channel access and predict the parameters for RTPC algorithm. We assume the followings.

i. The transmission mode is the immediate first transmission (IFT) mode. That is, an SS having a packet for random access immediately accesses in the first available frame.

ii. The number of code-slots per frame, C, is fixed. It is noted that in a real implementation, the number of code-slots can be adjusted according to traffic conditions. We don't consider interference between Cs.

iii. The number of SS is large and the aggregated arrival of new packets for random access obeys Poisson process with rate λ users/frame.

iv. The BS has a data base on delay vs.

throughput of BEB algorithm. For the given minimum and maximum collision resolution window used in the BEB algorithm, $CW_{\min}$, and $CW_{\max}$, the delay vs. throughput is known by the BS. It is shown that this assumption can be relaxed without much performance degradation.

We also use the usual definition of offered load as the expected number of SSs trying to access in a frame. This includes newly arrived packets and retransmission. We also define the backlogged users as the SSs having packets to be retransmitted.

3.2 Design of RTPC Algorithm

Ranging subsystem of the IEEE 802.16 uses essentially the slotted ALOHA protocol. The throughput of slotted ALOHA is

$$S = Ge^{-1} \tag{1}$$

Because there are C code-slots per frame in the IEEE 802.16 ranging subsystem, (1) is normalized with code-slots.

$$\frac{S}{C} = \frac{G}{C}e^{-\frac{G}{C}} \tag{2}$$

The throughput of slotted ALOHA is $e^{-1} \approx 0.368$ and the offered load for maximum throughput is 1. Therefore, in this system, the normalized maximum throughput is e^{-1}, and the optimum offered load for maximum throughput is C. Since the given by offered load incudes the new arrivals and retransmissions. Offered load is

$$G = \lambda + \rho B \tag{3}$$

where ρ is retransmission probability, B is the number of backlogged users. In contrast to the BEB where the number of collisions experienced by each SS determines backoff interval. Using

$$\rho = f(k) \tag{4}$$

where k is the number of collision experienced by the SS, the retransmission probability in the proposed algorithm is the same for all the backlogged users. This is possible because the BS is the common destination of all the packets transmitted in the ranging subchannel. Thus allowing BW makes the optimal decision without wasting the information regarding channel state. We not that, in the BEB each SS makes a decision independent of other SSs and after a successful transmission, the window is reset to $CW_{\min}$ by throwing away the useful information about the channel traffic condition. The RTPC algorithm computes the retransmission probability such that the expected number of SSs accessing in a frame is 'C'. Thus, from (3), we let

$$\rho = \frac{G-\lambda}{B} \tag{5}$$

in which ρ is the optimal retransmission probability. Since λ, B, and G are unknown we use estimated values $\widehat{\lambda}_E$, $\widehat{B}_E$, and $\widehat{G}_E$ respectively for these. In order to increase the robustness of the RTPC algorithm, we use a conservative approach when the delay performance becomes worse than that of BEB. It is summarized in Table 1. In the table, D_{RTPC} is measured the mean delay of RTPC algorithm and D_{BEB} is the mean delay of the corresponding BEB algorithm. In order to estimate the arrival rate, the number of backlogged users and the offered load, we assume that users whose retransmission are successful in their random access report their number of collisions and access delays.

Table 1. The retransmission probability of RTPC algorithm, $G_{opt} = C$

Case	ρ
$\widehat{G}_E \geq G_{opt}$ or $\widehat{G}_E \leq G_{opt}$ and $D_{RTPC} < D_{BEB}$	$\frac{G_{opt} - \widehat{\lambda}_E}{\widehat{B}_E}$
$\widehat{G}_E < G_{opt}$ and $D_{RTPC} \geq D_{BEB}$	$\frac{\widehat{G}_E - \widehat{\lambda}_E}{\widehat{B}_E}$

3.3 Estimation Scheme

In this section, we give an estimation algorithm used in 3.2. In OFDMA-CDMA, the BS knows the number of successful code-slots in each frame. However it does not know how many code-slots are involved in a collision. Therefore, information for estimation is the following.

i. The number of successful SS to random access in a frame.
ii. The number of trials of each successful SS in a frame.
iii. The delay of each SS in a frame.

The arrival rate, the offered load and the number of backlogged users are estimated as follows.

3.3.1 Arrival Rate Estimation

Let S_i be the number of successful SSs in the i^{th} frame. Then, the average number of successful SSs per frame in a window of m frames measured at the end of the k^{th} frame, $\lambda_E(k)$, is given by

$$\lambda_E(k) = \frac{\sum_{i=k-m-1}^{k} S_i}{m} \qquad (6)$$

In (6), we took the average of the successful SSs from $(k-m-1)^{th}$ frame to k^{th} frame. In order to further reduce the random fluctuation we used autoregressive processing[6].

$$\widehat{\lambda_E}(k) = (1-\theta)\widehat{\lambda_E}(k-1) + \theta\lambda_E(k) \qquad (7)$$

Here θ is a smoothing factor. In case the channel is inundated with collisions and the number of successful transmission approaches zero, we set the estimated arrival rate to possible maximum value of slotted ALOHA. That is

$$\widehat{\lambda_E}(k) = C \times e^{-1} \qquad \text{if } \sum_{i=k-m-1}^{k} S_i = 0 \qquad (8)$$

The BS also estimates the offered load and the number of backlogged users. There the arrival rate estimation not only used in the computation of ρ in Table. 1, but also is used in the estimation of the number of backlogged users and the offered load. Because of the low pass filtering, there can be a significant estimation delay. It is expected that the actual arrival rate does not change abruptly in a real system under which our estimation algorithm works quite well as shown in section 4.

3.3.2 Offered Load and the number of Backlogged user Estimation

Using the estimated arrival rate, we estimate the offered load and the number of backlogged users as follows. Let T_n be the sum of transmission attempts of all the successful SSs in the n^{th} frame and D_n be the sum of access delays of all the successful SSs in the n^{th} frame.

$$G_E(k) = \frac{\widehat{\lambda_E}(K) \sum_{n=k-q-1}^{k} T_n}{\sum_{n-q-1}^{k} S_n} \qquad (9)$$

$$B_E(k) = \frac{\widehat{\lambda_E}(K) \sum_{n=k-q-1}^{k} D_n}{\sum_{n-q-1}^{k} S_n} \qquad (10)$$

The window size of q frame is used for smoothing the estimation. We further smoothen our estimation using autoregressive processing as follows:

$$\widehat{G_E}(k) = (1-\beta)\widehat{G_E}(k-1) + \beta G_E(k) \qquad (11)$$

$$\widehat{B_E}(k) = (1-\gamma)\widehat{B_E}(k-1) + \gamma B_E(k) \qquad (12)$$

Here β and γ are smoothing factors for the offered load estimation and the number of backlogged users estimation, respectively.

Ⅳ. Analysis of Simulation

The simulation environments for the examples

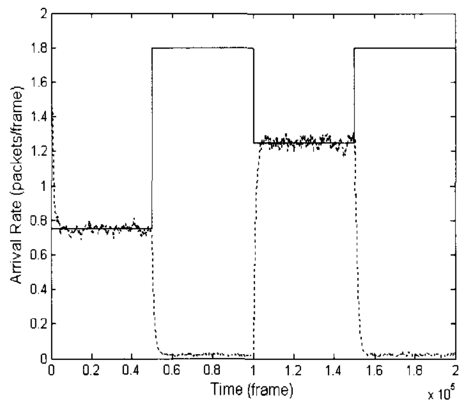

Fig. 2 Arrival rate estimation without initialization procedure (*solid line* : actual arrival rate, *dotted line* : estimated arrival rate)

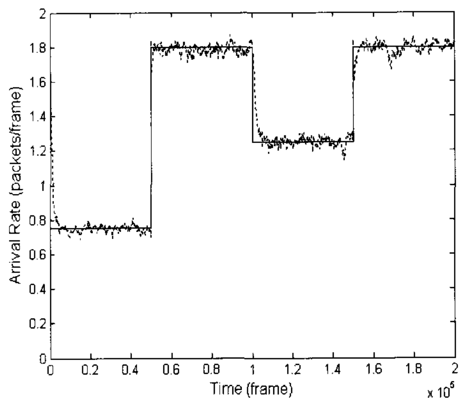

Fig. 3 Arrival rate estimation with initialization procedure (*solid line* : actual arrival rate, *dotted line* : estimated arrival rate)

analysis are as follows.

- The $CW_{\min}$ is 1 and the $CW_{\max}$ is 1024 on the BEB algorithm.
- The normalized arrival rate, arrival rate divided by the number of code-slots, C, is between 0 and e^{-1}.
- A length of a frame is 5 ms. And a frame is TDD structure.
- The simulation time is 200000 frames.
- The initial arrival rate estimation in the autoregressive processing set up e^{-1}.
- A SS has an opportunity of maximum of 20 transmissions, after which it discards the packets.
- The number of code-slots per frame fixed to 5 code-slots. ($C=5$)
- The window sizes used are : $m=50$ (arrival rate), $q=10$ (backlogged users, offered load)
- The parameter values for autoregressive processing : $\theta=0.05$ (arrival rate), $\beta=0.9$ (offered load), $\gamma=0.85$ (backlogged users)

In Fig. 2, we demonstrate the tracking capability of arrival rate estimation without the resetting component of (8). Initially the arrival rate is set to $5\times e^{-1}$ packets/frame. After small delay 열새 initialization of $5\times e^{-1}$ packets/frame, the estimated arrival rate closely follows the actual rate. However, as the arrival rate jumps abruptly to 1.8 (5×0.36) packets/frame at 5×10^4 frame time, the algorithm cannot estimate the arrival rate and gives extremely small value. As the rate returns to $5\times e^{-1}$ packets/frame, the algorithm is capable of estimating the rate.

In Fig. 3, we incorporate the resetting component of (8) and repeat the same experiment as in Fig. 2 It is seen in Fig.3 that as the sudden increase in arrival rate results is a sudden drop in successful random access transmission, which is caused in part by the inherent delay in arrival rate estimation and the subsequent sub-optimal retransmission calculation, the propose algorithm is taking a corrective measure by assuming that the arrival rate is increased to maximum possible rate. This overestimates the backlogged users and reduces the retransmission probability until the successful number of packets increases.

In Fig. 4, we plot the throughput in packets/frame vs. the normalized arrival rate in packets/codeslot/frame. The results of the BEB are shown in a solid line and those of the RTPC is shown in a dotted line. As the normalized arrival rate, λ/C increases, the throughput of both BEB and RTPC increases linearly as expected until $\lambda/C\approx 0.32$. As λ/C increases beyond approximately 0.32, the throughput decreases abruptly indicating the unstable behavior near $e^{-1}\approx 0.368$. However, the RTPC exhibits throughput increase

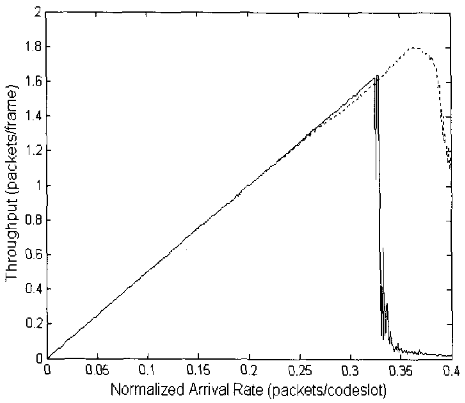

Fig. 4 Throughput vs. normalized arrival rate (*solid line* : BEB algorithm, *dotted line* : RTPC algorithm)

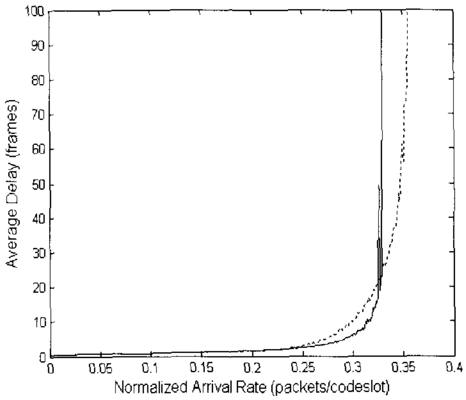

Fig. 5 Average dealy vs. normalized arrival rate (*solid line* : BEB algorithm, *dotted line* : RTPC algorithm)

until $\lambda/C \approx 0.36$ and gradual decrease beyond $\lambda/C \approx 0.38$. Even if $\lambda/C \approx 0.4$, the significant throughput greater than 0.1 (normalized throughput of 0.2) is maintained. This behavior is significant in the sence that even if there are short bursts of arrivals which could make the system unusable due to collision accumulation in the BEB algorithm, the RTPC functions satisfactorily, albeit non-optimal, and quickly restores when the bursts disappear.

Before we compare the delay performance, we note that the throughput of the RTPC is slightly smaller than that of the BEB where λ/C is between 0.25 and 0.32. This seems to be the result of estimation errors.

The average access delay vs. normalized arrival rate is shown in Fig.5. As expected from the observation of throughput performance, the average access delay of the BEB (shown in solid line) increases rapidly near $\lambda/C \approx 0.32$. The average delay of the RTPC, however, increase relatively slowly until λ/C reaches 0.35. It is also noted that the average access delay of the RTPC is slightly greater than that of the BEB for, $0.25 \leq \lambda/C \leq 0.32$. This, as explained before, seems to be mainly due to in accurate estimation of the arrival rate, offered load and the number of backlogged user. In order to see the advantage of the RTPC futher, we plot the standard deviation of the access delay vs. normalized arrival

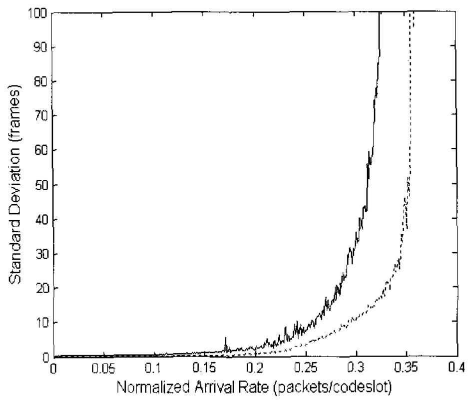

Fig. 6 Standard deviation vs. normalized arrival rate (*solid line* : BEB algorithm, *dotted line* : RTPC algorithm)

rate in Fig. 6 Because in the BEB algorithm, the access delay increases exponentially, an unfortunate SS may suffer a long delay while other SSs may have relatively short access delay. The RTPC, by making all the backlogged SSs use the same retransmission probability, can achieve a substantially smaller delay variation, thereby easing the fairness problem inherent in the BEB algorithm

In Fig. 7 and Fig. 8, we have additional simulation results comparing the throughput and the number of backlogged users. I starting with a relatively large arrival rate of $\lambda = 1.8$, we decrease λ/C every 50000 frames. While the BEB (shown in solid line) exibits unstable behavior until well after λ/C is decreased to 1.0 packets/frame, the RTPC (shown in dotted line) is capable of ach-

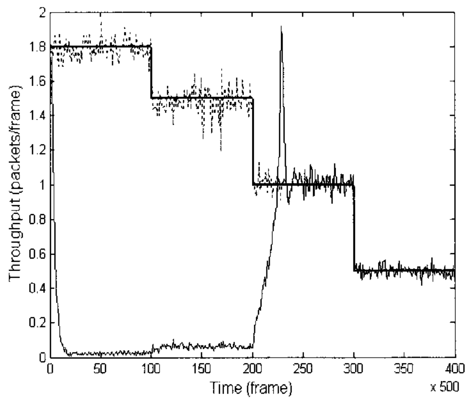

Fig. 7 SThroughput in actual arrival rate variation on stair case (*solid line(width:1.0)* : BEB algorithm, *dotted line* : RTPC algorithm, *solid line(width:1.5)* : actual arrival rate)

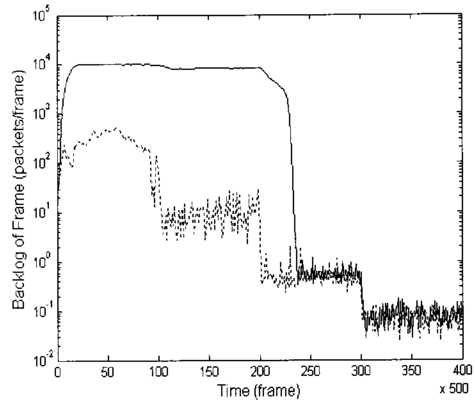

Fig. 8 SThe number of backlogged users in actual arrival rate variation on stair case (*solid line* : BEB algorithm, *dotted line* : RTPC algorithm)

ieving the desired throughputs even under a server load of $\lambda/C=0.35$ and follows closely to the applied arrival rate.

In Fig.8 shows the number of backlogged SSs. The BEB is seen to be unable to handle the initial overload of $\lambda/C=0.35$ and even when the arrival rate is decreased to $\lambda/C=0.2$ at the time of the $50{,}000^{th}$ frame, there remains a substantial number of backlogged users which do not disappear until well after λ/C is reduced to 0.2 at $100{,}000^{th}$ frame.

V. Conclusions and Discussions

In this paper, we are concerned about retransmission strategy of backlogged SSs using the ranging subchannel of IEEE 802.16 networks. Given the fact that transmission are coordinated by a BS in IEEe 802.16 networks, the performance in throughput, delay, fairness and stability can be enhanced by making the backlogged SSs to backoff their retransmissions based on the information provided by the BS. In order to calculate the retransmission probability, the proposed algorithm, RTPC, requires the arrival rate estimation, the offered load estimation and the number of backlogged users. It is shown that the resulting algorithm improves the performance compared with the BEB algorithm of the IEEE 802.16 standard. It is noted that the RTPC algorithm achieves substantially lower delay variation and exhibits graceful throughput degradation under an extreme traffic condition.

The algorithm can be implemented by making same minor modification the current IEEE 802.16 standard. They are

i) Add two fields in the ranging request message.
- Number of trials the SS have until the successful transmission.
- Delay experienced by the SS until the success transmission.

ii) Add a field of DL-MAP
- Retransmission probability to be used by the backlogged SSs.

The RTPC is shown to depend on the estimation algorithm and shows inferior performance in some range of arrival rates compared with the BEB. This can be improved by a more accurate estimation algorithm.

참 고 문 헌

〔1〕 IEEE Standard for Local and Metropolitan Area Networks, "Part16: Air Interface for Fixed Broadband Wireless Access Systems," IEEE Std 802.16-2001.

〔2〕 IEEE Standard for Local and Metropolitan Area Networks, "Part16: Air Interface for Fixed Broadband Wireless Access Systems-Amendment 2: Medium Access Control Modification and Additional Physical Layer Specification for 2-11GHz," IEEE Std 802.16a-2003 (Amendment to IEEE Std 802.16-2001).

〔3〕 IEEE Standard for Local and Metropolitan Area Networks, "Part16: Air Interface for Fixed Broadband Wireless Access Systems," IEEE P802.16-REVd/D5-2004.

〔4〕 Chang-keum Seo, Weidong Wang, Sang-Jo Yoo, "Load-based Dynamic Backoff Algorithm in Contention-based Wireless Shared Medium," *THE JOURNAL OF THE KOREAN INSTITUTE OF COMMUNICATION SCIENCES*, Vol.30, NO. 6B, pp.406~415, June, 2005.

〔5〕 Atheer Al-Amir, Mustafa K. Gurcan, "Arrival Rate Estimation Algorithm for Single Group Slotted ALOHA Systems," *IEEE COMMUNICATIONS LETTERS*, Vol.5, NO. 9, pp.387~389, Sept, 2001.

〔6〕 Jeong-Jae Won, et.al, "Performance Analysis of Contention Based Multiple Access Protocol in IEEE 802.16a Wireless MAN," *CIC2004 (The 9th CDMA International Conference)*, pp.334~338, Seoul, Oct 25~28, 2004.

〔7〕 L. Loyola, T. Miki, "An OFDMA/CDMA-based bandwidth resource allocation scheme for future broadband mobile networks," *IEEE VTC 2002-Fall*, Vol.1, pp.42~46, Sept, 2002.

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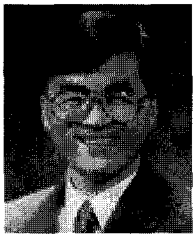
<관심분야> 통신망 설계 및 성능 분석, MAC 프로토콜

이 형 우 (Hyong-Woo Lee) 정회원



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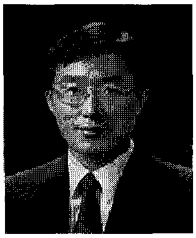

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