Performance Analysis of Scheduling Algorithms For VoIP Services in IEEE 802.16 Systems

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Abstract

VoIP applications are being widely used in today's networks challenging their capabilities to provide a good quality of experience level to the users. In particular, There are several scheduling algorithms for Voice over IP (VoIP) services in IEEE 802.16 systems, such as unsolicited grant service (UGS), real-time polling service (rtPS), using Grant-Me bit of the generic MAC header. However, these algorithms have some problems of a waste of uplink resources, additional access delay, and MAC overhead for supporting VoIP services with variable data rates and silence suppression. To solve these problems, we propose a novel uplink-scheduling algorithm (Extended-rtPS) for the VoIP services in IEEE 802.16 systems through the performance analysis and simulation results of resource utilization, total throughput, and packet transmission delay, we show that our proposed algorithm can solve the problems of the conventional algorithms, and has the best performance among these algorithms.

Keywords: *Worldwide interoperable Microwave Access (WiMAX); Voice over Internet Protocol (VOIP); Optimized Network Engineering Tools (OPNET) and Quality of Service (QoS)*

1. INTRODUCTION

VoIP (Voice over Internet Protocol) is widely deployed technology and telecommunication operators seek to profit from it. The main advantage of this technology is utilization of existing infrastructure in the form of internet connection. The usage of this type of communication is very cost effective. Unfortunately, this advantage brings someweak points that are expectable due to the low quality of internet connection. The Quality of Service (QoS) is mostly monitored issue by telecommunication operators and vendors. The quality of speech affected by many factors such as packet loss, packet delay, jitter, echo, noise [1], harmonic and inharmonic distortion [2], etc.

The QoS parameters are closely connected; with user’s satisfaction with a received speech quality. The user’s satisfaction expressed with a subjective listening score that is a result of subjective listening tests. The substitutions to the subjective tests are objective methods. The objective methods assess the speech based on a signal processing without need of real listeners. PESQ method (Perceptual Evaluation of Speech Quality) is mostly used objective method for evaluation of the speech quality [3].

The IEEE 802.16 standard for BWA, WiMAX forum promise to offer high data rate over large areas to a large number of users where broadband is unavailable. Use this first industry wide standard can for fixed wireless access with substantially higher bandwidth than most cellular networks [4]. WiMAX (IEEE802.16) technology ensures broadband access for the last mile up to 30 miles (50 km) for fixed stations, and 3 - 10 miles (5 - 15 km) for mobile stations. It provides a wireless backhaul network that enables high speed Internet access to residential, small and medium business customers, as well as Internet access for Wi-Fi hot spots and cellular base stations [5].

There are five scheduling algorithms to support variable requirements of QoS in IEEE 802.16e systems, such as unsolicited grant service (UGS), real-time polling service (rtPS), extended real-time polling service (ertPS), non-realtime polling service (nrtPS), and best effort service (BE). The UGS, rtPS and ertPS algorithms are designed to support real-time services. So, these algorithms could be used for VoIP services. However, because the nrtPS and BE algorithms are designed to support non-realtime services, these algorithms are not suitable for VoIP services in IEEE 802.16 systems. Although IEEE 802.16 systems have the UGS and rtPS algorithms for VoIP services, these algorithms are not proper as an uplink scheduling algorithm for VoIP services with variable data rates and silence suppression [6]. That is, these algorithms have some problems for the VoIP services, such as the waste of uplink resources in the UGS algorithm and MAC overhead as well as additional access delay in the rtPS algorithm. Therefore, for efficient VoIP services, ertPS is proposed and accepted as a new scheduling algorithm in IEEE 802.16 systems. In this paper, we analyze and discuss the performance of these scheduling algorithms (UGS, rtPS and ertPS) in IEEE 802.16 systems.

The purpose of this study was to examine a case of QoS deployment over a WiMAX network and to examine the capability of a WiMAX network to deliver adequate QoS to voice application. The methodologies taken include creating the WiMAX network using OPNET, carried out extensive simulations to analyze the MOS, packet end-to-end delay, jitter and packet delay variation, propose an improved ertPS scheduling algorithm based on channel state for VoIP services in IEEE 802.16 system. Conclusions are drawn for an IEEE 802.16 wireless system working in Point-to-Multipoint (PMP) mode, with Time Division Duplex (TDD). The target air interface is Wireless MAN-OFDM, based on Orthogonal Frequency Division Multiplexing (OFDM).

The rest of the paper organized as follows. Section (2) gives background of IEEE 802.16 MAC Protocol. Section (3) deals with UPLINK SCHEDULING ALGORITHMS, Section (4) deals with the Performance Metrics, Section (5) deals with Results and Discussion. Finally, in Section (6) we conclude this paper and future work.

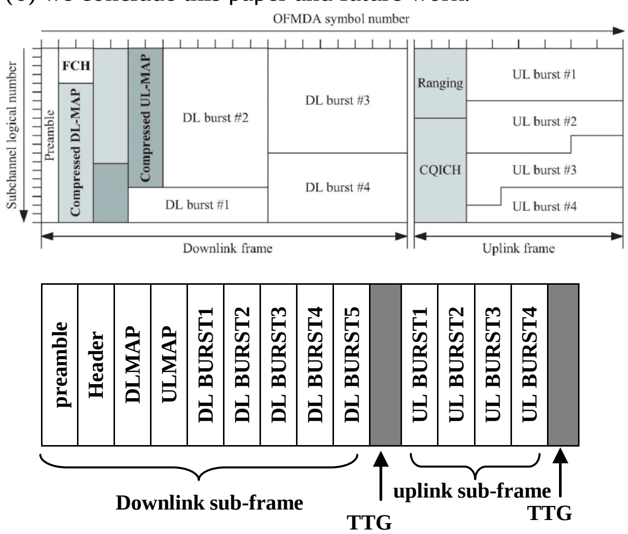

Fig.1 frame structure

2. IEEE 802.16 MAC Protocol

PMP architecture, which consists of one BS managing multiple SSs. Transmissions between the BS and SSs are realized in fixed-sized frames by means of time division multiple access (TDMA) / time division duplexing (TDD) mode of operation. the frame structure consists of downlink sub-frame for transmission from the BS to SSs and an uplink sub-frame for transmission in the reverse direction as shown in fig 1. The Tx/Rx transition gap (TTG) and Rx/Tx transition gap(RTG) shall be inserted between the sub-frames to allow terminals to turn around from reception to transmission and vice versa. In the downlink sub-frame the Downlink Map (DL-MAP) and Uplink Map (UL-MAP) message are transmitted by the BS, which comprise the bandwidth allocation for data transmission in both downlink and uplink direction, respectively [7].

There are five types of basic services described in the standard. Namely, Unsolicited Grant Service (UGS); Real-Time Polling Service (rtPS); Non-Real-Time Polling Service (nrtPS); Extended-Real-Time Polling Service (ertPS); Best-Effort (BE) service. Variable bandwidth assignment is possible in rtPS, nrtPS, ertPS and BE services. Whereas UGS service needs fixed and dedicated bandwidth assignment. Figure (2) shows the QoS architecture of IEEE 802.16 based services.

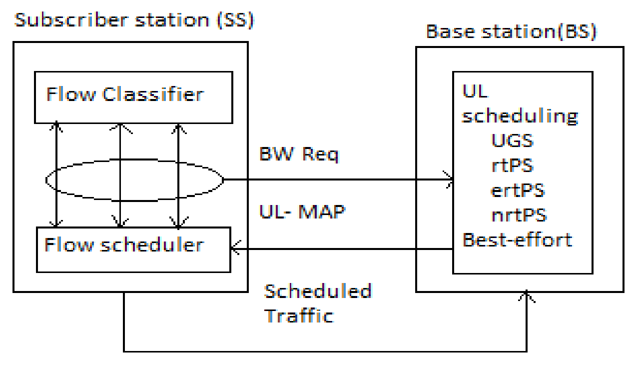

Fig.2 shows the QoS architecture of IEEE 802.16

3. UPLINK SCHEDULING ALGORITHMS

3.1 UGS Algorithm

The UGS algorithm is designed to support real-time service flows that generate fixed-size data packets periodically [8]. In this algorithm, the base station (BS) periodically assigns fixed size grants to the voice user, and these fixed-size grants are sufficient to send voice data packets generated by the maximum data rate of EVRC. Thus, this algorithm can minimize MAC overhead and uplink access delay caused by the bandwidth request process of the user to send voice packets. However, this algorithm has only a small capacity for VoIP services. Generally, voice users do not always have voice packets with the same size, because they have variable data rates and silence suppression [9], [10]. In this algorithm, since the BS always assigns fixed-size grants that are sufficient to send voice packets generated by the maximum data rate of EVRC, it causes a waste of many uplink resources, as shown in Fig. 3.

In this figure, a dot line and a solid line show the amount of assigned uplink resources by the BS and the amount of used uplink resources by the user, respectively. The blank regions represent the waste of uplink resources. In other words, in the UGS algorithm, since the BS always allocates the same amount of uplink resources to each user regardless of his voice status, it causes the waste of many uplink resources.

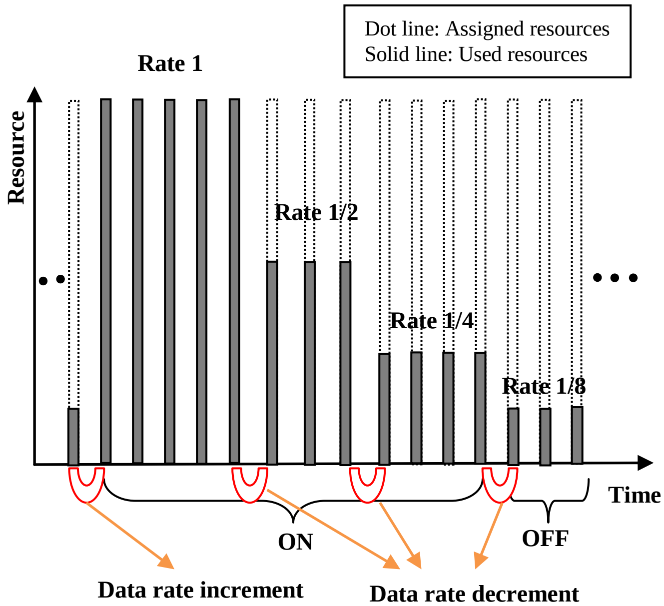

Fig.3 shows the UGS Algorithm

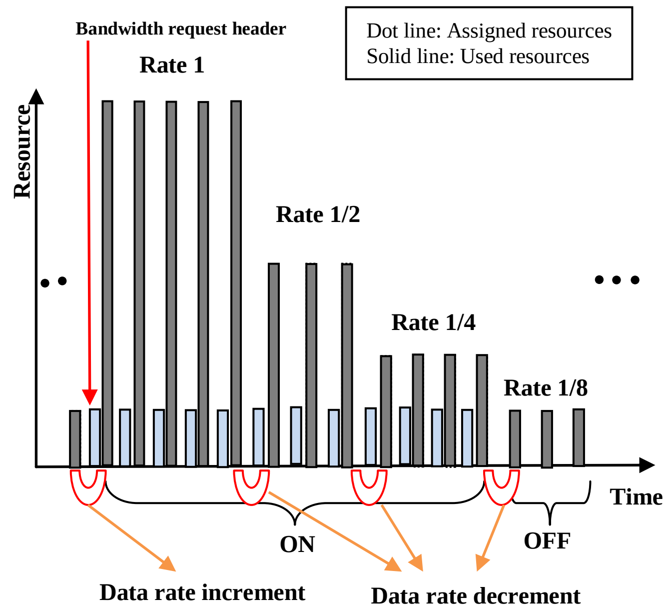

Fig.4 shows the rtps Algorithm

3.2 RTPS Algorithm

The rtPS algorithm is designed to support real-time service flows that generate variable size data packets periodically [8]. The BS assigns uplink resources that are sufficient for unicast bandwidth requests to the voice user. This period is negotiated in the initialization process of the voice sessions. Generally, this process is called a bandwidth request process, or polling process. Because this algorithm always uses a bandwidth request process for suitable size grants, it transports data more efficiently than the UGS algorithm. However, this bandwidth request process always causes MAC overhead and additional access delay. Hence, the rtPS algorithm has larger MAC overhead and access delay than the UGS and ertPS algorithms. In Fig. 4, the dot line and the solid line show the amount of assigned uplink resources by the BS and the amount of used uplink resources by the voice user, respectively. In this algorithm, since the user requests exact amount of uplink resources for transmitting his voice packet, the dot line and the solid line are nearly the same. To avoid the polling process in case of silence duration, we assume that minimum polling size is the size of voice packet generated by the minimum data rate of the voice codec. Thus, there is no polling process in the silence duration of the users. In the rtPS algorithm, the user can use the piggyback requests of the grant management sub-header for VoIP services. But, since VoIP services are delay-sensitive and the usage of piggyback requests is not proper method for supporting service-flows that have periodic property, the use of piggyback requests is not a desirable method for VoIP services. By precise negotiation of the polling period in the initialization process, the use of piggyback requests may be avoided for VoIP services.

3.3 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 silence suppression [11]. This algorithm is recently proposed and accepted in IEEE 802.16e standard. In order not only to reduce MAC overhead and access delay of the rtPS algorithm, but also to prevent the waste of uplink resources of the UGS algorithm, the ertPS algorithm assigns uplink resources according to the status of the voice users without MAC overhead. Here, we describe the detailed operation of the ertPS algorithm as follows. Firstly, the voice user informs the BS of his voice status information using Grant Management sub-header in case that the size of a voice data packet is decreased [11]. The user requests the bandwidth for sending the voice packets using extended PBR (PiggyBack Request) bits of Grant Management sub-header. In the ertPS algorithm, to distinguish these extended PBR bits with general PBR bits, the user sets the MSB of PBR bits to 1. In this case, the BS assigns uplink resources according to the requested size periodically, until the voice user requests another size of the bandwidth. Secondly, the voice user informs the BS of his voice status information using Bandwidth request header in case that the size of a voice data packet is increased [11]. The user requests the bandwidth for sending the voice packets using BR (Bandwidth Request) bits of Bandwidth request header. In the same way as the case of the data rate decrement, to distinguish these BR bits with general BR bits, the user sets the MSB of BR bits to 1. In this case, the BS assigns uplink resources according to the requested size periodically, until the user 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, as shown in Fig. 5.

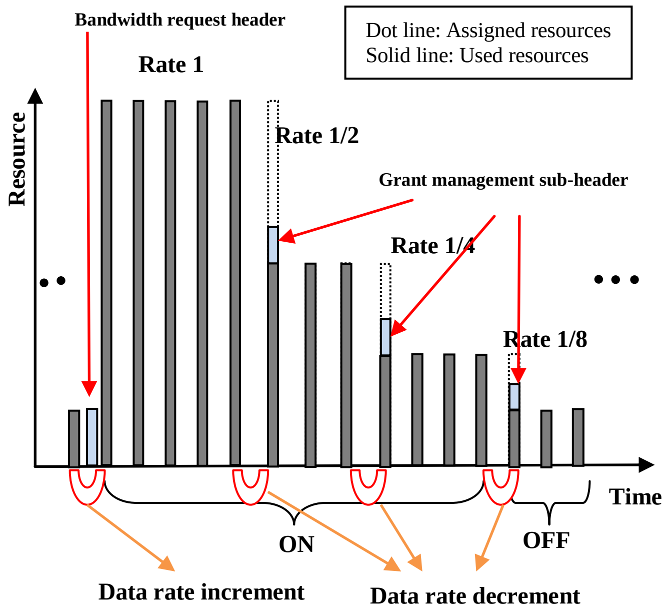

Fig.5 shows the ertps Algorithm

In summary, in case of the VoIP services using this algorithm, the BS recognizes Grant Management sub-header and Bandwidth request header especially. In this algorithm, if the user requests the bandwidth for sending the voice packets, then the BS shall change its polling size according to the bandwidth size requested by the user, and keeps its changed polling size until the user sends another request. In other words, the BS may not change its polling size without any requests from the voice users. Using this ertPS algorithm, the BS can obtain better data transport efficiency compared with the UGS and rtPS algorithms.

4. Performance Metrics

In our simulations, we use the following metrics toevaluate the performance of WiMAX network in terms of end-to-end QoS for VoIP.

4.1 Mean Opinion Score (MOS)

MOS provides a numerical measure of the quality of human speech in voicetelecommunications, with value ranging from 1 to 5 where 1 is the worst quality and 5 is the best quality. Inour simulation, we compute MOS through a non-linearmapping from R-factor as in [12]:

$$MOS = 1 + 0.035 \times R + 7 \times 10^{-6}\left[R(R-60)(100-R)\right] \quad (1)$$

where $R = 100 - I_s - I_e - I_d + A$; Is: the effect of impairments that occur with the voice signal; Ie: the impairments caused by different typesof lossesoccurred due to codec's and network, and Id: represents the impairment caused by delay particularly mouth-to-eardelay. Using the default setting for Is and A Eq. (1) can be reduced to: $R = 94.2 - I_e - I_d$

4.2 Packet end-to-end delay (D_{e2e})

The total voice packet delay; De2e calculated using the formula:

$$D_{e2e} = D_n + D_e + D_d + D_c + D_{de} \quad (2)$$

where D_n, D_e, D_d, D_c and D_{de} represent the network, encoding, decoding, compression and decompression delay, respectively[13].

4.3 Jitter

Jitter defined as the signed maximum difference in one-way delay of the packets overa particular time interval. Let t(i) and t'(i) be the timetransmitted at the transmitter and the time received at thereceiver, respectively. Jitter is then, calculated as follows:

$$Jitter = \max_{1 \le i \le n}\left\{\left[t'(n) - t'(n-1)\right] - \left[t(n) - t(n-1)\right]\right\} \quad (3)$$

4.4 Packet delay variation (PDV)

PDV defined as the variance of the packet delay, which can be, calculated from the following Eq.

$$PDV = \frac{\sum_{i=1}^{n}\left(\left[t'(n) - t(n)\right] - \mu\right)^2}{n} \quad (4)$$

where u is the average delay of the n selected packets.

5. Results and Discussion

This implementation was performed using OPNET 14.5 on an Intel Core I 7; 1.7 GHz/2MB Cache processor sing Windows-7 64-bit operating system.

In our OPNET modeler implementation of VOIP over WiMAX, there are some parameters in which specified as Voice parameters, which listed in Table.1 for all scenarios and some specified as IEEE802.16 parameters listed in Table.1. The simulation duration 500 sec

5.1 MOS

Fig. 6 plots the average MOS for different scenarios Variables, which lists in tables (1) and (2). A major observation is that the average MOS almost the same for voice silence and voice without silence for all algorithms of QoS

5.2 Packet end-to-end delay

Packet end-to-end delay is one of the most important performances
Metric in VoIP, Figure 7 show the average packet end-to-end delay, we can notice that the UGS has the lost delay because it has not much header over like RTPS algorithm.

5.3 Jitter

According to equation (3), the jitter value can be negative which means that the time difference between the packets at the destination is less than that at the source. Figures8plot the jitter

5.4 Packet delay variation (PDV)

Packet delay variation plays a crucial role in the networkperformance degradation and affects the user-perceptualquality. Higher packet delay variation results in congestionof the packets, which can results in the network overhead. Figures9 plot packet delay variation.

5.5 WiMAX uplink packets dropped

Figure 10 plots the WiMAX uplink packets dropped, we can notice the ERTPS algorithm has less packet dropped than UGS and RTPS algorithms

5.6 WiMAX delay and throughput

Figures 11 and 12 respectively plot WiMAX delay and throughput

Table.2 The voice parameter's

ATTRIBUTE	*VALUE*
Voice	PCM Quality Speech
Codec type	PCM G.711with silence suspicion
Frame size	10 ms
Coding rate	64kbit/s
Voice frames per packet	1 frame
Compression and decompression delay	0.02 sec

6. CONCLUSIONS AND FUTURE WORK

Next generation networks with multiple technologies offerdifferent multimedia services to the user. It also providesthe luxury of utilizing the best available technology for therequired service to a user, companies and business organizations.In this study, we have conducted extensive simulationstudy to evaluate the performance of WiMAX forsupporting VoIP traffic. We have analyzed several importantcritical parameters such as MOS, end-to-end delay, jitter , packet delay variation, WiMAX uplink packet dropped and WiMAX delay and throughput. Simulation results show that the average MOS for all algorithms almost the same. We can notice the ERTPS algorithm has less packet dropped and low delay .Future work includes advanced uplink scheduling algorithm for decrease the delay and decrease the wastage of uplink sub-frame resources.

Table.2 The IEEE802.16 parameters

ATRIBUTE	*VALUE*
Scheduling type	UGS,RTPS,ERTPS
Frame duration	12 ms
Duplexing Technique	TDD
Base Frequency	5.8 GHz
Bandwidth	20 MHz
Transmission power	0.5 W
Modulation and coding	QPSK ½
Average SDU size	120 bytes

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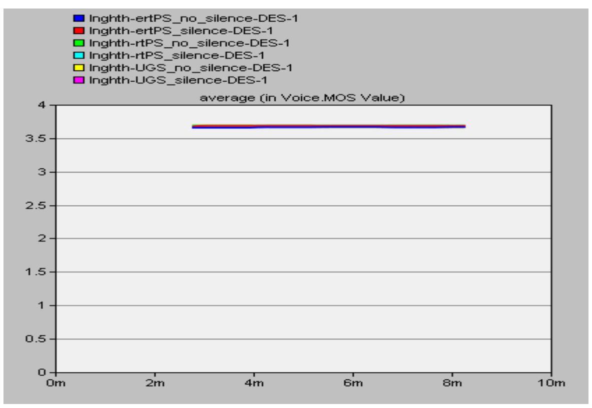

Fig.6 shows the MOS

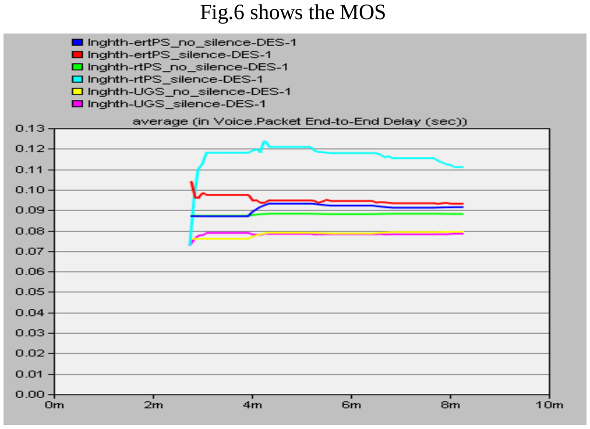

Fig.7 voice packet End-to-End delay

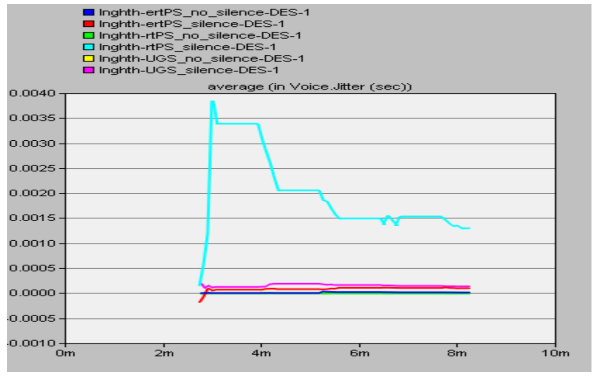

Fig.8 shows voice jitter

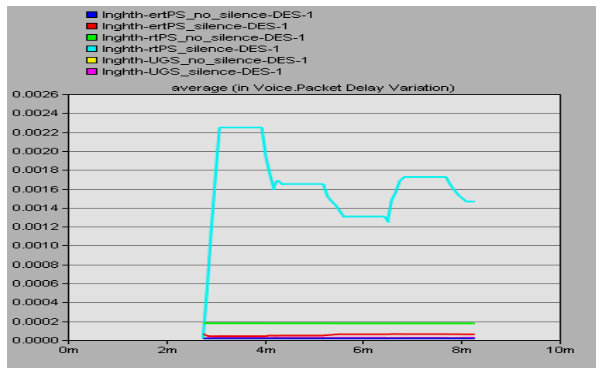

Fig.9 shows voice's packet delay variation

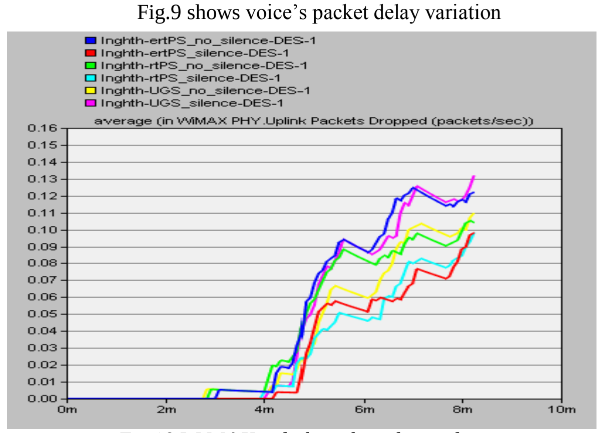

Fig.10 WiMAX uplink packets dropped

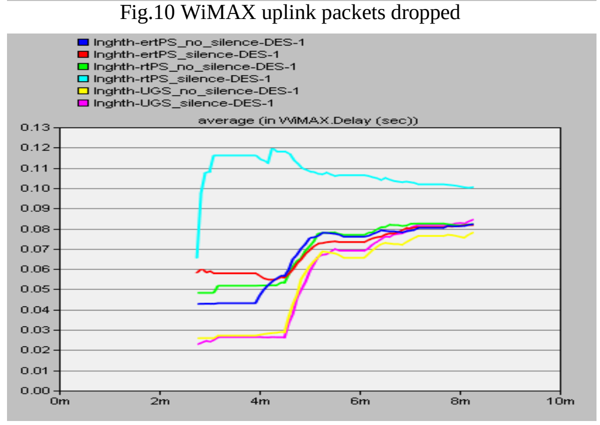

Fig.11 WiMAX delay

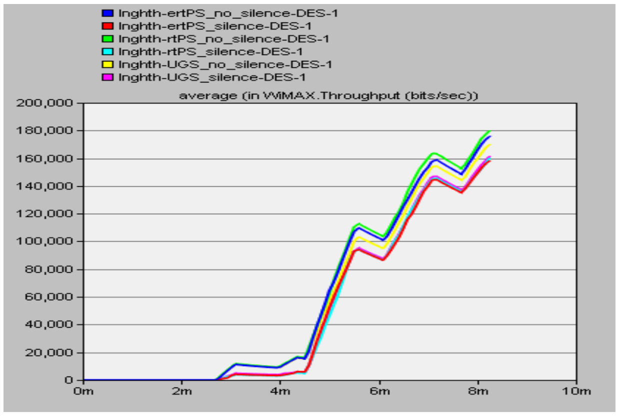

Fig.12 WiMAX throughput

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