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Re:	Re: Sponsor ballot on IEEE P802.16e/D5
Abstract	Efficient uplink scheduling for VoIP services
Purpose	Adoption of proposed changes into IEEE P802.16e/D5
Notice	
Release	
Patent Policy and Procedures	

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. Generally, voice users do not always have fixed-size voice data packets to send because their voice codecs have 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.

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. We assume that the basic polling size is the same with the size for sending voice data packet generated in silence duration of the SS. However, a polling process is not necessary during talk-spurt duration in VoIP services, since the SS usually has voice data packets whose sizes are larger than the basic polling size. Therefore, the conventional polling process causes unnecessary MAC overhead and access delay in talk-spurt duration of the SS. On the other hand, if we assume that this basic polling size is the same with bandwidth request header size (6.3.2.1.2) like general rtPS services, the polling process always causes unnecessary MAC overhead and access delay in silence duration as well as talk-spurt duration.

We propose an efficient uplink scheduling method considering the voice on/off property for VoIP services and add changes that let the scheduler know codec type and coding rate. This document describes changes suggested for 802.16e draft to support proposed scheduling method.

2. Overview of Proposed Method

In order not only to reduce MAC overhead and access delay, but to prevent a waste of uplink resources, the proposed method assigns uplink resources according to the voice status of the SSs. In our method, if the SS requests the bandwidth for sending the voice data packets, then the BS shall change its polling size according to that and keeps its changed polling size until the SS sends another requests. In other words, the BS may not change its polling size without any requests from the SSs.

Firstly, the SS informs the BS of its voice status information using Grant Management subheader (6.3.2.2.2) in case that the size of the voice data packet is decreased. The SS requests the bandwidth for sending the voice data packets using PBR (PiggyBack Request) bits of Grant Management subheader (6.3.2.2.2). In our method, to distinguish these PBR bits with general PBR bits, the SS sets the MSB of PBR bits to 1. In this case, the BS assigns uplink resources according to the requested size periodically, until the SS requests another size of the bandwidth. Secondly, the SS informs the BS of its voice status information using Bandwidth request header (6.3.2.1.2) in case that the size of the voice data packet is increased. The SS requests the bandwidth for sending the voice data packets using BR (Bandwidth Request) bits of Bandwidth request header (6.3.2.1.2). In the same way with the case of Grant Management subheader, to distinguish these PBR bits with general BR bits, the SS sets the MSB of BR bits to 1. In this case, the BS assigns uplink resources according to the requested size periodically, until the SS requests another size of the bandwidth. 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 the MSB of the BR bits set to 1.

In case of VoIP services using our method, the BS recognizes Grant Management subheader and Bandwidth request header especially. If the bandwidth is requested from the SS by using PBR bits whose MSB is 1 of Grant Management subheader or BR bits whose MSB is 1 of Bandwidth request header, the BS keeps its polling size until the SS requests another size of the bandwidth. Using our method, we can obtain better data transport efficiency than those of UGS and rtPS. Fig. 1 shows the operation of the proposed method when the SS uses EVRC codec. Our proposed method can be used with any other voice codec and header compression method.

Whether the SS can support the Extended rtPS or not is negotiated in registration process using REG-REQ/REG-RSP messages.

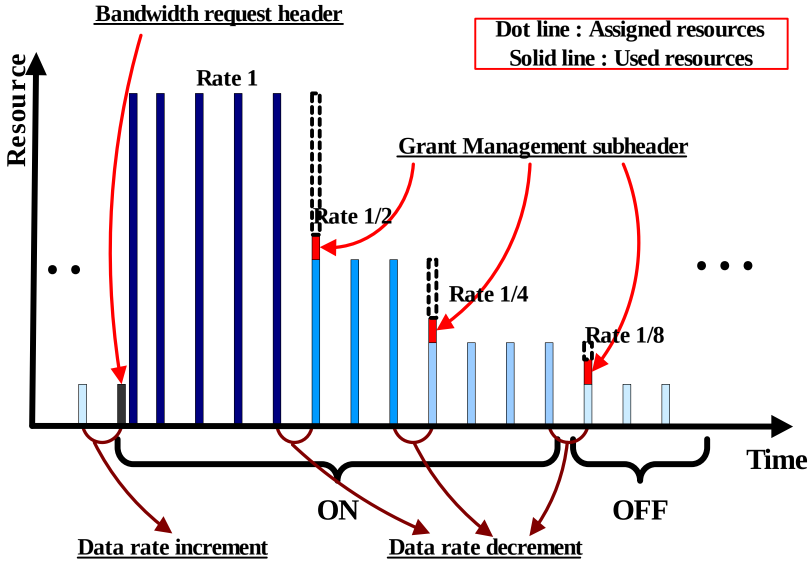

Fig. 1 Operation of the proposed method

For the proposed VoIP scheduling algorithm to work properly, convergence sublayer lets the MAC sublayer know the the coding rate of the frame. Convergence sublayer extracts the coding rate information from the Codec Mode Rate (CMR) fields in RTP payload header and sends it to the MAC sublayer through MAC_DATA.request or MAC_DATA.indication primitive. Because RTP payload header is not suppressed or compressed even if the IP/UDP/RTP header is compressed, convergence sublayer can obtain the value from the first 4-bits in RTP payload without de-suppression or de-compression of header. Fig. 2 shows an example of RTP payload header and the bit-setting of the CMR field.

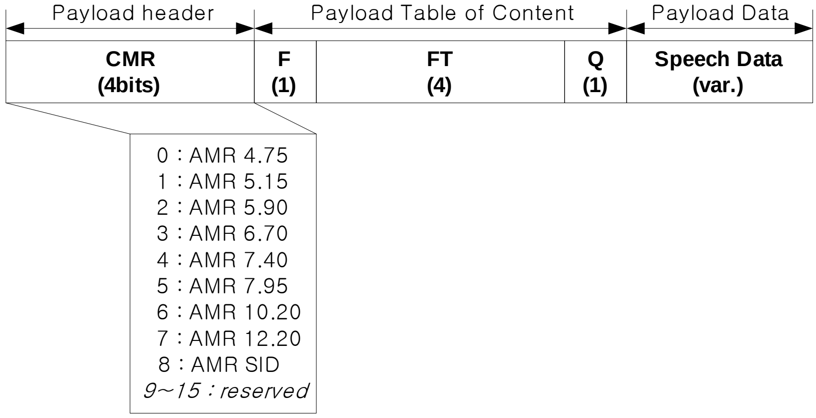

Fig. 2 Example of RTP-AMR payload header with Codec Mode Rate (CMR)-bit

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 Example of 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 Example of 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 6.3.2.1.2]

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 In case of the Extended rtPS, if the MSB is 1, the BS changes its polling size into the size requested from the SS.
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

[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. In case of the Extended rtPS, if the MSB is 1, the BS changes its polling size into the size requested from the SS.
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. Firstly, the Extended rtPS enabled SS can use this service by using PBR (PiggyBack Request) bits of Grant Management subheader (6.3.2.2.2) in case that the size of the voice data packet is decreased. The SS should request the bandwidth for sending the decreased size of the voice data packet using PBR (PiggyBack Request) bits of Grant Management subheader (6.3.2.2.2). To distinguish these PBR bits with general PBR bits, the SS should set the MSB of the PBR bits to 1. Then the BS assigns uplink resources according to the requested size periodically, until the SS requests another size of the bandwidth. Secondly, the Extended rtPS enabled SS can use this service by using BR (Bandwidth Request) bits of Bandwidth request header (6.3.2.1.2) in case that the size of the voice data packet is increased. The SS should request the bandwidth for sending the increased size of the voice data packet using BR (Bandwidth Request) bits of Bandwidth request header (6.3.2.1.2). In the same way with the case of Grant Management subheader, the SS should set the MSB of the BR bits to 1. Then the BS assigns uplink resources according to the requested size periodically, until the SS requests another size of the bandwidth. 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 the MSB of the BR bits set to 1.

In Extended rtPS, if the SS requests the bandwidth for sending the voice data packets, the BS shall change its polling size according to that and keeps its changed polling size until the SS sends another requests. In other words, the BS may not change its polling size without any requests from the SSs.

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

In page 720, line 14, Add a new section as shown below:

11.13.19.3.4.16 RTP payload type
The values of the field specify the type of codec, and its sampling frequency from RTP header.

Type	Length	Value
[145/146].cst.3.15	1	0~127: RTP payload type (PT in RTP header) 128~255: reserved

11.13.19.3.4.17 Codec mode

The values of the field specify the codec mode from RTP payload header.

Type	Length	Value
[145/146].cst.3.16	1	0~15: Codec mode (CMR in RTP payload header) 16~255: reserved

In page 856, line 42, Modify the text to read:

C.1.1.1.11 Semantics of the service primitive

The parameters of the primitive are as follows:

MAC_DATA.request
(
Connection ID,
Length,
RTP payload type,
Codec mode,
Data,
Discard-eligible flag
)

The RTP payload type indicates the type of codec (such as G.711, G.729, AMR, etc) and its sampling frequency. The Codec mode parameter specifies which coding rate is being applied now. The values for RTP payload type and codec mode shall be selected from MIME (Multi-purpose Internet Mail Extension) or standard RTP payload type.

C.1.1.1.12 MAC_DATA.indication

The parameters of the primitive are as follows:

MAC_DATA.indication
(
Connection ID,
Length,
RTP payload type,
Codec mode,
Data,
Reception status
)

The RTP payload type indicates the type of codec (such as G.711, G.729, AMR, etc) and its sampling frequency. The Codec mode parameter specifies which coding rate is being applied now. The values for RTP payload type and codec mode shall be selected from MIME (Multi-purpose Internet Mail Extension) or standard RTP payload type.