VoIP Scheduling Algorithm for AMR Speech Codec in IEEE 802.16e/m System

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***Abstract*— This letter proposes an efficient uplink scheduling algorithm for voice over Internet protocol (VoIP) services with adaptive multi-rate (AMR) speech codec in IEEE 802.16e/m systems. The proposed scheduling algorithm adopts the random access scheme during silent-period to reduce the waste of uplink bandwidth considering the characteristics of AMR speech codec. The numerical results show that the proposed algorithm can increase the maximum supportable number of voice users by 26% compared to the conventional extended real-time polling service (ertPS).**

***Index Terms*— VoIP scheduling algorithm, AMR speech codec, IEEE 802.16e/m system, ertPS.**

I. Introduction

VOIP service requires the same quality of service (QoS) level as constant bit rate services while the VoIP traffics are generated with variable bit rate. For this reason, there is a trade-off between the QoS guarantee of VoIP services and the efficient use of radio resources. To solve this problem, various scheduling algorithms have been proposed [1], [2]. Especially, IEEE 802.16e defines a new uplink scheduling method, i.e. ertPS, to support VoIP services with minimum resource usage [1]. In [2], Lee *et al.* proposed an enhanced VoIP scheduler using ertPS. This scheduler is called as Lee's algorithm in this letter. The Lee's algorithm mainly considers enhanced variable rate codec (EVRC) in VoIP services. The EVRC generates variable-size frame every 20 msec regardless of voice activity.

Recently, the IEEE 802.16 Task Group m (TGm), which is developing an amendment to IEEE 802.16e standard, published the draft evaluation methodology document which considers AMR as a strong candidate of VoIP codec [3]. The AMR speech codec generates 95 $\sim$ 477 bits speech frame every 20 msec in talk-spurt while it generates 40 bits silence descriptor (SID) frame every 160 msec in silent-period [4].

Unfortunately, the Lee's algorithm can cause the waste of uplink bandwidth and the failure of transmitting SID frames in silent-period for the VoIP services with AMR speech codec. The detailed inefficiencies of the Lee's algorithm will be presented in the following section. To overcome these inefficiencies, this letter proposes an efficient VoIP scheduling algorithm for the AMR speech codec in IEEE 802.16e/m systems.

Manuscript received December 13, 2007. The associate editor coordinating the review of this letter and approving it for publication was C. Assi. This work was supported by the IT R&D program of MKE/IITA (2008-F-015-01, Research on Ubiquitous Mobility Management Methods for Higher Service Availability), and the MIC under the ITRC support program supervised by the IITA (IITA-2008-(C1090-0801-0003)).

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Digital Object Identifier 10.1109/LCOMM.2008.072098.

II. VoIP Scheduling Algorithm for AMR Speech Codec

IEEE 802.16e defines two VoIP scheduling services such as unsolicited grant service (UGS) and ertPS. The UGS is unsuitable to support the VoIP service with silence suppression because a base station (BS) periodically assigns a fixed-size grant to a subscriber station (SS) regardless of voice activity. This inefficiency still happens in the VoIP services with AMR speech codec. On the other hand, the Lee's algorithm, which is a representative scheduler exploiting the ertPS, can efficiently manage the VoIP services with silence suppression such as EVRC while it may not efficiently manage the VoIP services with AMR speech codec. Therefore, we investigate the inefficiencies of the Lee's algorithm in the following section.

A. Inefficiencies of the Lee's algorithm with AMR speech codec

The AMR speech codec generates a speech frame every 20 msec in talk-spurt, while it generates a SID frame every 160 msec in silent-period [4], [5]. The increase of the packet transmission interval in silent-period can cause several inefficiencies of the Lee's algorithm.

The first inefficiency is the waste of uplink resources in silent-period. In the Lee's algorithm, a BS periodically allocates the minimum size grant to a SS every 20 msec in silent-period even though the AMR speech codec transmits a SID frame every 160 msec. Therefore, seven grants are always wasted every 160 msec in the Lee's algorithm.

The second inefficiency is the failure of transmitting SID frames. In the Lee's algorithm, the grant size in silent period is same with the generic-MAC header size [2]. However, it is too small to transmit a SID frame because the SID frame size is 16 bytes while the generic-MAC header size is 6 bytes. (e.g. the data packet size = SID frame size (5bytes) + compressed RTP/UDP/IP header size (3bytes) + generic-MAC header size (6bytes) + CRC (2bytes)) [3]. Moreover, the SS cannot request an additional bandwidth during silent-period in the Lee's algorithm because the SS can transmit only a generic-MAC-header which does not include the bandwidth request information. The failure of transmitting SID frames can cause the deterioration of VoIP service quality because the SID frame includes the information for the transmit side background noise [5]. However, in case of ertPS, a SS can transmit a SID frame in silent-period, because a SS can request an additional bandwidth using a bandwidth-request-and-UL-Tx-power-report header. If the Lee's algorithm is modified to use this feature of the ertPS, it can also transmit the SID frame.

B. Proposed VoIP scheduling algorithm

The proposed algorithm has three main features. Firstly, a BS does not periodically allocate a grant to a SS in silent-period to save the uplink bandwidth. Secondly, the proposed algorithm adopts the random access scheme to transmit SID frames in silent-period. Thirdly, it also uses the random access scheme when the voice activity changes from silent-period to talk-spurt, because the transition time from silent-period to talk-spurt is unpredictable. The operation of the proposed algorithm is described in detail as follows.

In case of silent-period: As depicted in Fig. 1, a BS stops the allocation of the periodic grant when the voice activity of a SS changes from talk-spurt to silent-period. To inform the BS of a voice activity, a SS uses a reserved bit in bandwidth-request-and-uplink-sleep-control (BRUSC) header defined in [1] instead of a generic-MAC header. The BRUSC header also has a bandwidth-request (BR) field which indicates the number of bytes required by the SS [1]. When the voice activity changes from silent-period to talk-spurt, the SS transmits the BRUSC header with ST bit '1' by the random access scheme. At this time, the grant-size is determined by the size of the speech frame. The BS allocates a grant to the SS at the next frame, and it periodically assigns a grant to the SS every 20 msec. On the other hand, the SS transmits the BRUSC header with ST bit '0' by the random access scheme whenever a SID frame is arrived in silent-period. The SS specifies the size of required bandwidth using a BR field in the BRUSC header. In this case, the size of required bandwidth is same with the size of a packet includes a SID frame. When a BS receives the BRUSC header with ST bit '0', the BS allocates a grant to the SS at the next frame. The grant-size corresponds to the bandwidth requested in the BR field of the BRUSC header. Consequently, the proposed algorithm enables to transmit the SID frames without unnecessary waste of uplink bandwidth in silent-period.

In case of talk-spurt: A BS periodically allocates a grant to a SS. The grant size can be variable according to the data rate of the AMR speech codec. The proposed algorithm uses a BRUSC header or grant-management subheader defined in [2] for the variable data rate in talk-spurt. When the voice activity changes from talk-spurt to silent-period, the SS sends the BRUSC header with ST bit '0' through a grant allocated by polling scheme as shown in Fig. 1. The BS assigns a grant to the SS at the next frame to transmit the SID frame.

III. Numerical Analysis and Discussion

In this section, we analyze the VoIP capacity and the average access delay according to the VoIP scheduling algorithms. In this letter, the VoIP capacity means the maximum supportable number of VoIP users, and the average access delay means the average time to transmit a VoIP packet from a SS to a BS.

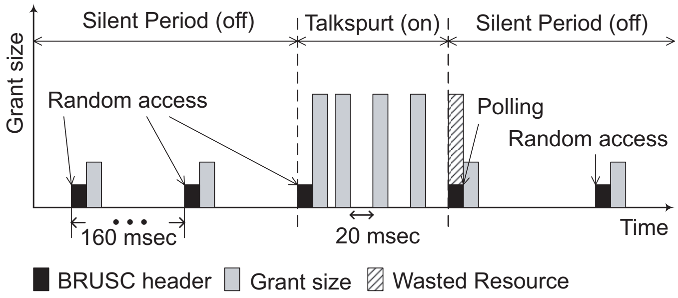

Fig. 1. The proposed VoIP scheduling algorithm with AMR speech codec.

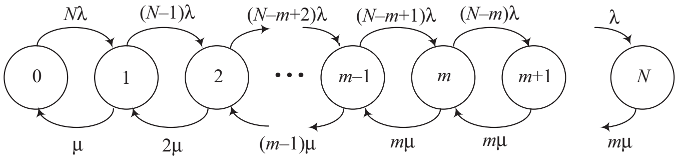

Fig. 2. Markov chain model for N independent VoIP users with exponentially distributed ON-OFF system when the VoIP capacity is m.

A. System model

We model the voice traffic as an exponentially distributed ON-OFF system with mean ON-time $1/\lambda$ and mean OFF-time $1/\mu$ [2]. Fig. 2 represents the one-dimensional Markov chain for N independent VoIP users when the VoIP capacity is m users. In Fig. 2, each state indicates the number of VoIP users in ON state. Since the sum of the whole steady-state probability is unit, the steady-state probability is derived as

$$p_N(k) = \begin{cases} \dfrac{\Psi_N(k)}{\sum_{k=0}^{m-1}\Psi_N(k) + \sum_{k=m}^{N}\Psi_N(k)\Upsilon_m(k)}, 0 \le k < m \\ \dfrac{\Psi_N(k)\Upsilon_m(k)}{\sum_{k=0}^{m-1}\Psi_N(k) + \sum_{k=m}^{N}\Psi_N(k)\Upsilon_m(k)}, m \le k \le N, \end{cases} \tag{1}$$

where $\Psi_N(k) = \left(\dfrac{\lambda}{\mu}\right)^k \begin{pmatrix} N \\ k \end{pmatrix}$ and $\Upsilon_m(k) = \dfrac{k!}{m!}m^{m-k}$.

B. VoIP capacity

In this letter, we define the unit of grant-size as the number of slots mentioned in [1]. The average number of uplink slots required every grant-interval for a VoIP user is given by

$$\bar{S}_{UGS} = S_{ON\_max}, \tag{2}$$

$$\bar{S}_{Lee} = \left(\frac{S_{ON}}{\lambda} + \frac{S_{GMH}}{\mu}\right), \tag{3}$$

$$\bar{S}_{Proposed} = \left(\frac{S_{ON}}{\lambda} + \frac{S_{SID} + S_{BRUSC}}{\mu T_{PTI}/T_{GI}}\right), \tag{4}$$

where S_{ON_max}, S_{ON}, S_{GMH}, S_{SID}, and S_{BRUSC} are the number of uplink slots required to send a maximum-size speech frame, variable-size speech frame, generic-MAC header, SID frame, and BRUSC header, respectively. S_{ON_max} is a constant in the UGS because a BS allocates a maximum-size grant to a SS in talk-spurt. However, in

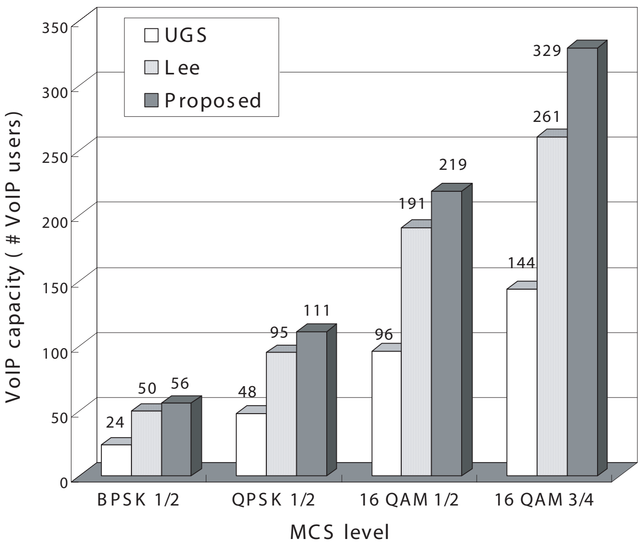

Fig. 3. VoIP capacity according to MCS levels (S_{TOT} = 144 slots, T_{MF} = 5 msec, T_{GI} = 20 msec, FFT size = 1024, and bandwidth = 10 MHz).

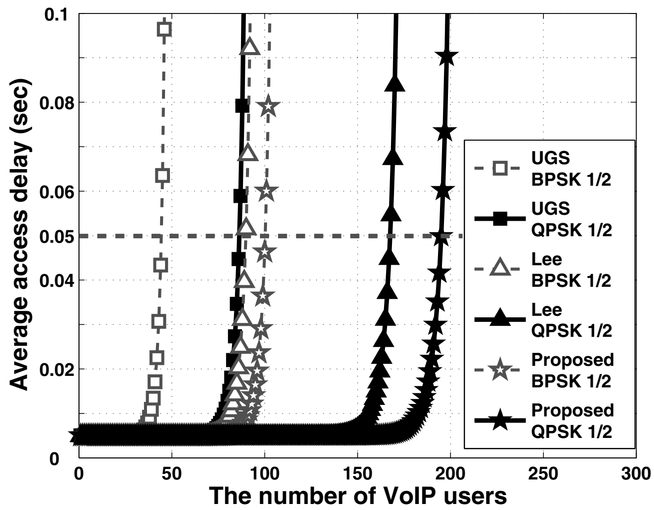

Fig. 4. The average access delay according to the number of VoIP users.

the Lee's and the proposed algorithms, S_{ON} varies during talk-spurt according to the data rate. T_{GI} and T_{PTI} indicate the grant-interval (msec) in talk-spurt and packet transmission interval (msec) of VoIP codec in silent-period, respectively. T_{PTI}/T_{GI} indicates the saved bandwidth ratio with respect to the conventional VoIP scheduling algorithms in silent-period. Using (2), (3), and (4), we can derive the VoIP capacity for each VoIP scheduling algorithm as follows.

$$m = \frac{T_{GI}}{T_{MF}} \times \frac{S_{TOT}}{\bar{S}_{Scheduler}}, \quad (5)$$

where T_{MF} is the MAC frame duration, and S_{TOT} is the total number of uplink slots in a MAC frame [2]. The $\bar{S}_{Scheduler}$ means the average number of uplink slots required every grant-interval for a VoIP user in each VoIP scheduling algorithms such as $\bar{S}_{UGS}$, $\bar{S}_{Lee}$, and $\bar{S}_{Proposed}$. In (5), the term on the right side represents the product of the number of MAC frame during packet transmission interval of VoIP codec and the maximum supportable number of VoIP users in a MAC frame.

Fig. 3 depicts the numerical result for the VoIP capacity according to modulation coding scheme (MCS) levels. Fig. 3 shows that the proposed algorithm can increase the VoIP capacity 26% and 128% compared to the Lee's algorithm and UGS, respectively. The reason is that the proposed algorithm can save the uplink resources in silent-period by adoption of the random access scheme. Note that the VoIP capacity gain of the proposed algorithm increases in proportion to MCS level because of the increase of the saved bandwidth per a slot.

C. Average access delay

Unlike the Lee's algorithm, the random access delay is added to the access delay in the proposed VoIP algorithm. The random access scheme can add an unpredictable variation to the access delay of the system. However, the maximum random access delay is about 8.3 msec [6] while the maximum tolerable delay of VoIP packets is 50 msec [3]. It means that the random access delay does not affect to the VoIP capacity. Therefore, we analyze the access delay focusing on the queuing delay in this letter.

The average queuing delay of the system exponentially increases when the number of active VoIP users becomes larger than the VoIP capacity. In this case, the probability is $p_N[k > m]$ as shown in Fig. 2. It means that some VoIP users are backlogged. Otherwise, the probability for the system without the queuing delay is $p_N[k \leq m]$, then the average access delay is given by

$$\bar{D} = T_{MF} \times \sum_{k=1}^{\infty} k \times p_N[k \leq m] \times p_N[k > m]^{k-1}. \quad (6)$$

Fig. 4 depicts the numerical result for the average access delay according to the number of VoIP users. As shown in Fig. 4, the proposed algorithm can increase the maximum supportable number of VoIP users whose average access delay is less than the maximum tolerable access delay. If the maximum tolerable access delay is assumed to be 50 msec and the MCS level is QPSK 1/2, the VoIP capacity of UGS, Lee's algorithm, and proposed algorithm are 93 users, 176 users, and 204 users, respectively. As shown in Fig. 4, it is also expected that VoIP capacity gain of the proposed algorithm increases when the MCS level is higher.

REFERENCES

[1] IEEE 802.16e^{TM}-2005, "IEEE Standard for Local and Metropolitan Area Networks - Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems Amendment 2," Feb. 28, 2006.

[2] H. W. Lee, T. S. Kwon, and D. H. Cho, "An enhanced uplink scheduling algorithm based on voice activity for VoIP services in IEEE 802.16d/e system," *IEEE Commun. Lett.*, vol. 9, pp. 691-693, Aug. 2005.

[3] R. Srinivasan, J. Zhuang, L. Jalloul, R. Novak, and J. H. Park, "Draft IEEE 802.16m evaluation methodology document," Apr. 17, 2007.

[4] 3GPP TS 26.201 V5.0.0, "AMR wideband speech codec; frame structure," Mar. 2001.

[5] 3GPP TS 26.092 V5.0.0, "AMR speech codec; comfort noise aspects," June 2002.

[6] S. M. Oh and J. H. Kim, "The analysis of the optimal contention period for broadband wireless access network," in *Proc. PWN05*, Kauai Island HA, USA, Mar. 2005, pp. 215-220.