Improving ertPS Grant Allocation for VoIP Traffic in Silence Duration

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Abstract—**This paper proposes a new uplink scheduling algorithm that can increase the uplink VoIP user capacity for ertPS scheduling in 802.16e networks. This algorithm solves the problems of uplink resources wastes and large overhead which causes during silence period. We compare our method with four related schemes including conventional ertPS. In these algorithms, efficient schemes are proposed to improve the uplink resource allocation procedure for ertPS scheduling. We use OPNET simulator to investigate delay, jitter and loss in our experiments. We show that our scheme improves the capacity by 20% while the voice quality is affected minimally.**

Index Terms—**802.16, ERTPS, scheduling algorithm, OPNET, VoIP, WiMAX.**

I. INTRODUCTION

IEEE 802.16e, also referred to as Mobile WiMAX (M-WiMAX), is an IEEE standard for wireless broadband access network [1]. The main advantages of IEEE 802.16e are long range and strong quality of service (QoS) at the MAC level. The MAC layer supports convergence between several applications and services. The standard defines two basic operational modes: mesh and point-to-multipoint (PMP). In the first mode, subscriber stations (SS) can communicate with each other and with the base station (BS). In the PMP mode, the SSs are only allowed to communicate through the BS. It is anticipated that providers will use the PMP mode to connect their user to the Internet. In this case, the SSs do not send data to each other but rather communicate through the BSs. Thus, providers can control the QoS requirements of their customers.

Voice over IP (VoIP) is a technology that utilizes packet switching instead of the traditional circuit-switching to transfer voice signals. One key benefit of VoIP is the great cost savings. Therefore, in recent years, VoIP has appeared as one of the promising applications on the Internet. Mobile WiMAX technology can support mobile VoIP traffic and provides most of the main features that are necessary for and can further optimize VoIP capacity. VoIP over M-WiMAX will play a critical role in M-WiMAX deployment.

In M-WiMAX uplink, one of the main factors affecting the efficiency of VoIP traffic is the considerable resources waste during silence periods. In traditional circuit switching, during silence periods, the line is still occupied. But in packet switching, especially when VoIP Coder/Decoder (CODECs) is considered, measures such as VAD (Voice Activity Detection) / DTX (Discontinuous Transmission) / CNG (Comfort Noise Generator) are applied to reduce the allocation of resources during this period.

VoIP CODECs with VAD/DTX/CNG connection consist of an active period during talk spurts and inactive period, called silence period. During the active (talk spurt) period, VoIP frames are generated periodically. However, during the inactive (silence) period, only Silence Descriptor (SID) frames are generated at a much lower rate, and these are used to provide background noise information, SID frames size and duration vary from one codec to another, depending on the codec employed.

M-WiMAX defines three types of uplink scheduling algorithms for real time application and supporting VoIP services: Unsolicited Grant Service (UGS), real time Polling Service (rtPS) and extended real time Polling Services (ertPS). Because of the resource waste and overhead that caused by UGS and rtPS respectively, the most popular scheduling algorithms for VoIP service is still the ertPS scheduling algorithm [2]. Details of these three algorithms will be discussed in section 2.

Although VoIP uplink resource scheduling has been discussed by many recent papers as [2], only a few papers investigate the influence of the silence period on uplink resource scheduling. In [3] a reserved bit is defined as a Grant-Me (GM) bit and a new scheduling algorithm is introduced using it. Reference [4] proposed an adaptive polling scheme for real time services. Reference [5] introduces enhanced ertPS with taking into account the variation of packet size and packet generation period. In [6] a scheduling algorithm that is suitable for adaptive multi-rate (AMR) CODEC is introduced. In [7], users are divided into two categories based on average silence duration. Grant interval time scheme is selected by related category.

The rest of this paper is organized as follows. In section II, brief introduction to the existing VoIP uplink scheduling algorithms in M-WiMAX is presented. In section III our proposed algorithm and its functionality is described. In section IV, we talk about the simulation. In section V simulation results are demonstrated, and finally conclusion of our work is provided in section VI.

II. VOIP UPLINK SCHEDULING ALGORITHMS IN M-WIMAX

A. UGS Algorithm

UGS algorithm enables BS to allocate fixed-size bandwidth periodically to MS. This method allows MS to access the channel without additional signaling, UGS is only efficient for constant bit rate (CBR) applications with

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continuous fixed bandwidth requirement, and will appear inefficient for VoIP CODECS with VAD/DTX/CNG.

B. rtPS Algorithm

The rtPS algorithm is designed for real-time applications. It enables the BS to poll MS with fixed intervals so that MS can send its request of bandwidth. In case of VoIP CODECS with VAD/DTX/CNG, as the rtPS polling interval are scheduled periodically, it may accrue overhead and additional access delay during active period, and a lot of uplink resource is wasted during silence period.

C. ertPS Algorithm

The ertPS algorithm is designed to support real-time service flows that generate variable size data packets on a periodic basis, such as VoIP services with variable rate and silence suppression. In ertPS, the BS does not change the size of the periodic uplink allocations until receiving a bandwidth change request from the MS, and the MS reduces or increases the size of this uplink allocation by using the Grant Management Sub-Header (GMSH), piggybacking, or using bandwidth request header. This algorithm is suitable for EVRC[1] codec, but, in case of applying VoIP CODECs with VAD/DTX/CNG, since the ertPS during the silence period can only reduce the size of resource allocation, it continues this allocation periodically where may cause a lot of resources waste.

D. ertPS Scheduling Methods for Silence Duration

Conventional Scheme is shown in Fig. 1. In this scheme, grant interval time in silence duration, T_{alloc}, is same as that of talk spurt duration. If the grant interval time is equal to packet generation interval time no packet is queued in this method. Different grant allocation interval schemes were proposed for scheduling ertPS uplink channel in silence duration. Reference [7] proposed three methods:

1) *Maximum Interval (MI)*: In this case the BS applies the maximum increment grant interval scheme:

$$T_{alloc} = T_{\max} \tag{1}$$

In this scheme the BS will allocate grant interval every maximum allowed delay time during uplink transmission T_{max}. The value of T_{max} depends on the whole network end to end delay, Fig. 2.

2) *Random Interval (UD):* in this case the BS adopts the random grant interval scheme with uniform distribution. Grant interval is allocated randomly within [T_{max} - $2T_{min}$, T_{max} + $2T_{min}$]. Here T_{min} is the ON period grant interval time. We do not find any reason to use above range and recognize it is suitable to use [T_{min}, T_{max}] that T_{min} can be greater than ON period grant interval time.

3) *Binary Exponential Interval (BE):* in this scheme the BS applies the binary exponential increment grant interval scheme:

$$T_{alloc} = \min(2 \times T_{prev\_alloc}, T_{\max}) \tag{2}$$

In this scheme the grant interval is increased exponentially without exceeding T_{max}. See Fig. 3.

[1] Enhanced Variable Rate CODEC

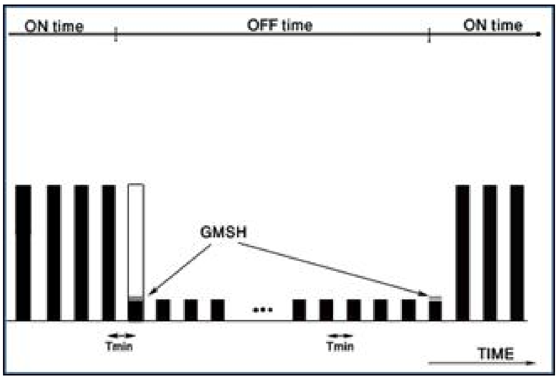

Fig. 1. Conventional ertPS: in this scheme, grant allocation intervals in silence duration is constant and equal to talk spurt duration, T_{min}.

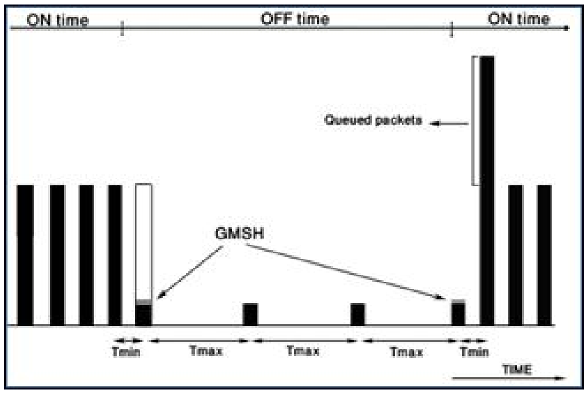

Fig. 2. Maximum grant interval: in this scheme, grant allocation intervals in silence duration is constant and larger than talk spurt duration, T_{max}.

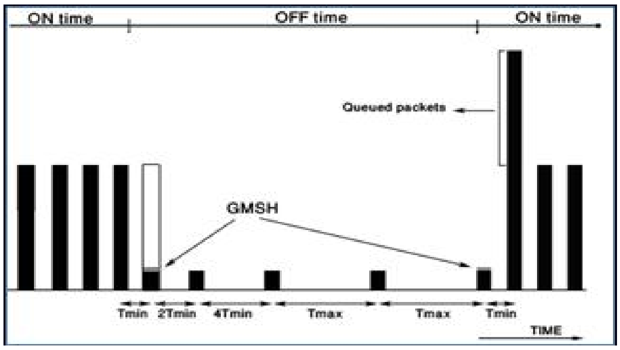

Fig. 3. Binary exponential interval: in this scheme, grant allocation intervals in silence duration is incremental and increased exponentially without exceeding T_{max}.

If the talkspurt duration starts before the next grant interval time, these schemes cause access delay for some packets, however, delayed packets are queued and transmitted within a delay less than Tmax. This delay can affect the voice quality. The aim of this paper is to define new method that improves the capacity but keeps on the voice quality.

III. Proposed Scheme

We propose a decreasing grant interval scheme. The initial value of grant interval is T_{max} and is decreased linearly if the OFF-time continues. It reaches to its lowest value T_{low} before the off period passes its average value:

$$T_{alloc} = max(T_{prev\_alloc} - \alpha, T_{low}) \tag{3}$$

The α is a decreasing step that can be equal to ON period grant interval time or other values. T_{max}, T_{low} and α are determined by statistical characteristic of voice silence duration, Fig. 4.

IV. Performance Evaluation

We use OPNET simulator in our work [12]. We modified the simulation code to implement above schemes. It involves both subscriber station (SS) and base station (BS) node models.

In our simulations, each SS sends a voice stream to BS which is then routed through a LAN to a single PC. Voice codec is G729A [13] with voice activity detection. This carried over RTP/UDP/IP protocol. We also use packet header compression to compress the IP packets. The voice frame length is 10 byte and is generated every 10ms.So its bit rate is 8kbps.Packetizing adds 40 byte as protocol header to frame,20 bytes of IP header, 8 bytes of UDP header, and 12 bytes of RTP header. The bit rate becomes 40 kbps. By compressions, it reduces to16.8 kbps.

This traffic is loaded to WiMAX network. We model the voice traffic as an exponentially distributed ON-OFF system with mean ON-time 1/λ and OFF-time 1/μ [3].

A. Simulation Scenarios

We define two scenarios for our simulations:

1) Scenario1: In this scenario, we have one SS and one BS. Our aim is to analyze the effect of changing grant interval time in silence duration (OFF-time) on the voice quality. The main effect belongs to the delayed queued packets.

We run the simulation for four different seeds. Time duration of each simulation is 250s.

2) Scenario 2*:* In this scenario, we have one BS and many SSs. The aim is to obtain the uplink capacity for VoIP users. The capacity is the maximum number of simultaneous VoIP users that can use uplink without decreasing voice quality (additional loss, delay and jitter).

We run this scenario for three different seeds. Time duration of each simulation is 250s.

For both scenarios we assume that the mean active duration T_{on}= 352ms and the mean silence duration T_{off} = 650ms, also assume that:

$$\mu = 1/T_{off} \tag{4}$$

$$\lambda = 1/T_{on} \tag{5}$$

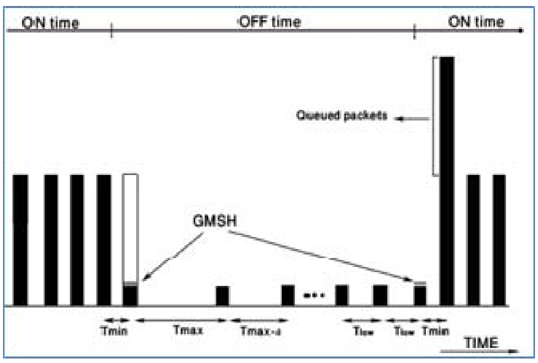

Fig. 4. Decreasing grant interval (proposed scheme): in this scheme grant allocation intervals are decreasing in silence duration in linear manner, and fix in T_{low}

The WiMAX simulation parameters are listed in Table I

B. Simulation Details

We simulate five methods for ertPS uplink grant interval in silence duration: conventional ertPS, three methods from previous works and our proposed method. For some of them we run two simulations by different parameters. The implemented methods are shown in Table II Note that in UD method we choose interval time randomly, T_{alloc} is multiple of 10 ms and $T_{low} \le T_{alloc} \le T_{max}$. Also we set the grant interval time in ON-time 10ms for all methods.

V. Simulation Results

For the first scenario we know that increasing the grant interval in silence duration increasing capacity [7]. But the delayed packets can decrease the voice quality by increasing the delay and jitter.

We assume that the jitter buffer delay (for reducing jitter effect) is 20ms.It means that the delayed packets which arrive 20ms later than other packets will be discarded.

TABLE I: WiMAX Parameters

parameter	value
PHY	OFDMA
Bandwidth	1.75MHz
Base frequency	5GHz
FFT size	512
TTG	106μs
RTG	60μs
Duplexing technique	TDD
Frame length	5 ms
UL subframe size	4symboles
UL modulation	16QAM
UL coding rate	3/4

TABLE II: Implemented Methods

Scheme	parameter
CM*	T_{alloc} = 10 ms
MI60	T_{max} = 60 ms
MI100	T_{max} = 100 ms
UD30x60	T_{low} = 30, T_{max} = 60 ms
UD50x80	T_{low} = 50, T_{max} = 80 ms
BE60	T_{max} = 60 ms
DI60x20**	T_{low} = 20 ms, T_{max} = 60 ms, α=10 ms
DI60x10	T_{low} = 10 ms, T_{max} = 60 ms, α=10 ms

*CM stands for Conventional Method.

** DI stands for Decreasing grant Interval (proposed scheme)

Table III shows the results for eight simulations. In the following, we describe each column of this table:

Average Delay: shows the mean network delay. Actually, delay is the one-way transit delay across the IP transport network from the source transport layer to destination one. Note that the transport layer protocol is RTP. This value is obtained from the first scenario.

Max Delay: is the maximum packet delay that was recorded and is obtained from the first scenario.

Loss: The loss is defined as the percent of packets which were delayed 20ms more than average. It is the play out loss and different from network loss. This value is obtained from the first scenario.

Capacity: the capacity value is calculated from the second scenario. Non integer values in this column are due to averaging different results obtained by various simulation seeds. The capacity is the maximum number of simultaneous VoIP users that can use uplink without decreasing voice quality (additional loss, delay and jitter).

Capacity Improvement*:* is relative to conventional method (CM).

As shown in TABLE III, all methods have nearly same average delay, which is about 10ms, because the number of packets with large delay is low, and only occur at the beginning of active period. On the other hand, the percentage of loss (jittered packets) is considerable. For the conventional method, we have 38 VoIP users that can have a call simultaneously without any jitter loss. The results show that the binary exponential and maximum interval schemes with T_{max}= 60ms and uniform distribution grant interval with random interval from [50, 80] ms have the most capacity, 46.5, 46.5 and 47 users, respectively. Also these methods have 22.4, 22.4 and 23.7% capacity improvements relative to the conventional method, respectively. But these methods already have high loss, 1.86, 2.04 and 2.00 %. Same is the maximum interval with T_{max} = 100ms, its loss and capacity improvement is 2.61% and 21 %. In uniform distribution grant interval with random interval from [30, 60] both loss and capacity are decreased. They are 1.46 % and 45 users. This is more efficient from previous cases, but the loss is still high.

For our proposed method, decreasing scheme, we have two rows. For the first case, Tlow = 10ms and T_{max} = 60ms, the loss is reduced to 0.37% but for the capacity improvement we have a bad result about 5%. For the second one, T_{low} = 20ms and T_{max} = 60ms, we are keeping the loss below 0.5% and achieved to 20.3% capacity improvement. We observe that comparing to previous proposed method, our method, decreasing scheme, have a good capacity improvement while the loss is kept low. Thus voice quality is not too impressed. For the proposed scheme, loss is 0.47% and capacity improves by 20%.

VI. Conclusion

In this paper, we propose an effective method to allocate the uplink resource of VoIP CODECs with voice activity detection over M-WiMAX. The capacity improvement almost is same as other methods but the strength of this approach is that the voice quality is minimally affected. Through simulation, we have demonstrated that our proposed algorithm performed better than conventional algorithm and other previously proposed methods.

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TABLE III: Results

Type	Averagedelay (ms)	MAX delay (ms)	Loss (%)	capacity	Capacity improvement%
CM	9.82	19.80	0	38	0
BE60	10.65	74.36	1.864	46.5	22.4
MI60	10.85	69.55	2.041	46.5	22.4
DI60x10	10.20	62.14	0.372	40	5.3
DI60x20	10.48	68.45	0.469	45.7	20.3
UD30x60	10.31	64.02	1.465	45	18.4
UD50x80	10.64	74.16	2.001	47	23.7
MI100	11.03	113.71	2.612	46	21.0

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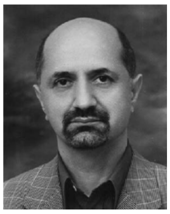

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