Journal of Engineering Sciences, Assiut University, Vol. 39, No 6, pp.1403-1415, November 2011

OPTIMIZING WIMAX MAC LAYER PARAMETERS

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(Received June 21, 2011 Accepted September 4, 2011)

In this paper some WiMAX MAC layer parameters will be optimized to improve applications performance. The study is aimed to show, via simulation, the effect of Automatic Repeat Request (ARQ) parameters on the WiMAX performance. Also, it is aimed to verify the effectiveness of Unsolicited Grant Service (UGS), Real Time Polling Service (rtPS), and Enhanced Real Time Polling Service (ertPS) in managing voice traffic and their effect on data traffic which use Best Effort (BE) service. The results show that the video delay will be increased by from 5 ms to 24 ms when enabling ARQ protocol on WiMAX network. It also give the optimum parameters for ARQ protocol to improve the WiMAX network performance. The research also shows that the best scheduling service for voice is UGS; but it reduces the throughput for BE data.

I- INTRODUCTION

The widespread use of IP-based technologies resulted in the vision of converged networks that promises cost-efficiency by supporting voice, video, and data on a single network. This gave rise to the popularity of Voice over IP (VoIP), and Video over IP which provide efficient voice and video delivery over packet-switched networks by better resource utilization as compared to traditional circuit-switched mode.

WiMAX (Worldwide Interoperability for Microwave Access) is a telecommunication technology which is based on The IEEE 802.16 standard. Foundation of first WiMAX model was lay down in June, 2001 in the form of IEEE 802.16a. WiMAX provides fixed broadband wireless access services up to IEEE 802.16d (IEEE 802.16-2004). IEEE 802.16e and the entire standard after it support the mobility feature. Latest version of WiMAX is IEEE 802.16j which has developed in 2009 [1].

WiMAX physical layer implements Orthogonal Frequency Division Multiplexing (OFDM) which called fixed WiMAX, and Orthogonal Frequency Division Multiple Access (OFDMA) which is called mobile WiMAX. WiMAX MAC (Medium Access Control) layer consists of three sublayer as shown in Fig. 1, service-specific Convergence Sublayer (CS), Common Part Sublayer (CPS), and privacy sublayer; each of them has specific functions.

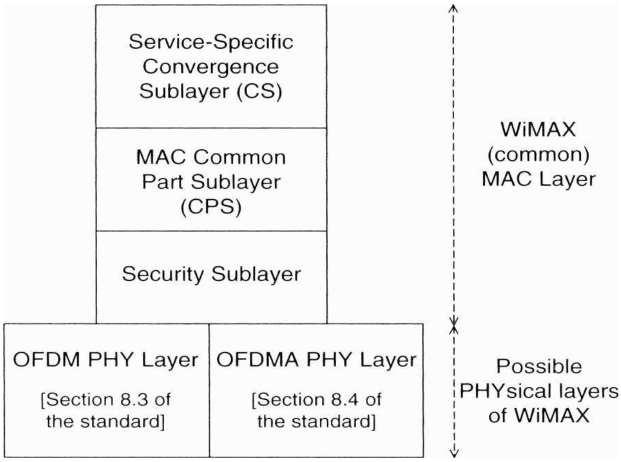

Figure 1 – WiMAX layers.

The related works is discussed below; Sayenko provides a comparison of ARQ and HARQ performance in IEEE 802.16 networks [2]. This paper also compares the overhead size generated by ARQ and HARQ. Evaluation of the type of packet acknowledgment for different channel condition is presented in [3]. The optimal PDU size and MAC overhead due to the packets retransmission is analyzed by Hoymann in [4]. Sengupta propose to adjust the MAC PDU size depending on the channel state to achieve the best ARQ performance [5]. In [6], the effect of ARQ parameters on the TCP performance in WiMAX networks is examined. The paper shows that by optimizing ARQ parameters, the overall TCP throughput is enhanced.

In [7] a simple analytical method calculates the maximum number of VoIP users in a mobile WiMAX system for different voice codecs, but the author assumes that the scheduling service is UGS only. Authors in [8] have done a work on VoIP transmission on ertPS. In paper [9] authors have evaluated the performance of different VoIP codecs for both WiMAX and WLAN scenario and for an integrated environment

An investigation of the performance of WiMAX/WiFi networks for VoIP application presented in [10] using different codecs, but the author didn’t mention the scheduling service used for voice. The performance evaluation of VoIP for a mobile user and how the QoS parameters vary for different speeds are studied in [11].

Most of the above mentioned papers didn’t focus on the effect of optimization of ARQ parameters on the video application’s end-to-end performance. Also, the effect of changing the scheduling service for VoIPe on the Best Effort data didn’t presented.

This paper discusses the optimization of ARQ parameters to enhance the video performance on WiMAX networks. Also it studies the effect of using different scheduling services for voice and its effect on Best Effort data.

The rest of the paper is organized as follows. Section II, presents an overview about the WiMAX physical layer. Introduction about the WiMAX MAC layer operations including ARQ and scheduling services will be presented in section III. Section VI, presents simulations and results. The paper will be concluded in section V followed by the relevant references.

II - PHYSICAL LAYER

Physical layer in WiMAX performs modulation, signal mapping, MIMO Processing and Forward error correction coding. WiMAX contains BPSK, QPSK, 16-QAM, and 64-QAM modulation techniques in downlink (DL) and uplink (UL) [1].

Each WiMAX frame consists of DL and UL subframes. A preamble is used for time synchronization. The downlink map (DL-MAP) and uplink map (UL-MAP) define the burst-start time and burst-end time, modulation types and forward error control (FEC) for each SS. The MAP's lengths and usable subcarriers are defined by the Frame Control Header (FCH). The SS allocation is in terms of bursts. Since the channel state condition keeps changing over time because of the nature of wireless media, WiMAX supports adaptive modulation and coding [12].

III - MAC LAYER

The convergence sublayer (CS) responsible for classification of the traffic and optionally suppress the higher layer header. The Common Part Sublayer (CPS) provides a connection identifier (CID) to identify which connection the MAC Protocol Data Unit (MPDU) is servicing. The QoS is taken into account for the transmission and scheduling of data over the PHY Layer. The CPS includes many procedures of different types: frame construction, multiple access, bandwidth demands and allocation, scheduling, radio resource management, QoS management, etc. [13][14].

III-I - Automatic Repeat Request (ARQ)

ARQ is an optional MAC feature in [802.16-2004] standard. However, the implementation of ARQ is mandatory for Mobile WiMAX. When implemented, ARQ may be enabled on a per-connection basis. The per-connection ARQ shall be specified and negotiated during connection creation. A connection cannot have a mixture of ARQ and non-ARQ traffic. MAC layer ARQ alone does not improve the spectral efficiency. However, with its retransmissions mechanism to correct packet errors at the cost of extra delays, ARQ provides a more reliable link layer as seen by applications.

III-I-I - ARQ Operations

1. Fragmentation. When ARQ is enabled, Service Data Unit (SDU) is treated as fragmented into logical ARQ blocks with fixed block size defined by ARQ_BLOCK_SIZE as in fig. 2. Fragmentation shall occur only on ARQ block boundaries [15].

The number of blocks is given by the following equation:

$$N_{blocks} = S_{data} / S_{ARQ\_block} \qquad (1)$$

Where S_{data} is a total size of data in one frame in bytes, parameter S_{ARQ_block} represents a block size defined by ARQ_BLOCK_SIZE in bytes [16].

When the length of the SDU is not an integer multiple of ARQ_BLOCK_SIZE, the final block of the SDU is formed using the SDU bytes remaining after the final full block has been determined. Fragmentation sub-header (FSH) is attached to each fragmentation boundary block and contains a Block

Sequence Number (BSN) of the first ARQ block of each SDU. When a PDU is packed, packing sub-header (PSH) is attached instead of fragmentation sub-header.

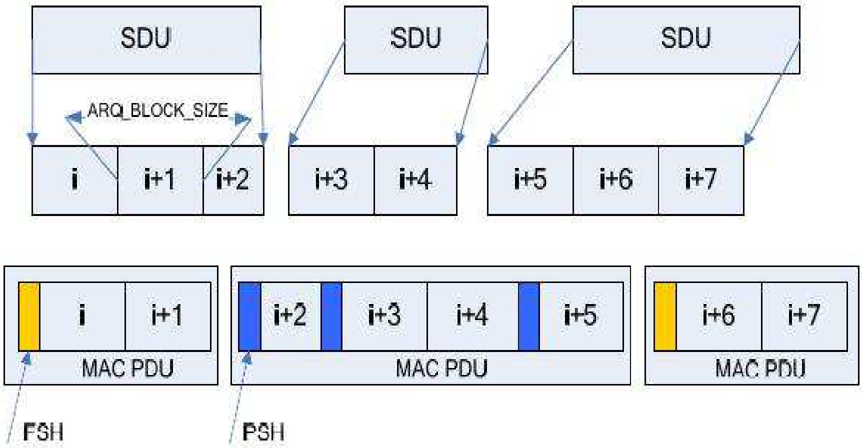

Figure 2 - ARQ operation

2. Buffer management. ARQ transmitter and receiver each manage blocks in a sliding window. Due to the sliding window mechanism, the minimum buffer size shall be ARQ_WINDOW_SIZE * ARQ_BLOCK_SIZE.
3. ARQ retransmissions. Retransmission will be triggered by an ARQ NACK or a transmitter side retransmission timer defined by ARQ_RETRY_TIMEOUT. ARQ_BLOCK_LIFETIME controls the estimated maximal retransmission count. Therefore, conservatively setting ARQ_BLOCK_LIFETIME to a large value is generally safe as long as the buffer allows and this will be unlikely to cause extremely long delays as consecutive packet errors happen with geometrically decreasing probability [15].
4. ARQ feedback. ARQ feedback is per-block ACK/NAK response sent from ARQ receiver to ARQ transmitter. ARQ feedback should be treated as a regular MAC payload, i.e., resource allocation will go through scheduling and uplink bandwidth request if necessary. ARQ feedback information can be sent as a standalone MAC management message on basic management connection, or piggybacked on an existing transport connection. ARQ feedback cannot be fragmented. The feedback message contains 1 byte (8 bits) field Message ID and field ARQ_Feedback_Payload. The ARQ_Feedback_Payload consists of one or more Information Elements (IE) carried by ARQ_Feedback_IE. Every ARQ_Feedback_IE is related to just one CID (Connection ID).

The size of IE of each ARQ feedback message (in bits) can be calculated according to the next equation:

$$Size_{ARQ\_FB\_IE} = 32 + (M*16) \qquad (2)$$

Where M represents a number of maps carried in one ARQ_Feedback_IE (Maximum 4 maps). The overall size (in bits) of whole feedback message is given in the following formula:

$$\text{Size}_{\text{ARQ\_FB}} = 8 + \sum_{i} \text{Size}_{\text{ARQ\_FB\_IE(i)}} \quad (3)$$

Where i is a number of information elements carried in one ARQ Feedback message and the number 8 (bits) represents the ARQ feedback message overhead (Message ID field). The overhead transmitted in all considered frames (N) is equal to the sum of partial overheads over the N:

$$\text{OH}_{\text{ARQ}} = \sum_{N} \text{Size}_{\text{ARQ\_FB(N)}} \quad (4)$$

The OH_{ARQ} is presented in bits.

There are two main ARQ feedback types Selective ACK, Cumulative ACK.

In case of cumulative acknowledgements, only one block sequence number is sent indicating the "last in-sequence" ARQ block successfully received [17]. In case of selective ARQ the receiver keeps track of the sequence number of received PDUs and send back the ACK/NACK to the BS at the following UL sub-frame to report the information about whether or not the PDUs transmitted at the current DL sub-frame are successfully received or not. Based on the feedback information, only failed PDUs are retransmitted next time when this queue obtains the transmission opportunity. Since less retransmissions are involved, the selective ARQ is more efficient than the cumulative ARQ. On the other hand, due to out-of-order sequence numbers, the implementation of the selective ARQ need more buffer and complexity compared with the cumulative ARQ. Also, selective ACK feedback messages are bigger in size. For large ARQ block size, selective ACK should be preferred over the cumulative-only ACKs [18].

III-II - Scheduling service

WiMAX support multiple applications (data, voice, and video) with different QoS requirements. The MAC layer protocol defines four QoS services:

- Unsolicited Grant Service (UGS): It is designed for services which require Constant Bit Rate (CBR) such as voice application and T1/E1.
- Real-Time Polling Service (RTPS): It is designed for services which generate variable size data packets but delay requirements should be met e.g. MPEG video
- Non-Real-Time Polling Service (NRTPS): It is designed for services which require good average data rate performance but can tolerate delay e.g. FTP.
- Best Effort (BE) service: It is designed for services which don’t require any specific QoS guarantee e.g. HTTP and Web Browsing [19].

In mobile WiMAX standard (802.16e-2005) there is new service called Extended Real Time Polling Service (ERTPS). It is designed for variable rate real time applications that have data rate and delay requirements, like VoIP without silence suppression [20]. The method of requesting bandwidth differs from one method to another. In a UGS service, the BS provides fixed-size data grants at periodic intervals. This eliminates the overhead and latency of SS requests. In rtPS, the BS provides periodic unicast (uplink) 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 overheads than UGS, but supports variable grant sizes for optimum real-time data transport efficiency. The standard states that the BS typically polls nrtPS CIDs on an interval on the order of one second or less. In addition, the SS is allowed to use contention request opportunities. In BE the SS may use contention request opportunities. In ertPS the BS provides unicast grants in an unsolicited manner like in UGS, thus saving the latency of a bandwidth request; however, whereas UGS allocations are fixed in size, ertPS allocations are dynamic.

VI - SIMULATION AND RESULTS

OPNET Modeler [21] is used to construct a WiMAX network. We measure the performance in a typical sector, typically 3 sectors per cell, with radius 1Km where SSs are uniformly located in the sector. We have two scenarios, the first one simulates WiMAX network consists of two SSs, one of them use ARQ feature and the other one doesn’t use it. The two SSs send video traffic with 1400 byte frame size and 10 ms interarrival time. In this scenario we assume that the loss percentage is 50 %. The second scenario simulates WiMAX network which consists of two SSs, one of them sends BE data and the other sends voice frames. The voice traffic uses G711 codec with UDP transport protocol. The voice rate is 96 Kbps. The WiMAX physical layer uses OFDMA with 20 MHz bandwidth, 5 ms frame size, and downlink/uplink ratio equal to 50 %.

VI-I Optimizing ARQ Parameters

Figure 3 shows the effect of enabling ARQ on the delay of the service flow, the figure shows that to mitigate 50% loss rate the average delay jump from 5 ms (which is the WiMAX frame duration) to about 24 ms; This means that the delay increased by 380% when enabling ARQ protocol. This is because one retransmission attempt makes delay equal to 3 times frame duration. Also when enabling ARQ the average jitter value becomes 25 ms as shown in fig. 4 (note that the maximum allowed jitter value for voice is 30 ms). Thus enabling ARQ mitigate 50% loss rate with accepted delay and jitter.

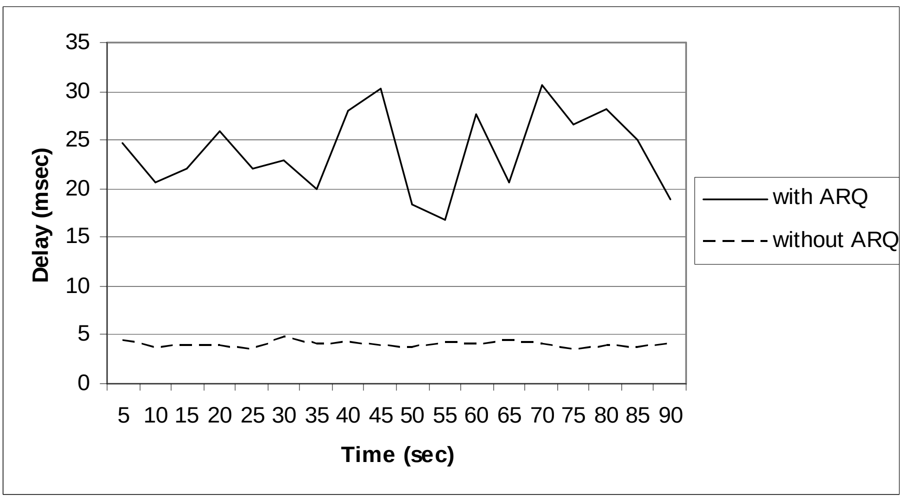

Figure 3- Video delay with and without ARQ

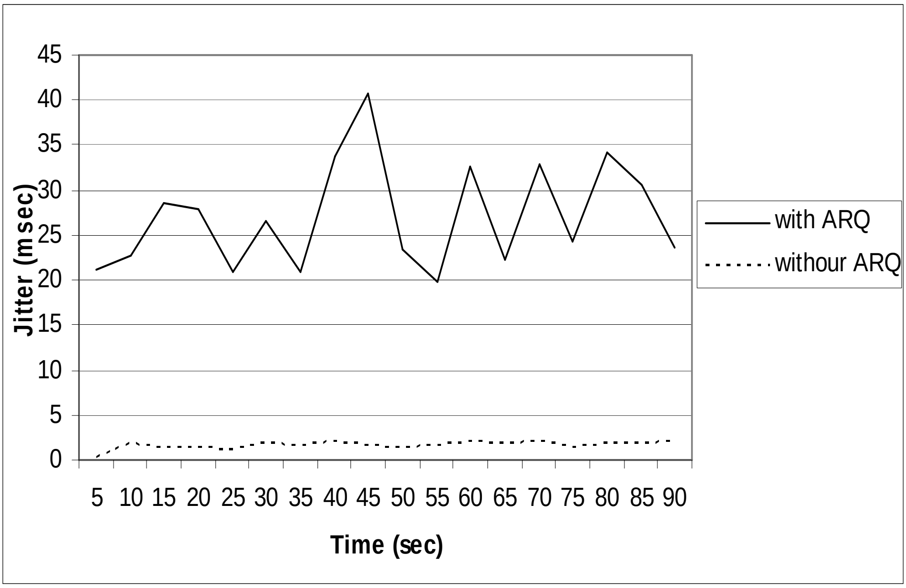

Figure 4- Video jitter with and without ARQ

Figure 5 shows the effect of loss rate probability on jitter when the ARQ is enabled. The jitter increases with the loss rate as shown in the figure. This is because with increasing loss rate the retransmission rate increases also which means increasing jitter and delay. The figure shows that the critical point in loss rate is 53 % at which the jitter starts to increase over 30 ms which is not accepted value for many real time applications.

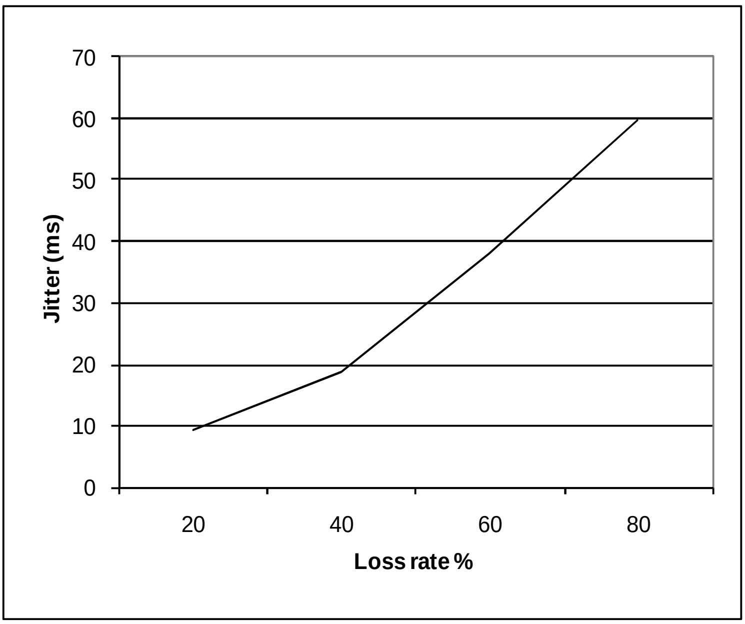

Figure 5- Relation between loss rate and jitter.

Figure 6 and 7 show the effect of ARQ_BLOCK_LIFETIME / ARQ_RETRY_TIMEOUT on delay and normalized throughput respectively. ARQ_RETRY_TIMEOUT will be 30 ms and ARQ_BLOCK_LIFETIME / ARQ_RETRY_TIMEOUT will be varied from 2 to 10. The figures show that to achieve throughput equal to 95% with delay equal to 22 ms the ARQ_BLOCK_LIFETIME / ARQ_RETRY_TIMEOUT must be set to 4. This result matched with the same ratio but using TCP packets [6].

Figure 6 – Delay vs. (life time / retry time)

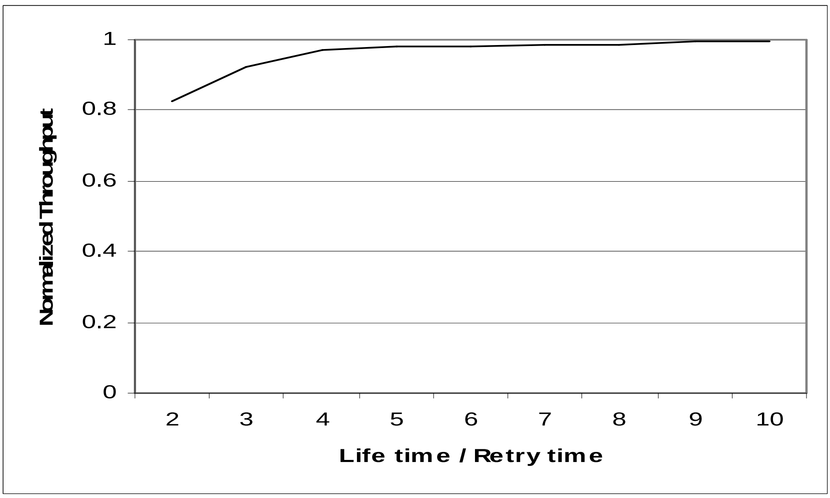

Figure 7 – Throughput vs. (life time / retry time)

The effect of different ACK types on the delay is shown in Fig. 8. The figure shows that Cumulative ACK offers average packet end to end delay equal to 22 ms. The selective ACK offers average delay equal to 15 ms (64 % delay reduction). This is because in case of cumulative acknowledgements, only one block sequence number is sent indicating the "last in-sequence" ARQ block successfully received. Thus if there are a missing block, the sender will not retransmit it until the whole blocks inside the window have been transmitted, which increase the average delay.

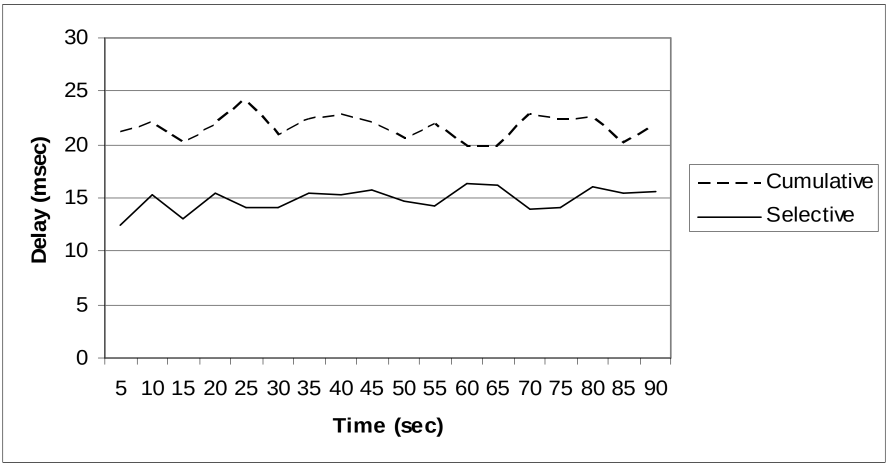

Figure 8 – Effect of ARQ ACK types on delay.

VI-II Scheduling services

Figure 9 shows the effect of the three different scheduling services on the voice delay. The figure shows that the best scheduling service for voice is UGS which achieve minimum delay (65 ms). This is because in UGS service the slots reserved for every voice session during the negotiation phase, which eliminate the need to request bandwidth, thus reduce the delay. The minimum delay value (65 ms) coincides with the results in [7].

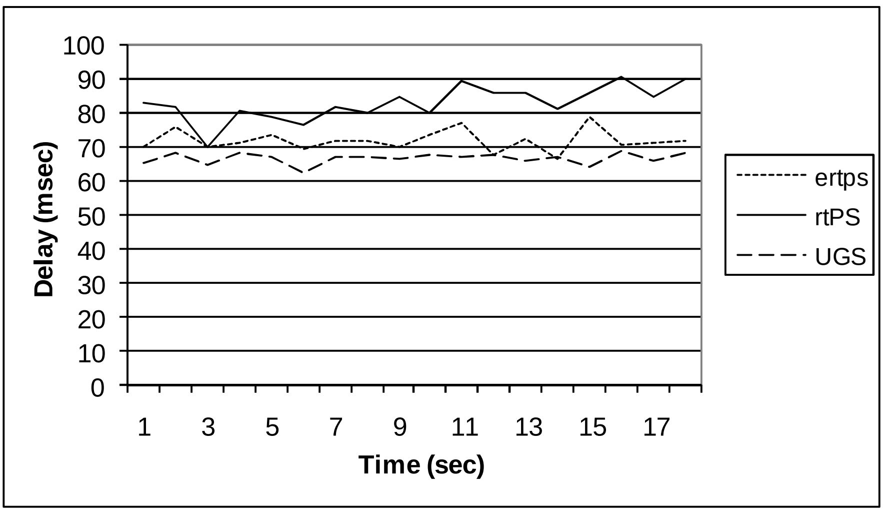

Figure 9 - Voice delay vs. scheduling service

Figure 10 shows Best Effort (BE) delay against scheduling services. The figure shows that the BE delay increases when the voice uses UGS scheduling service, this is because in UGS service the slots reserved for the voice session even if there isn't voice traffic to send. The figure also shows that to minimize the BE delay, voice should use rtPS or ertPS.

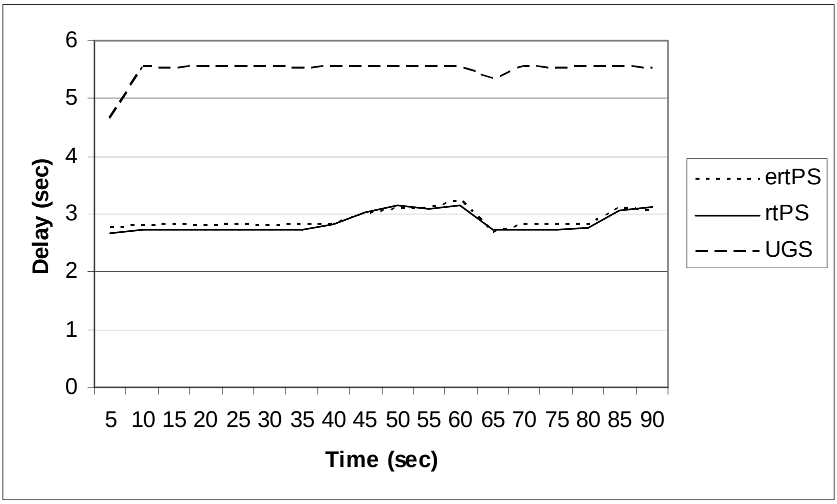

Figure 10 – Best Effort delay vs. scheduling services

Table 1 shows the average voice delay and jitter with 1, 5, and 10 voice SS numbers. The results show that when the voice traffic uses rtPS scheduling service, the delay and jitter increase rapidly to unaccepted values with increasing the number of voice SSs. This is because in rtPS service, when increasing the number of SSs, the SSs use group polling to request B.W. This introduces delay when using rtPS scheduling service, which is not accepted in VoIP service. The table also shows that, the best scheduling service for voice which achieves minimum delay and jitter is UGS. But this increases the BE delay because UGS reserve the slots for the voice frames even if there isn't voice traffic to send. The table shows that ertPS is the best choice for voice in case of mixed traffic (BE data + voice).

Table 1- Voice delay vs. voice SS numbers with different scheduling services.

SS#	Delay (sec.)			Jitter (sec.)		
	ertPS	rtPS	UGS	ertPS	rtPS	UGS
1	0.072078	0.084967	0.066875	0	0	0
5	0.083484	0.670739	0.075292	0.00006	0.272841	0.000128
10	0.094298	1.266746	0.075361	0.000171	0.636219	0.00013

Figure 11 shows the BE throughput against scheduling service. During voice idle periods the BE connection cannot increase its transmission rate (assuming 10 voice SSs used), this is because UGS doesn't dellocate its resources during idle periods and this reduce the BE throughput to around 11 %.

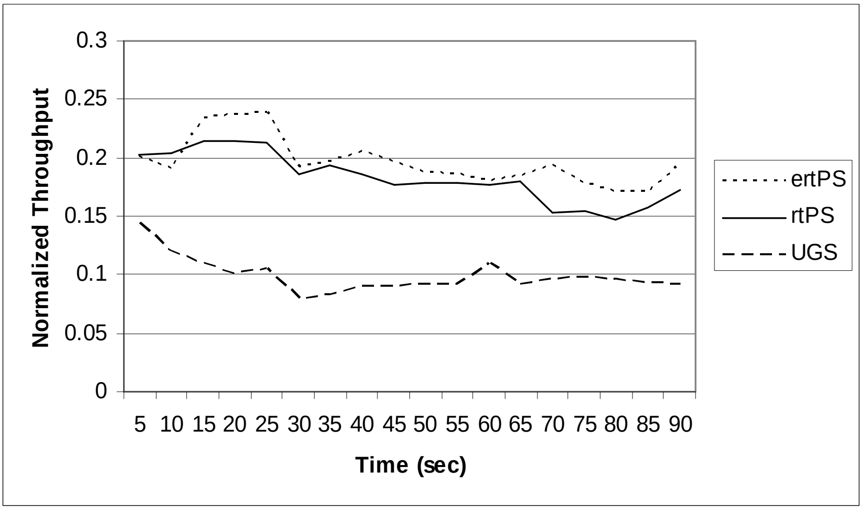

Figure 11 – Throughput vs. scheduling services

By using ertPS for voice traffic the BE throughput increases to around 17 %, this is because ertPS deallocates its resources (to a minimum) during idle periods. The rtPS shouldn’t be used for voice because it increases the voice delay as shown in table 1. But generally the BE throughput is very small because there is no QoS guarantee for this scheduling service and the only method for bandwidth request is contention.

Table 2 shows the BE average delay against scheduling services with variable number of voice SS. The results show that by using UGS the BE data starve, and the scheduling service which produce minimum BE delay is rtPS. This is because no reservation for any slots for voice sessions. But generally the BE delay is very large with 5 and 10 voice SS.

Table 2 – BE delay vs. SSs with different scheduling services.

SS	ertPS	rtPS	UGS
1	2.975009	2.923384	5.494709
5	8.195031	4.005171	Inf
10	9.239348	7.040595	Inf

V- CONCLUSION

The paper showed that by optimizing ARQ protocol parameters and scheduling services, the overall applications performance in WiMAX networks in terms of delay and throughput can be enhanced. The study showed the effect of ARQ protocol on the average delay and jitter of video. Also, it suggested the ratio of “life time / retry time” to be 4 in order to achieve 95 % throughput. The paper showed the best ARQ ACK type from the delay perspective is Cumulative and Selective Bitmap ACK.

The scheduling service helps in providing QoS to different application in WiMAX networks. The results showed that best scheduling service for voice in WiMAX network is UGS, but it increases the BE traffic delay because it reserves bandwidth for voice even there is not voice traffic to send. Using ertPS for voice minimize the voice delay comparing to rtPS and improve the BE traffic delay and throughput.

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تحسين معاملات مرحلة التحكم فى الولوج على الوسط فى شبكات الواى- ماكس

فى هذا البحث يتم تحديد قيم بروتوكول التحكم فى الولوج لشبكات الـ WiMAX وذلك لتحسين أداء التطبيقات. ويهدف البحث الى عرض تأثير بروتوكول إعادة الارسال الاوتوماتيكى على أداء شبكة الـ WiMAX وايضا الى التحقق من فاعلية خدمات الـ UGS و rtPS و ertPS على نقل الصوت فى شبكات الـ WiMAX وتأثيرهم على نقل البيانات من خلال خدمة الـ Best Effort . ويبين البحث ان باستخدام بروتوكول إعادة الارسال الاوتوماتيكى يزيد التأخير فى نقل الفيديو من 5ms الى 24ms . ويعطى البحث القيم المثلى لبروتوكول إعادة الارسال الاوتوماتيكى. ويبين البحث ان افضل نظام جدولة لنقل الصوت فى شبكات الـ WiMAX هو UGS ولكنة يقلل معدل نقل البيانات لخدمة الـ BE.