An Enhanced Uplink Scheduling Algorithm Based on Voice Activity for VoIP Services in IEEE 802.16d/e System

Howon Lee, Taesoo Kwon, *Student Member, IEEE,* and Dong-Ho Cho, *Senior Member, IEEE*

***Abstract*— An efficient uplink scheduling algorithm that can be used in IEEE 802.16d/e system for VoIP services is proposed to support as many voice users as possible. Our proposed algorithm requires no modification of the framework of IEEE 802.16d/e system. In this letter, we analyze the system throughput and the access delay of our proposed algorithm, and show, via OPNET simulation, that the throughput of our algorithm is superior to that of conventional algorithms in IEEE 802.16d/e system.**

***Index Terms*— Uplink scheduling algorithm, VoIP service, IEEE 802.16d/e system, UGS, rtPS.**

I. Introduction

The recent explosive growth of the Internet has given rise to demands for higher capacity, higher data rate, and more advanced multimedia services to residential and small business customers. The IEEE 802.16 standard is designed to support these demands and specifies the WirelessMAN^{TM} air interface [1], [2]. This IEEE 802.16's Task Group (TG) d/e system has many advantages, such as rapid deployment, high speed data rate, high scalability, multimedia services, and lower maintenance and upgrade costs.

Unfortunately, there is no suitable uplink scheduling algorithm for VoIP services in IEEE 802.16d/e system [1]. This system contains two uplink scheduling algorithms for real-time services, such as unsolicited grant service (UGS) and real-time polling service (rtPS), but these algorithms are not suitable for VoIP services. Therefore, we propose an efficient uplink scheduling algorithm for VoIP services. We assume that voice users in this system use a voice codec with silence detection.

II. Conventional and Proposed Algorithms for VoIP Services in IEEE 802.16d/e System

A. Conventional Algorithm

1) UGS algorithm: The UGS algorithm is designed to support real-time service flows that generate fixed-size data packets periodically [1]. The base station (BS) periodically assigns fixed-size grants to the subscriber station (SS). These fixed-size grants are sufficient to send voice data packets. The grant size and grant period are negotiated in the initialization process of the voice session. Thus, this algorithm can minimize MAC overhead and uplink access delay caused by the bandwidth request process of the SS to send the voice data packets.

However, this algorithm has only a small capacity for VoIP services. Generally, voice users do not always have voice data packets to send, because they have periods of silence [3]. In this algorithm, since the BS always assigns fixed-size grants that are sufficient to send voice data packets to the SS, it causes a waste of uplink resources.

2) rtPS algorithm: The rtPS algorithm is designed to support real-time service flows that generate variable size data packets periodically [1]. The BS assigns uplink resources that are sufficient for unicast bandwidth requests to the SS. This period is negotiated in the initialization process of the voice sessions. Generally, this process is called a bandwidth request process, or polling process. Because this algorithm always uses a bandwidth request process for suitable size grants, it transports data more efficiently than the UGS algorithm. However, this bandwidth request process always causes MAC overhead and access delay. Hence, the rtPS algorithm has larger MAC overhead and access delay than the UGS algorithm.

In the rtPS algorithm, the SS can use the piggyback requests of the grant management subheader for VoIP services. But, because VoIP services are delay-sensitive, the usage of piggyback requests is not a desirable method for VoIP services. By precise negotiation of the polling period in the initialization process, the use of piggyback requests may be avoided for VoIP services

B. Proposed Algorithm

Our proposed algorithm can solve the problems caused by the UGS algorithm (waste of uplink resources) and the rtPS algorithm (MAC overhead and access delay), because the BS basically assigns uplink resources to the SSs by considering voice state transitions of the SSs.

In our proposed algorithm, the BS has to know the voice state transitions of the SSs. When using a voice codec with a voice activity detector (VAD) or silence detector (SD), the SS can know whether its state is on or off by using a VAD or SD in the higher layer. This higher layer information can be known in the MAC layer by using primitives of Convergence Sublayer in IEEE 802.16d/e system [1]. In our algorithm, since

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The authors are with the Department of Electrical Engineering and Computer Science, Korea Advanced Institute of Science and Technology (KAIST), Daejeon, Republic of Korea (e-mail: {hwlee, tskwon80}@comis.kaist.ac.kr, dhcho@ee.kaist.ac.kr).

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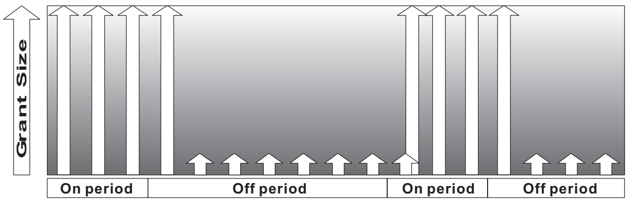

Fig. 1. Operation of BS according to GM bit in proposed algorithm

the SS has to inform the BS of its voice state transitions, it requires a method for relaying its voice status information.

The conventional generic MAC header of IEEE 802.16d/e system has two reserved bits for other additive operations. So, we use one reserved bit in this header for a method to inform the BS of the SS's voice state transitions. We define this reserved bit as a Grant-Me (GM) bit. When the voice state of the SS is 'on', the SS sets the GM bit to '1', otherwise it sets the GM bit to '0'. The SS can inform the BS of its voice state transitions effectively without MAC overhead, because it conveys the voice status information using the conventional generic MAC header. The generic MAC header is always transmitted to the BS whenever the SS has something to transmit, except the process related to the bandwidth request header of IEEE 802.16d/e system [1]. In this system, this bandwidth request header can be transmitted separately to acquire uplink resources. In general, the duration of GM bit transmission is determined by the duration of the voice codec frame used in the SS, because voice data packets are generated according to the duration of the voice codec frame.

The BS controls the grant size to provide to the SS by monitoring the GM bit. We first describe the operation of the BS according to the GM bit transmitted by the SS.

Operation of the BS:

1. In case that the GM bit is '0': The BS assigns the minimum grant size to the SS. This minimum size is sufficient in case that the SS informs the BS of its voice state transitions. In case that the GM bit is changed, '1' into '0', the BS once assigns maximum grant size to the SS whose voice state is 'off', as shown in Fig. 1. Hence, it causes a little waste of uplink resources, which could be negligible.

2. In case that the GM bit is '1': The BS assigns the maximum grant size to the SS. That size is sufficient to send voice data packets. Usually, this maximum grant size is the same as the grant size of the UGS algorithm in IEEE 802.16d/e system. In case that the GM bit is changed, '0' into '1', the BS once assigns minimum grant size to the SS whose voice state is 'on'. Thus, the SS cannot send the voice packet using this grant size. In this case, the SS can send this voice packet by using piggyback requests of the grant management subheader or bandwidth requests of bandwidth request header, as shown in Fig. 1 [1].

III. Numerical Analysis

A. System model

Voice traffic can be modeled as an exponentially distributed on-off system with mean on-time $1/\lambda\ (= T_{ON})$ and mean off-time $1/\mu\ (= T_{OFF})$ [3]. Using this voice traffic model, we can represent the system model as an one-dimensional Markov chain when considering N independent voice users, as shown in Fig. 2 [4]. In Fig. 2, each state denotes the number of the voice users in the on-state. The steady-state probability of this system model (P_N) can be represented by

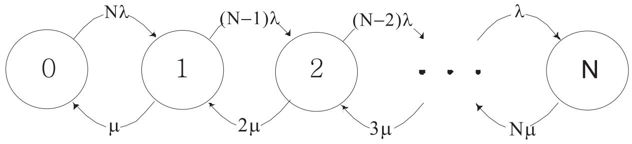

Fig. 2. System model

$$P_N(n) = \binom{N}{n} \left(\frac{T_{ON}}{T_{ON}+T_{OFF}}\right)^n \left(\frac{T_{OFF}}{T_{ON}+T_{OFF}}\right)^{N-n}. \quad (1)$$

B. Throughput

In this system model of Fig. 2, from the equation (1), the average number of voice users in the on-state is

$$\overline{N} = \sum_{n=1}^{N} n \cdot \binom{N}{n} \left(\frac{T_{ON}}{T_{ON}+T_{OFF}}\right)^n \left(\frac{T_{OFF}}{T_{ON}+T_{OFF}}\right)^{N-n} \quad (2)$$

Using the voice codec frame duration (T_{VC}), information bit per voice codec frame (L_{VC}), and compressed RTP/UDP/IP header size (L_{UH}), the system throughput can be calculated as

$$\begin{aligned} S_N &= \frac{1}{T_{VC}} \times (L_{VC} + L_{UH}) \times \overline{N} \\ &= \frac{1}{T_{VC}} \times (L_{VC} + L_{UH}) \\ &\times \sum_{n=1}^{N} n \cdot \binom{N}{n} \left(\frac{T_{ON}}{T_{ON}+T_{OFF}}\right)^n \left(\frac{T_{OFF}}{T_{ON}+T_{OFF}}\right)^{N-n} \quad (3) \end{aligned}$$

The S_N of (3) is the throughput of the system that uses the proposed algorithm for VoIP services in IEEE 802.16d/e system. Similarly, the throughput of the UGS algorithm and the rtPS algorithm can be calculated. The throughput analysis results are shown in Fig. 3.

C. Access Delay

1) UGS algorithm: The BS always provides fixed-size grants that are negotiated in the initialization process of the voice sessions to the SS. Thus, the SS can always send its voice data packets to the BS and the access delay (T_{UGS}) is the same as the MAC frame duration (T_{MF}) of IEEE 802.16d/e system. In this algorithm, the access delay of the SS is $5ms$ (T_{MF}).

Let N and N_{UGS} be the total number of users in our algorithm and the maximum number of users which can be serviced in one MAC frame duration, respectively. When the number of users $(n) > N_{UGS}$, $n - N_{UGS}$ users will have larger delays than T_{UGS}. So, we have to show that, in our proposed algorithm, there will be no problem caused by access delays.

The probability that $n > N_{UGS}$ is

$$P_N(n > N_{UGS}) = \sum_{n=N_{UGS}+1}^{N} \binom{N}{n} \left(\frac{T_{ON}}{T_{ON}+T_{OFF}}\right)^n \left(\frac{T_{OFF}}{T_{ON}+T_{OFF}}\right)^{N-n} \quad (4)$$

And the probability that $n \leq N_{UGS}$ is

$$P_N(n \leq N_{UGS}) \;=\; 1 - P_N(n > N_{UGS}). \tag{5}$$

So, the average access delay (T_{AV}) is obtained as

$$\begin{aligned} T_{AV} &= T_{UGS} \cdot P_N(n \leq N_{UGS}) \cdot \sum_{n=1}^{\infty} n \cdot P_N(n > N_{UGS})^{n-1} \\ &= T_{UGS} \cdot P_N(n \leq N_{UGS}) \cdot \frac{1}{(1 - P_N(n > N_{UGS}))^2} \quad (6) \end{aligned}$$

By Equation (5), the average access delay of (6) is described as

$$T_{AV} \;=\; T_{UGS} \cdot \frac{1}{1 - P_N(n > N_{UGS})}. \tag{7}$$

In general environment, we can prove that the access delay of our algorithm is almost the same as that of the UGS algorithm. Here, we assume an OFDMA system and define one resource unit as 6 OFDM symbols (Time Domain) and 1 OFDM subchannel (Frequency Domain). One OFDM subchannel consists of 16 OFDM subcarriers. The total uplink capacity (R_{TC}) is 40 resource units, and one resource unit (R_{BU}) is the same as eight bytes when using QPSK and 1/3 coding. For sending one voice packet, the SS uses four resource units (R_{VP}) in this environment. Also, we assume that $T_{ON} = 352ms$, $T_{OFF} = 650ms$, $T_{MF} = 5ms$, $T_{VC} = 20ms$, $L_{VC} = 22bytes$ and $L_{UH} = 2bytes$. Thus, $N_{UGS} = (\frac{R_{TC}}{R_{VP}}) \times (\frac{T_{VC}}{T_{MF}})$ = 40. By contrast, the total number of users in our proposed algorithm (N) is obtained as follows.

$$N_{UGS} \times R_{VP} \;\simeq\; N \times \left(\frac{R_{VP} \times T_{ON} + R_{BU} \times T_{OFF}}{T_{ON} + T_{OFF}}\right). \tag{8}$$

From equation (8), we have

$$N \simeq \frac{N_{UGS} \times R_{VP}}{\left(\frac{R_{VP} \times T_{ON} + R_{BU} \times T_{OFF}}{T_{ON} + T_{OFF}}\right)} \simeq 77. \tag{9}$$

Compared with the UGS algorithm, our proposed algorithm can admit more voice users. And, by equation (4), (1 - $P_N(n > N_{UGS})$) = 0.9991. Therefore, $T_{AV} \simeq T_{UGS}$.

2) rtPS algorithm: The SS always experiences the bandwidth request process by using bandwidth request header [1]. Hence, this algorithm has additional access delay of at least 1 MAC frame duration, compared with the UGS algorithm and the proposed algorithm.

In summary, the access delay of our algorithm causes no problem in view of satisfying the delay requirement of VoIP services. The access delay of our proposed algorithm is almost the same as the access delay of the UGS algorithm, and less than the access delay of the rtPS algorithm.

IV. Simulation Results and Conclusions

For simulation results, we assume the same system environment as we assumed in Section III. Further, by using header compression mechanisms, we compress the 40 byte header to 2 bytes [5].

As shown in Fig. 3, the throughput of our proposed algorithm is much larger than that of any other algorithm in IEEE 802.16d/e system. And, as shown in Fig. 4, uplink resources of our algorithm are saturated later compared with the UGS algorithm and the rtPS algorithm. These results mean that our algorithm can support more users than any other algorithm in this system. In this simulation environment, the UGS algorithm, the rtPS algorithm, and our algorithm can support 40, 66, and 77 users, respectively. In any other simulation environment, our proposed algorithm has superior performance in view of throughput and capacity. Therefore, we can say that our proposed algorithm is the best algorithm for VoIP services in IEEE 802.16d/e system.

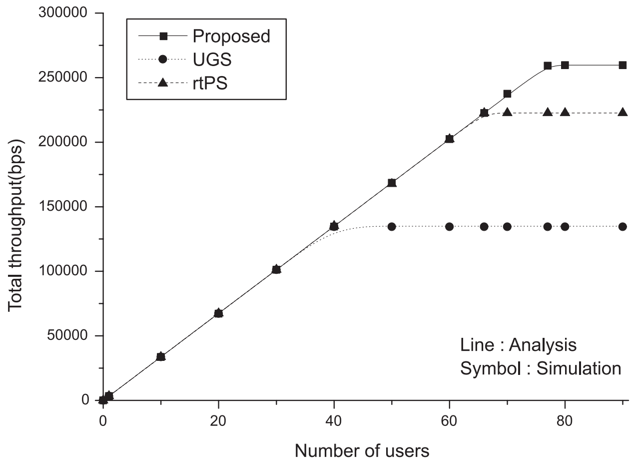

Fig. 3. Total throughput vs. users

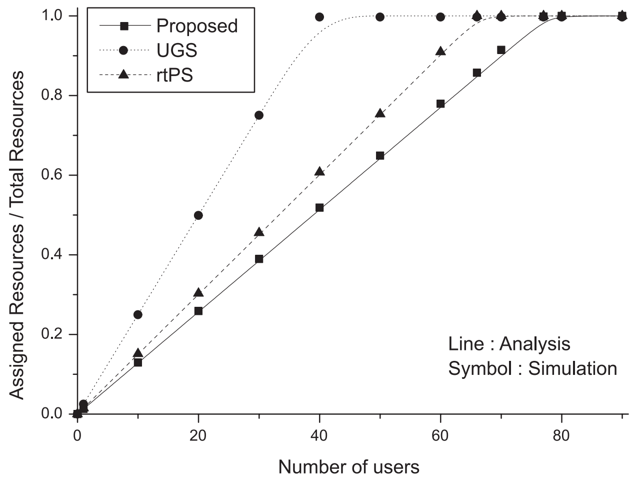

Fig. 4. Assigned resources/total resources vs. users

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