LETTER

An Efficient Uplink Scheduling Algorithm with Variable Grant-Interval for VoIP Service in BWA Systems

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SUMMARY This letter proposes an efficient uplink scheduling algorithm for the voice over Internet protocol (VoIP) service with variable frame-duration according to the voice activity in IEEE 802.16e/m systems. The proposed algorithm dynamically changes the grant-interval to save the uplink bandwidth, and it uses the random access scheme when the voice activity changes from silent-period to talk-spurt. Numerical results show that the proposed algorithm can increase the VoIP capacity by 26 percent compared to the conventional extended real-time polling service (ertPS).

key words: *uplink scheduling algorithm, AMR speech codec, IEEE 802.16e/m system, resource management*

1. Introduction

VoIP services are considered as one of the most important services in the next generation wireless systems. VoIP service requires the same quality of service (QoS) level as constant bit rate services even though the VoIP traffic is bursty. 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. In [2], an enhanced VoIP scheduler is proposed. This scheduler mainly considers enhanced variable rate codec (EVRC) in VoIP services.

Recently, the IEEE 802.16's Task Group m (TGm), which was approved by IEEE to develop an amendment to IEEE 802.16 standard in 2006, published the draft evaluation methodology document in which the AMR speech codec is considered as a default VoIP codec [3]. The AMR speech codec generates data packets by different frame-duration according to the voice activity. In talk-spurt, AMR speech codec generates 95–477 bits speech frame every 20 msec while it generates 40 bits silence descriptor (SID) frame every 160 msec during silent-period [4]–[6]. Unfortunately, several problems prevent the application of the conventional ertPS proposed in [2] to VoIP service with AMR speech codec. The conventional ertPS can not change the grant-interval even though the AMR speech codec changes the frame-duration according to the voice activity. In the conventional ertPS, a base station (BS) periodically assigns a grant to a subscriber station (SS) every 20 msec during silent-period even though the AMR speech codec sends an SID frame every 160 msec. Therefore, the uplink bandwidth can be wasted during silent-period in the conventional ertPS. In addition, a SS can not send the SID frame to a BS in the conventional ertPS, because the grant-size in silent-period is too small to send the SID frame in the conventional ertPS. These problems are detailed in the following section. To overcome these problems, this letter proposes an efficient VoIP scheduling algorithm for the AMR speech codec.

2. Problems of the Conventional ertPS

IEEE 802.16e defines five different uplink scheduling schemes such as unsolicited grant service (UGS), ertPS, real-time polling service (rtPS), non real-time polling service (nrtPS), and best-effort (BE). Among these scheduling algorithms, the UGS and ertPS can guarantee the QoS requirements of VoIP services. However, the UGS is unsuitable to support the VoIP service with silence suppression because a BS periodically assigns the fixed-size grants to a SS even though the data rate of VoIP service is variable. On the other hand, the ertPS can efficiently manage VoIP services because a BS can periodically allocate the variable-size grants to a SS. Therefore this letter focuses on ertPS.

The AMR speech codec transmits the speech frames every 20 msec during talk-spurt, while it sends the SID frames every 160 msec during silent-period [4]–[6]. The increase of the frame-duration during silent-period may cause two problems for the conventional ertPS. Firstly, the conventional ertPS can waste the uplink bandwidth in silent-period. In the conventional ertPS, the BS periodically assigns the minimum grant-size to the SS every 20 msec during silent-period, even though the AMR speech codec transmits the SID frame every 160 msec from application layer. Therefore, seven grants are always wasted every 160 msec in the conventional ertPS as depicted in Fig. 1. Secondly, a SS can not send an SID frame during silent-period in conventional ertPS. In the conventional ertPS, a BS periodically allocates the minimum grant-size to a SS in silent-period. Here, the minimum grant-size is determined by the size of the generic-MAC header. However, the grant-size is not enough to deliver the SID frame, because the size of a data packet includ-

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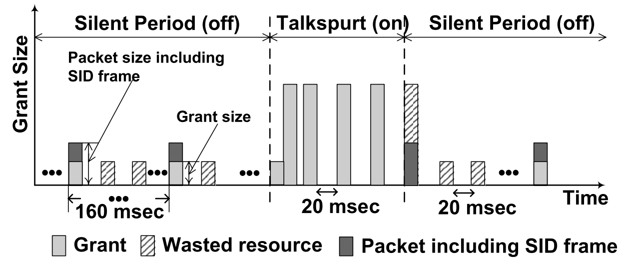

Fig. 1 The conventional ertPS with AMR speech codec.

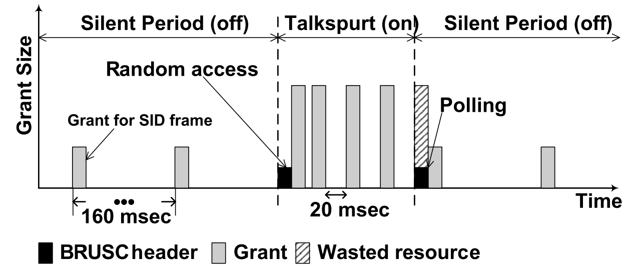

Fig. 2 The proposed VoIP scheduling algorithm with AMR speech codec.

ing the SID frame is 16 bytes while the generic-MAC header size is 6 bytes. (e.g. the data packet size = SID frame size (5 bytes) + compressed RTP/UDP/IP header size (3 bytes) + generic-MAC header size (6 bytes) + CRC (2 bytes)) [3]. Moreover, the SS can not receive an additional bandwidth to transmit the SID frame in silent-period because the SS does not have a sufficient resource for a bandwidth request.

The SID frame transmission-failure can cause the deterioration of VoIP service quality. The SID frame includes information for the transmission side background noise. If the receiver side does not receive the SID frame, it results in the discontinuity of the background noise. This effect can be very annoying for the listener in an environment with high background noise levels, because the source controlled rate (SCR) switching in a VoIP speech codec of the receiver side can take place rapidly [5].

3. Proposed VoIP Scheduling Algorithm

The proposed algorithm has two main features. The proposed algorithm adjusts the grant-interval according to the voice activity even in silent-period to save the uplink bandwidth. 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 detailed operation of the proposed algorithm is described as follows.

In case of silent-period: To inform a BS of a SS's voice activity, the proposed algorithm uses a reserved bit in bandwidth-request-and-uplink-sleep-control (BRUSC) header defined in [1] instead of the generic-MAC header. In the proposed algorithm, we define the reserved bit in a BRUSC header as the silent-period talk-spurt (ST) bit. A BRUSC header also has a bandwidth-request (BR) field which indicates the number of bytes required by a SS [1]. When the voice activity changes from talk-spurt to silent-period, a SS sends a BRUSC header with ST bit '0' by polling scheme as depicted in Fig. 2. When a BS receives the BRUSC header with ST bit '0,' the BS performs two operations. Firstly, the BS allocates a grant for an SID frame transmission at the next frame. Secondly, the BS changes the grant-interval from 20 msec to 160 msec. Thus, the BS periodically allocates a grant to the SS every 160 msec. The grant-size corresponds to the bandwidth specified in the BR field of the BRUSC header. In this case, the size of required bandwidth is same with the size of a packet includes an SID frame. Consequently, the proposed algorithm enables to transmit the SID frames without unnecessary waste of uplink bandwidth during 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 silent-period to talk-spurt, a SS transmits a BRUSC header with ST bit '1' by the random access scheme as depicted in Fig. 2. At this time, the grant-size is determined by the size of a speech frame. When a BS receives the BRUSC header, the BS allocates a grant to the SS at the next frame and it periodically assigns a grant to the SS every 20 msec.

4. Performance Analysis

In this section, we analyze the VoIP capacity and uplink utilization according to the VoIP scheduling algorithms. In addition, we analyze the packet transmission delay to show the robustness of the proposed algorithm.

4.1 VoIP Capacity

In this letter, the unit of the grant-size is defined as the number of slots. The voice traffic is modeled as an exponentially distributed ON-OFF system with mean ON-time (T_{ON}) and mean OFF-time (T_{OFF}) [7]. The sum of mean ON-time and mean OFF-time is assumed to be 1 second. The VoIP capacity means the maximum supportable number of VoIP users. The average number of uplink slots required every 20 msec for a VoIP user in each scheduler is given by

$$\bar{S}_{UGS} = S_{ON} \tag{1}$$

$$\bar{S}_{conv\_ertPS} = (S_{ON} \times T_{ON} + S_{GMH} \times T_{OFF}) \tag{2}$$

$$\bar{S}_{Proposed} = \left(S_{ON} \times T_{ON} + \frac{S_{SID} \times T_{OFF}}{T_{GIS}/T_{GIT}} \right), \tag{3}$$

where S_{ON}, S_{SID}, and S_{GMH} are the number of uplink slots required to send a speech frame, SID frame, and generic-MAC header, respectively. T_{GIT} and T_{GIS} indicate the grant-interval during talk-spurt and grant-interval during silent-period, respectively. Using (1), (2), and (3), we can derive the VoIP capacity (m) for each VoIP scheduling algorithm

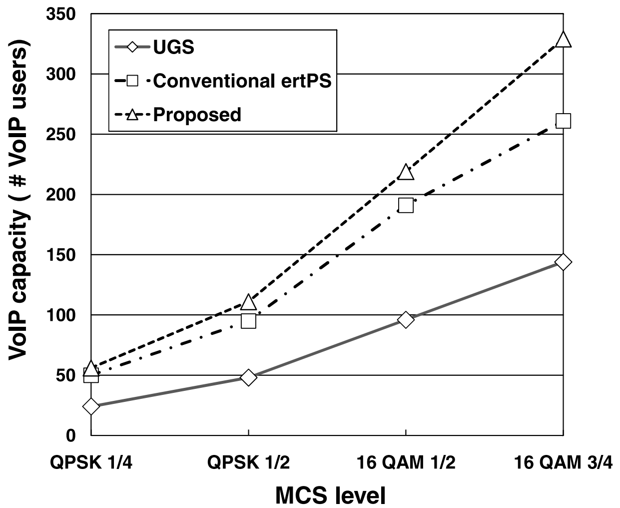

Fig. 3 The VoIP capacity according to the MCS levels. (S_{TOT} = 144 slots, T_{ON} = 400 msec, T_{OFF} = 600 msec, T_{MF} = 5 msec, T_{GIT} = 20 msec, T_{GIS} = 160 msec, FFT size = 1024, and bandwidth = 10 MHz)

as follows.

$$m = \frac{T_{GIT}}{T_{MF}} \times \frac{S_{TOT}}{\bar{S}}, \tag{4}$$

where T_{MF} is a MAC frame duration and S_{TOT} is the total number of uplink slots in a frame [2]. The $\bar{S}$ means the average number of uplink slots required every T_{GIT} for each VoIP scheduling algorithm such as $\bar{S}_{UGS}$, $\bar{S}_{conv\_ertPS}$, and $\bar{S}_{Proposed}$. In (4), the term on the right side represents the product of the number of frame during grant-interval of talk-spurt and the maximum supportable number of VoIP users in a frame. Figure 3 depicts numerical results for the VoIP capacity according to the modulation coding scheme (MCS) levels. Figure 3 shows that the proposed algorithm can increase the VoIP capacity by 26% and 128% compared to the conventional ertPS and UGS, respectively. The reason is that the proposed algorithm can save the uplink resources in silent-period by the adaptation of the grant-interval. Note that the gain of the proposed algorithm increases in proportion to MCS level because of the increase of the saved bandwidth per a slot.

4.2 Utilization

We define that the utilization is the ratio of the assigned uplink slots to the total uplink slots in a frame. Using (4), we can derive the utilization (U) for each VoIP scheduling algorithm as

$$U(N) = \frac{T_{MF}}{T_{GIT}} \times \frac{\bar{S} \times min\{N, m\}}{S_{TOT}}, \tag{5}$$

where N is the total number of VoIP users in a system. In (5), $\bar{S} \times min\{N, m\}$ indicates the average number of the required uplink slots for VoIP users in a frame, and S_{TOT} means the total number of slots in the frame. Figure 4 depicts the utilization according to the number of VoIP users. In Fig. 4,

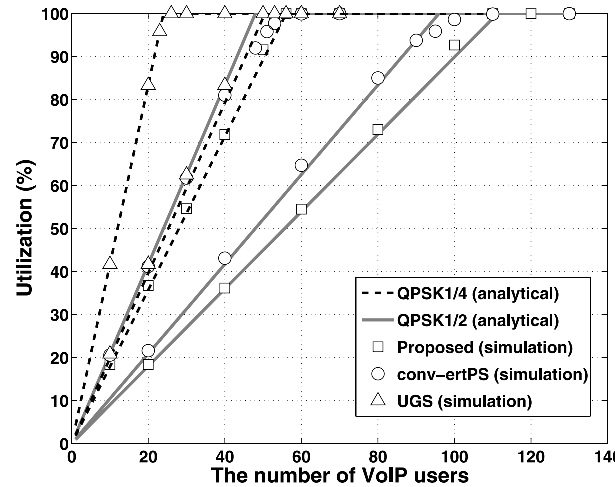

Fig. 4 The utilization according to the number of VoIP users.

lines and symbols indicate the analytical results and simulation results, respectively. The saturated point of the utilization means the VoIP capacity. As shown in Fig. 4, the VoIP capacity indicated by the utilization is very close to numerical results of Fig. 3. In addition, Fig. 4 shows that the proposed algorithm increases the VoIP capacity compared to the conventional VoIP scheduling algorithms.

4.3 Packet Transmission Delay

The packet transmission delay means the time to transmit a packet from a SS to a BS. The packet transmission delay of the proposed algorithm is different from that of UGS and conventional ertPS. In the proposed algorithm, the random access delay is especially added to the packet transmission delay.

When the voice activity changes from silent-period to talk-spurt, a SS transmits a BRUSC header to a BS by the random access scheme. In the random access scheme, the SS transmits a ranging-request (RNG-REQ) message through a ranging subchannel to obtain the radio bandwidth to transmit a BRUSC header. A RNG-REQ message includes an orthogonal ranging code randomly selected by the SS. When several SSs simultaneously choose the same orthogonal ranging code in a ranging subchannel, the SSs experience a collision. The SSs retransmit a RNG-REQ message after some delay. This random access delay may affect to VoIP capacity if the number of collision rapidly increases. Therefore, we analyze the packet transmission delay taking into consideration the random access delay in this section.

The p_{SUC} is the probability that a BRUSC header is successfully transmitted.

$$p_{SUC}(C) = \left(1 - \frac{1}{F}\right)^{C-1}, \tag{6}$$

where C indicates the number of the contending VoIP users in a frame, and F is the number of orthogonal ranging codes in a frame. The average random access delay ($\bar{D}_{RA}$) can be

Table 1 Average random access delay (msec) according to the number of contending VoIP users and orthogonal ranging codes.

	No. of ranging codes(F)		
No. of contending VoIP users(C)	150	225	300
20	5.6776	5.4416	5.3275
40	6.4903	5.9486	5.6954
60	7.4193	6.5029	6.0887
80	8.4813	7.1088	6.5092
100	9.6954	7.7712	6.9587
120	11.083	8.4952	7.4392

driven as [8]

$$\bar{D}_{RA}(C) = T_{MF} \times \sum_{k=0}^{\infty} k \times p_{SUC} \times (1 - p_{SUC})^{k-1}$$
$$= \frac{T_{MF}}{p_{SUC}} = T_{MF} \times \left(1 - \frac{1}{F}\right)^{1-C}. \quad (7)$$

Table 1 indicates numerical results for the average random access delay according to the number of contending VoIP users and orthogonal ranging codes. In Table 1, we assume that the number of ranging codes for bandwidth-requests and ranging subchannel in a frame are 75 and 2–4, respectively [1]. As shown in Table 1, the random access delay is generally smaller than 11 msec. These random access delays can be negligible for the packet transmission delay, because a SS uses the random access scheme only when the voice activity changes from silent-period to talk-spurt.

5. Conclusion

In this letter, we propose an efficient uplink scheduling algorithm with variable grant-interval for VoIP services in IEEE 802.16e/m systems. Numerical results show that the proposed algorithm can increase the VoIP capacity compared to the conventional ertPS when the AMR speech codec is applied in application layer. Therefore, the proposed algorithm is more suitable to VoIP services with variable frame-duration such as AMR speech codec than the conventional ertPS in IEEE 802.16e/m systems.

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