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Re:	This contribution is response to call for contribution about IEEE 802.16e-D4
Abstract	Extended rtPS, for VoIP services (efficient uplink scheduling for VoIP services)
Purpose	Adoption of proposed changes into P802.16-REVd/D5-2004
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Extended rtPS, for VoIP services

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1. Introduction

There are two scheduling types for real-time services in IEEE 802.16e such as UGS and rtPS. However, UGS and rtPS are not efficient in supporting VoIP service because these methods don't consider ON/OFF property of voice traffic.

1.1 Problem of UGS

Generally, voice users do not always have fixed-size voice data packets to send because its voice codec has various data rates. In case of UGS, the BS always assigns fixed-size grants that are sufficient to send voice packets. This method causes a waste of uplink resources, especially in silence - off - duration.

1.2 Problem of rtPS

In case of rtPS, the SSs comply with a bandwidth request process - polling process - using bandwidth request header (6.3.2.1.2) to transmit voice data packets. However, a polling process is not necessary during talk-spurt in VoIP services since the SS usually has voice data packets whose sizes are larger than that of packets in silence duration. Therefore, the conventional polling process causes unnecessary MAC overhead and access delay.

So, we propose an efficient uplink scheduling method considering the voice on/off property for VoIP services in IEEE 802.16e. This document describes changes suggested for 802.16e draft to support our proposed scheduling method.

2. Overview of Proposed Method

In order to reduce MAC overhead and access delay and prevent a waste of uplink resources, the proposed method assigns uplink resources according to the voice status of the SSs. In the proposed method, the SS informs the BS of its voice status information using Grant Management subheader (6.3.2.2.2) in case that its data rate of the voice codec is decreased. Then the BS reduces the polling size that is suitable for sending voice data packet generated in the decreased data rate of the voice codec of the SS. The minimum polling size is the size of voice data packet generated in the minimum data rate of the voice codec of the SS. In case that a data rate of the voice codec is increased, the SS can not send its voice data packet by using assigned polling resources. The SS has to request bandwidth to send voice data packets using Bandwidth request header (6.3.2.1.2). To inform its data rate increment, the SS sets all BR bits of the Bandwidth request header to 1. Then, the BS provides uplink resources to send voice data packets to the SS at periodic intervals according to the Maximum Sustained Traffic Rate of the service flow. In this case, the BS shall provide the first bandwidth allocation to the next MAC frame after this bandwidth request process. The second bandwidth allocation is done after the bandwidth allocation interval of the service flow based on the time which the BS allocated the bandwidth that used for the bandwidth request process using all BR bits set to 1. In case of VoIP services using our proposed method, the BS may not change its polling size without any indications or requests from the SSs, such as PBR (PiggyBack Request) bits of Grant Management subheader (6.3.2.2.2) or BR bits of Bandwidth request header (6.3.2.1.2). In other words, if the data rate of the voice codec of the SS does not alter, the BS can not change the size of polling resources for the SS. Using our method, we can obtain better data transport efficiency than those of UGS and rtPS. The general operation of our proposed method can be shown in Fig. 1. We define this method as Extended rtPS.

Whether the SS can support the Extended rtPS or not is negotiated in registration process using REG-REQ/REG-RSP messages. In case of VoIP services, the value of the Maximum Sustained Traffic Rate parameter has to set to the maximum data rate of the voice codec of the SS.

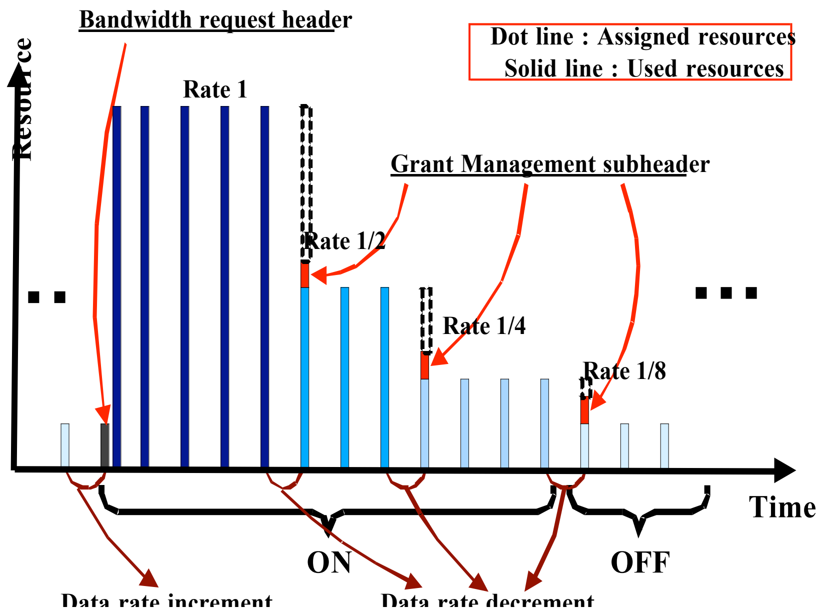

Fig. 1 Operation of the proposed method

3. Performance Analysis

3.1. Analysis Environment

A voice codec characteristic parameter and a voice on/off duration can be shown in Table 1 and Table 2, respectively.

Table 1 Voice codec parameter

	Frame Size (ms)	Data Size (bits)	
		Talk-spurt	Silence
TIA-IS-127 (EVRC)	20	171 (Rate 1, 29%) 80 (Rate 1/2, 4%) 40 (Rate 1/4, 7%)	16 (Rate 1/8, 60%)

Table 2 Voice on/off duration

Talk-spurt (on) Duration (ms)	0.352
Silence (off) Duration (ms)	0.650

3.2. Analysis results

3.2.1. UGS

- Average assigned uplink resources / voice codec frame / user
= (171 bits + 48 bits (Generic MAC header size)) * 100% = **219 bits/frame/user**

3.2.2. rtPS

We assume minimum polling size as a size of voice data packet generated in minimum data rate of the voice codec of the SS.

- Average assigned uplink resources / voice codec frame / user
= (171 bits + 48 bits (Generic MAC header size)) * 29% + (80 bits + 48 bits (Generic MAC header size)) * 4% + (40 bits + 48 bits (Generic MAC header size)) * 7% + (16 bits + 48 bits (Generic MAC header size)) * 60% + (16 bits + 48 bits) * 40% (Polling size in talk-spurt (on) duration) = **138.79 bits/frame/user**

3.2.3. Proposed method

- Average assigned uplink resources / voice codec frame / user

= (171 bits + 48 bits (Generic MAC header size)) * 29% + (80 bits + 48 bits (Generic MAC header size)) * 4% + (40 bits + 48 bits (Generic MAC header size)) * 7% + (16 bits + 48 bits (Generic MAC header size)) * 60% = **113.19 bits/frame/user**

Table 3 Average assigned uplink resources

	Average assigned uplink resources / voice codec frame (MAC frame) / user
UGS	219 bits/frame/user
rtPS	138.79 bits/frame/user
Proposed method	113.19 bits/frame/user

Average assigned uplink resources per user in one voice codec frame can be shown in Table 3. Our proposed method can save **105.81 bits** and **25.6 bits** per user compared with UGS and rtPS, respectively, as shown in Table 4. When N users use VoIP services in one MAC frame, our method can save **(105.81 * N) bits** and **(25.6 * N) bits of uplink resources** compared with UGS and rtPS, respectively. If N = 20, our method can save **2116.2 bits (264.525 bytes)** and **512 bits (64 bytes) of uplink resources**, respectively. Besides, compared with rtPS, our method does not need polling process for bandwidth requests of the SSs in talk-spurt (on) duration. Hence, our method can reduce a UL MAP size compared with that of rtPS. The general size of UL-MAP IE (8.4.5.4) is 36 bits as shown in Table 5. When N` users use only VoIP services in one MAC frame, our method can save **(36 * N`) bits of downlink resources** compared with rtPS. If N` = 20, our method can save **720 bits (90 bytes) of downlink resources**. Since UL MAP uses robust burst profile, a lot of downlink resources can be saved. These saved uplink and downlink resources could be used for other users (services). System capacity can be increased.

Table 4 Average saved resources compared with UGS

User	Average saved resources in our proposed method / voice codec frame (MAC frame)	
	Downlink (bits/frame)	Uplink (bits/frame)
1	0	105.81
10	0	1058.1
20	0	2116.2
30	0	3174.3
40	0	4232.4

Table 5 Average saved resources compared with rtPS

User	Average saved resources in our proposed method / voice codec frame (MAC frame)	
	Downlink (bits/frame)	Uplink (bits/frame)
1	36	25.6
10	360	256
20	720	512
30	1080	768
40	1440	1024

4. Proposed Text Changes

[Change the table in section 6.3.2.2.2]

Table 9 Grant Management subheader format

Syntax	Size	Notes
Grant Management subheader() {		
if (scheduling service type == UGS) {		
SI	1 bit	
PM	1 bit	
Reserved	14 bit	Shall be set to zero
}		
Else {		
PiggyBack Request	16 bit	
}		
}		

Table 10 Grant Management subheader fields

Name	Length (bits)	Description
PBR	16	PiggyBack Request The number of bytes of uplink bandwidth requested by the SS. The bandwidth request is for the CID. The request shall not include any PHY overhead. The request shall be incremental. 0b0000000000000000 = In case of the Extended rtPS, used by SS to indicate a status of data rate decrement.
PM	1	Poll-Me 0 = No action 1 = Used by the SS to request a bandwidth poll.
SI	1	Slip Indicator 0 = No action 1 = Used by the SS to indicate a slip of uplink grants relative to the uplink queue depth

In page 142, line 4, Modify the text to read:

The rtPS is designed to support real-time data streams consisting of variable-sized data packets that are issued at periodic intervals, such as moving pictures experts group (MPEG) video and Voice over IP with silence suppression. The mandatory QoS service flow parameters for this scheduling service are Minimum Reserved Traffic Rate (11.3.8), Maximum Sustained Traffic Rate (11.13.6), Maximum Latency (11.13.14), and Request/Transmission Policy (11.13.12).

In page 143, line 33, Modify the text to read:

6.3.5.2.2 rtPS

The rtPS is designed to support real-time service flows that generate variable size data packets on a periodic basis, such as moving pictures expert group (MPEG) video and Voice over IP with silence suppression. The service offers real-time, periodic, unicast request opportunities, which meet the flow's real-time needs and allow the SS to specify the size of the desired grant. This service requires more request overhead than UGS, but supports variable grant sizes for optimum data transport efficiency.

The BS shall provide periodic unicast request opportunities. In order for this service to work correctly, the Request/Transmission Policy setting (see 11.13.12) shall be such that the SS is prohibited from using any contention request opportunities for that connection. The BS may issue unicast request opportunities as prescribed by this service even if prior requests are currently unfulfilled. This results in the SS using only unicast request opportunities in order to obtain uplink transmission opportunities (the SS could still use unsolicited Data Grant Burst Types for uplink transmission as well). All other bits of the Request/Transmission Policy are irrelevant to the fundamental operation of this scheduling service and should be set according to network policy, The key service IEs are the Maximum Sustained Traffic Rate, the Minimum Reserved Traffic Rate, the Maximum Latency and the Request/Transmission Policy.

In page 143, line 55, Add a new section as shown below:

6.3.5.2.2.1 Extended rtPS

The Extended rtPS has the additional functionality of rtPS.

The Extended rtPS is designed to support for Voice over IP services with silence suppression. The Extended rtPS enabled SS can use this service by using PBR (PiggyBack Request) bits of the Grant Management subheader (6.3.2.2.2). To inform its data rate decrement the SS sets all PBR bits of the Grant Management subheader to 0. The SS shall set this flag once it detects that a data rate of its voice codec has decreased. And the Extended rtPS enabled SS can use the Bandwidth request header (6.3.2.1.2) to inform that the data rate of its voice codec has increased to the BS. To inform its data rate increment, the SS sets all BR bits of the Bandwidth request header to 1. Then, the BS provides bandwidth to send voice data packets to the SS at periodic intervals based upon the Maximum Sustained Traffic Rate of the service flow. In this case, the BS shall provide the first bandwidth allocation to the next MAC frame after this bandwidth request process. The second bandwidth allocation is done after the bandwidth allocation interval of the service flow based on the time which the BS allocated the bandwidth that used for the bandwidth request process using all BR bits set to 1.

In page 535, line 17, Add a new section as shown below:

11.7.8.9 MAC Extended rtPS support

This field indicates the availability of SS support for Extended rtPS.

Type	Length	Value	Scope
18	1	0 = No Extended rtPS support (default) 1 = Extended rtPS support	REG-REQ REG-RSP

[Change the table in section 6.3.2.1.2]

Name	Length (bits)	Description
BR	19	Bandwidth Request The number of bytes of uplink bandwidth requested by the SS. The bandwidth request is for the CID. The request shall not include any PHY overhead 0b1111111111111111111 = In case of the Extended rtPS, used by SS to indicate a status of data rate increment.
CID	16	Connection identifier
EC	1	Always set to zero.
HCS	8	Header Check Sequence Same usage as HCS entry in Table 5
HT	1	Header Type = 1
Type	3	Indicates the type of bandwidth request header