الجامعة الأردنيـــــة
كلية الدراسات العليا

التاريـــخ: / /

نموذج رقم (18)
اقرار والتزام بالمعايير الأخلاقية والأمانة العلمية وقوانين الجامعة الأردنية وأنظمتها وتعليماتها لطلبة الماجستير

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عنوان الرسالة

QoS-Aware Scheduling Scheme for IEEE 802.16 traffic

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QOS-AWARE SCHEDULING SCHEME FOR IEEE 802.16 TRAFFIC

By

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COMMITTEE DECISION

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LIST OF ABBREVIATIONS

AAS	Adaptive Antenna System
ADPQ	Adaptive Deficit Priority Queue
ATM	Asynchronous Transfer Mode
Bandwidthinbits	Available Bandwidth
Bavail	Available Bandwidth
BE	Best Effort
BEQ	BE Quantum
Breq	Bandwidth Requested
BS	Base Station
BWA	Broadband Wireless Access
CBR	Constant Bit Rate
CDMA	Code Division Multiple Access
CID	Connection Identifier
CPE	Customer Premises Equipment
CS	Convergence Sublayer
currentTime	Current Time
DC	Deficit Counter
DCD	Downlink Channel Descriptor
DCdynamic	The Dynamic DC
DCmax	The Maximum Deficit Counter
DCmin	The Minimum Deficit Counter
DFPQ	Deficit Fair Priority Queue
DL	DownLink

DL-MAP	DownLink MAP
DSA	Dynamic Service Addition
DSC	Dynamic Service Change
EADPQ	Extended Adaptive Deficit Priority Queue
EDF	Earliest Deadline First
ertPS	extended real-time Polling Service
FDD	Frequency Division Duplex
FDM	Frequency Division Multiplexing
FFT	Fast Fourier transform
FTP	File Transfer Protocol
GWR	Grant Without Request
H-FDD	Half duplex FDD
HTTP	Hypertext Transfer Protocol
IE	Information Elements
lastAllocTime	Last Allocation Time
lastBwRequested	Last Bandwidth Requested
LMDS	Local Multipoint Distribution Service
LOS	Line Of Sight
MAC	Medium Access Control
MCS	Modulation and Coding Schemes
MIMO	Multiple Input Multiple Output
MMDS	Multichannel Multipoint Distribution Service
MPEG	Moving Picture Experts Group
MS	Mobile Station
NLOS	Non Line Of Site

nrtPS	non-real-time Polling Service
nrtPSQ	nrtPS Quantum
numUcastPolls	Number of Unicast Polls
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency-Division Multiple Access
P2P	Point-to-Point
PDFPQ	Preemptive Deficit Fair Priority Queue
PDU	Protocol Data Unit
PHS	Payload Header Suppression
PHSF	payload Header Suppression Field
PHSI	Payload Header Suppression Index
PHY	Physical
PMP	Point-to-Multipoint
Q	Quantum
Q_{crit}	Quantum Critical
QLcurrent	The Current Length of the rtPS Queue
QLthreshold1	Low threshold of rtPS queue
QLthreshold2	The high threshold of rtPS queue
QoS	Quality-Of-Service
RED-based DFPQ	Random Early Detection based Deficit Fair Priority Queue
RF	Radio Frequency
RPF	Request Per Frame
RSSI	Received Signal Strength Indication
RTG	Receive/Transition Gap

rtPS	Real-time Polling Service
rtPS_Threshold	Maximum value of rtPS counter
rtPSQ	rtPS Quantum
RW	Request and Wait
SC	Single Carrier
SDMA	Spatial Division Multiple Access
SDU	Service Data Unit
sFlow	Service Flow
SNR	adaptive based on Signal to Noise Ratio
SS	Subscriber Station
Talloc	Last Allocation Time
Tarrival	The Arrival Time for the packet to the rtPS queue
Tcur	Current Time
TDD	Time Division Duplex
Temp	Temporary
Tf	Frame duration
timeDiff	Time Difference
Tlatency	The maximum latency of rtPS
Tlatency	Latency Time
Tnow	The current time
TTG	Transmit/Receive Transition Gap
UCD	Uplink Channel Descriptor
UGS	Unsolicited Grant Service
UL	Uplink
UL-MAP	Uplink MAP

VBR	Variable Bit Rate
VOIP	Voice Over Internet Protocol
WiMAX	Worldwide interoperability for Microwave Access

QOS-AWARE SCHEDULING SCHEME FOR IEEE 802.16 TRAFFIC

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ABSTRACT

Worldwide interoperability for Microwave Access (WiMAX) is one of recent Broadband Wireless Access (BWA) technologies. It is aimed to support high quality of service (QoS) for its different data traffic. Although different classes of data types are defined by the IEEE 802.16 standard, the scheduling scheme is left for vendors to specify. As the existing scheduling algorithms have not solve the starvation of nrtPS and BE data traffic, a scheduling algorithm named Extended Adaptive Deficit Priority Queue (EADPQ) is proposed in this thesis. The EADPQ is divided into two parts to enhance QoS in WiMAX networks. It starts by polling Best effort (BE) traffic and ignoring contention based approach to request bandwidth for BE traffic. Additionally, If the data of non-real-time Polling Service (nrtPS) and BE is queued for less than a specific threshold, the data continues to be queued until specific time. The second part of the proposed work is distributing bandwidth dynamically and fairly between different data types. A comparison is held between the proposed approach with other recent

scheduling algorithms. By conducting many experiments with different scenarios, it is concluded that EADPQ has increased the overall throughput of the network (rtPS, nrtPS and BE throughputs) by up to 5%. Although the focus was on nrtPS and BE data traffic, a new enhanced step to the throughput of rtPS traffic is made by 8%. Furthermore, EADPQ has ignored the huge starvation of nrtPS and BE data traffic by increasing the throughputs and decreasing the delays. For nrtPS data traffic, the enhanced throughput reached 13% while decreasing the delay by 22%. Additionally, the throughput of BE data traffic is enhanced by 70% and the delay was decreased by 8%. Finally, it is concluded that EADPQ can be used in real life networks which have different variations of data traffic and different loads.

CHAPTER 1: INTRODUCTION

1.1 Motivation and Objective

One of the most emerging technologies for Broadband Wireless Access (BWA) in metropolitan areas is Worldwide interoperability for Microwave Access (WiMAX). Because WiMAX has advantages of, high-speed Internet access and multimedia heavy loaded traffic, high resource utilization, easy implementation, and low cost, it is a viable alternative to the cable modem and DSL technologies. Moreover, WiMAX has long distance transmissions and support sophisticated Quality-Of-Service (QoS) at the Medium Access Control (MAC) layer that provides high data rate applications with an enhanced QoS characteristics. To support multimedia traffic, the Medium Access Control (MAC) protocols coordinates the transmission of traffic flows. The channel has diverse characteristics of users and traffic flow requirements so that it motivates the designers to design an efficient MAC layer protocols that can improve the system performance due to the channel and traffic dynamics. As a result, researchers designed a lot of bandwidth allocation algorithms to improve the performance and efficiency of data use.

The IEEE 802.16 standard defines two main modes: Point-to-Multipoint (PMP) and Mesh. In mesh mode a subscriber station (SS) is allowed to communicate through other stations and the base station (BS). But in PMP mode it is allowed to communicate only through the BS. WiMAX service providers are anticipated to use the PMP mode to connect customers to the Internet (Sayenko et al., 2008). Therefore, BS has the task to provide QoS for all the data going from and to SSs.

Basically BS is responsible for providing QoS for all data sent and received using scheduling algorithms for both uplink (UL) and downlink (DL) directions. That means the algorithm in the BS will translate the QoS requirements of the SSs into frame's slots. Then BS broadcasts the final results of the scheduling algorithm by using the uplink map (UL-MAP) and downlink map (DL-MAP) messages which are located at the beginning of each frame. The IEEE 802.16 WiMAX standard does not specify any scheduling algorithm to provide QoS for data traffic. Therefore, scheduling is considered a rich research area to investigate. As a result, WiMAX service providers will be able to invent their own scheduling algorithms or use any of the proposed scheduling approaches.

The IEEE 802.16 QoS has classified the traffic in the WiMAX network into five categories: Unsolicited Grant Service (UGS), extended real-time Polling Service (ertPS), real-time Polling Service (rtPS), non-real-time Polling Service (nrtPS), and Best Effort service (BE). According to their priority levels the five categories are classified as UGS > ertPS > rtPS > nrtPS > BE. While UGS is used for Constant Bit Rate (CBR), rtPS and ertPS are used for Variable Bit Rate (VBR) traffic.

1.2 Problem Statement

In WiMAX each one of the above categories sends a bandwidth request to BS to be able to transfer its data. For ertPS and rtPS, bandwidth requests cause an overhead and additional access delay while nrtPS and BE traffic will suffer from starvation when requesting bandwidth because of the contention based method used for them. Because old scheduling algorithms (Shreedhar and Varghese, 1996), (Katevenis et al., 1991) do not consider the latency metric of real-time traffic, they cannot be used in WiMAX networks. The authors in (Lin et al., 2008) have performed a performance evaluation for some old scheduling algorithms in WiMAX network. However, The focus in most of the recent proposed scheduling algorithms is on transmitting real-time VBR video-audio data traffic. This is because in real time applications the data traffic must have minimum duration of delay, minimum jitter. In other words, if such data arrived after the deadline time it will be useless. Two studies in (Dhrona et al., 2008), (Abu Ali et al., 2009) were made on the performance of some scheduling algorithms in PMP WiMAX networks. They tried in (Abu Ali et al., 2009) to study the performance of these algorithms when using contention and piggyback techniques.

Many recent researches have been discussed in the literature are focusing on scheduling the rtPS traffic. Thus, such algorithms caused a big starvation in the lower priority queues, the nrtPS and BE data traffic, when applied to congested networks. This research investigates the shortcomings of the existed scheduling algorithms in terms of: the fairness of bandwidth allocation, delay of data, and throughput which are considered a major metrics to achieve QoS. This research also proposes a scheduling algorithm that enhances the throughput and avoids the starvation of nrtPS and BE data traffic.

1.3 Thesis Contributions

In this thesis the contribution is divided into two parts as follows:

- The first part is using polling scheme for BE traffic instead of contention based scheme. Additionally, to limit the heavy loaded bandwidth requests, we poll nrtPS and BE when their queues reached specific thresholds that are specified to each one. Otherwise, the data will be queued until it reaches defined time for each one. This part has enhanced the overall throughput of the network(rtPS, nrtPS and BE throughputs) by up to 5%.

- The second part of the proposed work is distributing bandwidth dynamically for rtPS, nrtPS and BE data traffic. The available bandwidth is distributed to each traffic in percentages according to the latency which is the maximum time for packets to be queued, size of bandwidth requested for rtPS traffic, and according to the size of bandwidth requests for nrtPS and BE traffic.

1.4 Thesis Organization

The thesis has six chapters. Chapter 1 illustrates the motivations, objective and problem statement of the thesis. After that a contribution of the thesis is listed. Chapter 2 discusses a background of WiMAX in more details. The WiMAX architecture, layers, and QoS requirements are discussed. Chapter 3 discuses the recent related work papers and their shortcomings. Afterwards, Chapter 4 presents the proposed work and simulation environment in details. The results of the experiments with the analysis is illustrated in chapter 5. Finally, chapter 6 concludes the thesis and draws future work.

CHAPTER 2: BACKGROUND

2 BACKGROUND

In this chapter a background of Worldwide interoperability for Microwave Access (WiMAX) has been introduced with details like WiMAX layering, QoS specification, and data classes.

2.1 IEEE 802.16 standard

IEEE 802.16 is the base of WiMAX technology. It was originally intended to serve as backhaul in Point-to-Multipoint (PMP) network architecture. Now it is supporting the different forms of mobility. The base station in WiMAX can theoretically provide a broadband wireless access in a range of up to 50 kms for fixed stations and 5 to 15 kms for Mobile Stations (MSs) with a maximum data rate of up to 70 Mbps (So-In et al., 2009). The Orthogonal Frequency Division Multiplexing (OFDM) (IEEE Standard 802.16-2004, 2004) physical layer is a physical format that is supported in the standard. It enables better Non Line Of Site (NLOS) performance compared to Single Carrier (SC) and Code Division Multiple Access (CDMA) physical formats. WiMAX Forum has chosen the 256 carrier OFDM format for the 802.16-2004 revision of the standard.

IEEE 802.16-2004 (IEEE Standard 802.16-2004, 2004) version, known as “Fixed WiMAX”, is the combination of two extensions, 802.16a and 802.16c, with some modifications. It supports both Time Division Duplex (TDD) and Frequency Division Duplex (FDD) services. One of the main enhancements in this extension of WiMAX is the concatenation and fragmentation of Service Data Unit (SDU) into Protocol Data Unit (PDU) which reduces the Medium Access Control (MAC) overhead in headers as well as utilizing bandwidth and meeting QoS demands through packet resizing. Payload Header Suppression (PHS) can also used to omit redundant header information (IEEE Standard 802.16-2004, 2004). Moreover, IEEE 802.16-2004 adds a substantial improvement to the polling mechanism. It allows the SSs to be polled individually or in groups. It also allows piggybacking bandwidth requests which reduces overhead of sending bandwidth request packets and collisions.

The 802.16e (IEEE Standard 802.16e/D9, 2005) standard known as "Mobile WiMAX" adds mobility support for the technology. This extension preserves all aspects of the "Fixed WiMAX" while adding support for mobile broadband wireless access. The standard here specifies the use of Orthogonal Frequency-Division Multiple Access (OFDMA) technology with support for different Fast Fourier Transform (FFT) system profiles. The OFDMA technology increases the resistance to multi-path interference by allowing the signals to be divided into many sub-channels. It also has the ability to subdivide the carriers to specific users. A new scheduling service called Extended Real-Time Polling Service (ertPS) was added in 802.16e. The ertPS combines the efficiency of the two scheduling services Unsolicited Grant Service (UGS) and Real-Time Polling Service (rtPS). It gives unsolicited bandwidth grants like UGS, but with dynamic sizes like rtPS. This yields a suitable service class for Voice Over Internet Protocol (VOIP) with silence suppression that support real-time service flows with variable size data packets. IEEE 802.16e adds MAC-support for sleep/idle-mode for mobile subscriber stations, paging, locating, power saving classes and defines messages for handover procedures (IEEE Standard 802.16e/D9, 2005).

The MAC-layer is built to support Ethernet, Asynchronous Transfer Mode (ATM) and IP traffic through its convergence layer, with the five levels of QoS at MAC level in the form of Constant Bit Rate grant (CBR), ertPS, rtPS, non-real-time Polling Service (nrtPS), and Best Effort (BE) (IEEE Standard 802.16e/D9, 2005). In WiMAX all SSs send their QoS requirements according to their needs through a request/grant scheme to the BS. The BS allocates bandwidth dynamically to either users or services and schedules the Data traffic in the MAC layer. However, certain types of bandwidth requests and ranging can be transmitted over contention periods rather than scheduling to increase flexibility and reduce latency. The MAC-layer also has a security sublayer that is responsible for performing encryption of MAC PDUs, authentication and encryption key exchange (IEEE Standard 802.16-2004, 2004).

2.2 WiMAX Architecture

There are some architectures involved with WiMAX like Point-to-Point (P2P), PMP, Line Of Sight (LOS), NLOS. These architectures along with wireless radio antennas are discussed in the following sections:

2.2.1 P2P Vs PMP

In P2P architecture one transmitter and one receiver exists. This architecture is used in WiMAX as a backhaul. It consists of a base station that acts as a transmitter and another one as a receiver. The highly focused beam between these two points of this architecture results in higher range and throughput radios comparing with PMP products. This makes it possible to cover a large geographical area (WiMax.com, Wireless architectures, 2010). In the other hand, the PMP architecture is based upon IEEE 802.16.2004 standard. In this architecture there are one transmitter and can service hundreds of dissimilar receivers (WiMax.com, Wireless architectures, 2010).

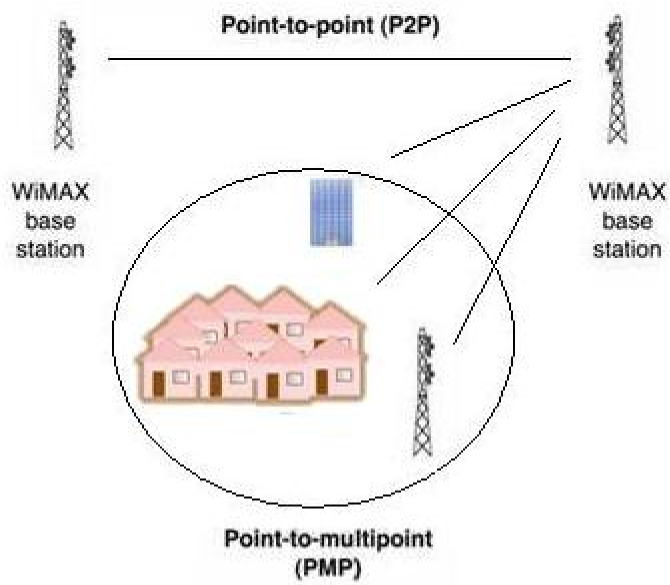

Figure 1: Point-to-Point and point-to-multipoint configurations

2.2.2 LOS Vs NLOS

The WiMAX technology functions best in line of sight situations. However, The earlier technologies (Local Multipoint Distribution Service (LMDS), Multichannel Multipoint Distribution Service (MMDS) for example) which offers acceptable range and throughput to subscribers, do not support LOS to the BS.

The earlier technologies limited the number of subscribers that could be reached. And due to the high cost of base stations and Customer Premises Equipment (CPE) many plans failed. Because buildings between the antennas and the Base Stations (BSs) or the subscribers in an urban environment diminish the range and throughput, the signal will still be strong enough to deliver adequate service. The ability of WiMAX to deliver services in NLOS method enables it to reach many customers in high-rise buildings with low cost per subscriber because many customers can connect to one base station (WiMax.com, Wireless architectures, 2010).

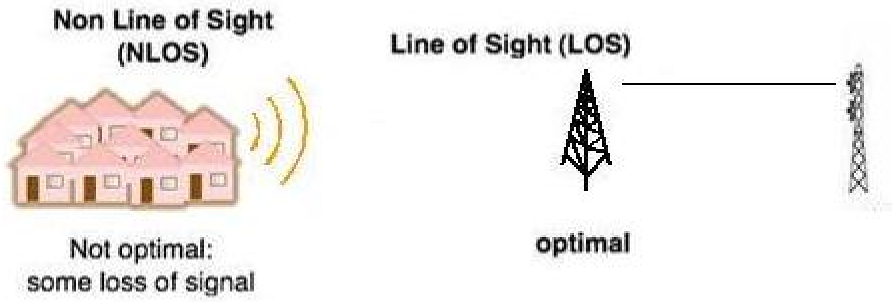

Figure 2: The difference between line of sight and non-line of sight

2.2.3 WiMAX Radios & Antennas

The WiMAX radio is the core of WiMAX that might be thought of as a networking device which is similar to a router or a bridge in that it is managed by software and is composed of circuit boards containing very complex chip sets. A radio contains both transmitter and receiver. It generates electrical oscillations at a carrier frequency (usually between 2 and 11 GHz). WiMAX architecture is consisting of two components: radios and antennas. WiMAX products offer a base station radio separate from the antenna. Many CPE devices are also two piece devices with an antenna on the outside of the building and subscriber station inside of the building.

The advantage of putting the radio inside the buildings is that the radio is protected from extremes of heat cold and humidity which reduce the performance and durability of radio. In addition, putting the antenna outdoors optimizes the performance of the wireless connection between transmitter and receiver especially in line of sight scenarios. The WiMAX's antenna and radio is connected via a cable known as a "pigtail". This cable must be as short as possible to reduce loss of data. For example popular LMR-400 cable lose about 1 dB for every 10 feet of cable (WiMax.com, WiMAX radios, 2010).

2.2.4 WiMAX Antennas

WiMAX antennas are just like any other antennas for cell phone or TV. It is designed to optimize performance for a given application. Figure 3 illustrates the three main types of antennas used in WiMAX deployments, an omni directional, sector and panel antenna.

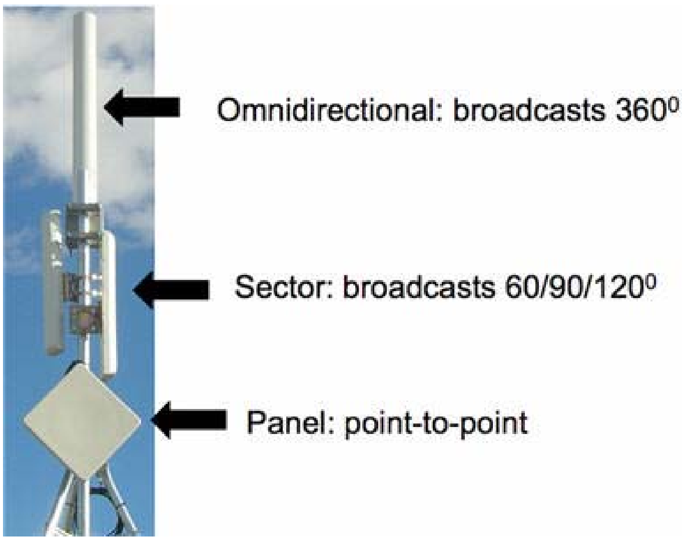

Figure 3: Different antenna types (WiMax.com, WiMAX Antennas, 2010).

2.2.5 Subscriber Stations (SSs)

Subscriber station is the technical term for CPE. The generally accepted marketing terms now focus on either "indoor CPE" or "outdoor CPE". An outdoor CPE offers somewhat better performance over indoor CPE because it does not face concrete or brick walls in WiMAX reception. Subscriber can install indoor CPE so that service provider need not provide installation for it. Consequently, it reduces waiting time needed for installation by service provider. Figure 4 and Figure 5 show typical outdoor and indoor CPE (WiMax.com, Subscriber Stations, 2010).

Figure 4: Outdoor antenna (WiMax.com, Subscriber Stations, 2010)

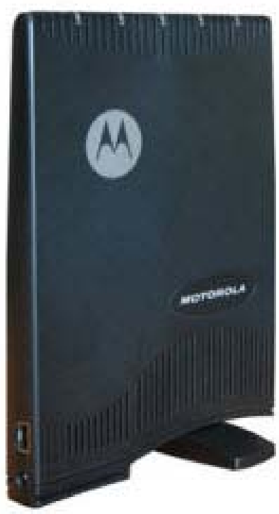

Figure 5: Indoor antenna (WiMax.com, Subscriber Stations, 2010)

2.3 WiMAX Characteristics

The following are the principal characteristics of WiMAX systems (So-In et al., 2009), (Benefits of WiMAX), (IEEE Standard 802.16e/D9, 2005):

- Uses OFDMA.
- Support NLOS that operates at 2 to 11 GHz, which at this lower frequency it is less susceptible to obstacles.
- Uses LOS that can go as high as 66 GHz which leads to greater bandwidth since the signal is stronger and more stable.
- Uses Multiple Input Multiple Output (MIMO).
- Uses TDD and FDD.
- Per subscriber adaptive modulation.
- Uses advanced coding techniques: like space-time coding and turbo coding Strong security techniques.
- Support mobility, roaming and meshing.(IEEE Standard 802.16e/D9, 2005)
- Can provide a long communication range of up to 30 miles.
- Reduces the expenses of expanding the network.
- Support high levels of QoS.
- Have multiple QoS classes that vary between data, voice and video services.
- High number of simultaneous sessions.

2.4 WiMAX Topology Support

The WiMAX technology supports two types of network topologies, PMP networks and a form of decentralized network topology called mesh. In PMP mode the

SSs only talk to the BS and the traffic goes through the BS, while in Mesh mode the SSs communicate with each other directly or by multihop routing protocol through other SSs in the network. However, in Mesh network the node which can access the backhaul connection is called Mesh BS, while the remaining nodes of the system called Mesh SSs. Although the mesh has node as Mesh BS, this node also has to coordinate broadcasts with other nodes in the network. In WiMAX Mesh mode, all the communications in the network are controlled by one of three ways, using a centralized scheduling algorithm, using a distributed scheduling algorithm or using a combination of these two types of algorithms (IEEE Standard 802.16-2004, 2004).

In the centralized scheduling, mesh relies on the 'Mesh BS' to gather resource requests of SSs within a certain range and allocates the requests with individual capacity. This capacity is shared with other Mesh SSs that relay their data through other SSs according to the Mesh BS. In contrast to PMP, mesh mode QoS classification is done on a packet-by-packet basis. While in PMP mode QoS classification is associated with links. Thus, only one link exists between any two communicating mesh nodes (IEEE Standard 802.16-2004, 2004).

2.5 IEEE 802.16 PHY Layer

This thesis will focus on the IEEE 802.16 PHY and MAC layers. Starting in this section with the physical layer of the 802.16 standard that is presented in Figure 6 (WiMax.com, Objections to WiMAX, 2010).

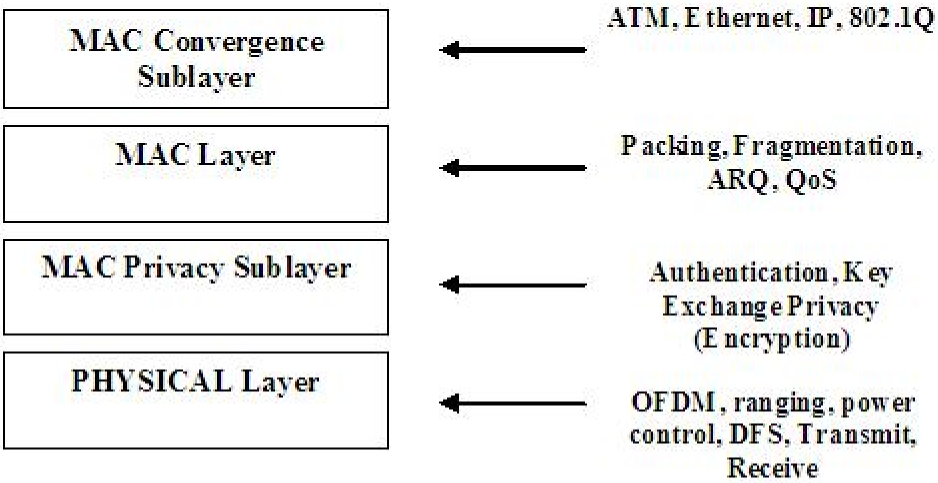

Figure 6: Layering in IEEE 802.16

In the IEEE 802.16-2004 standard, the PHY layer is defined for frequencies ranging from 2 to 66 GHz. The sub-range 10-66 GHz requiring LOS propagation while the NLOS propagation is in the 2-11 GHz frequency range (OFDM Variants 2–11 GHz, 2009). The PHY layer technologies of WiMAX are OFDM and OFDMA. OFDM has recently gained popularity for high-speed bidirectional wireless data communication. It is a multi-carrier transmission technique (Wolnicki, 2005). OFDM reduces the required bandwidth by squeezing multiple modulated carriers together and at the same time keeping the modulated signals orthogonal to each other so that they do not interfere with each other. The OFDM technique is based on Frequency Division Multiplexing (FDM) which uses many frequencies to transmit signals in parallel. OFDM is more efficient than FDM as it allows sub-channels to be spaced closer to each other by finding orthogonal frequencies.

On the other hand, OFDMA assigns different users to certain sub-carriers. Each SS has sub-channels and each sub-channel consists of a group of sub-carriers. The IEEE 802.16-2004 standard specified Both TDD and FDD. In TDD technique the system receives and transmits within the same frequency channel. It assigns time slices for transmitting and receiving modes. However, in FDD technique, two separate frequencies are required for transmitting and receiving. These frequencies usually separated by 50 to 100 MHz within the operating band. The frame structures of downlink and uplink in FDD are similar except that they are transmitted in separate channels. When using half duplex FDD (H-FDD) at the SSs, the BS must make sure to not schedule the SSs to transmit and receive at the same time. Figure 7 and Figure 8 show illustration of OFDM and OFDMA.

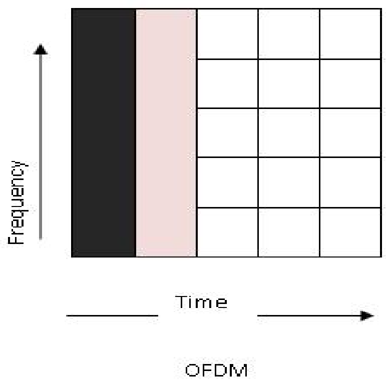

Figure 7: IEEE 802.16 PHYs: OFDM

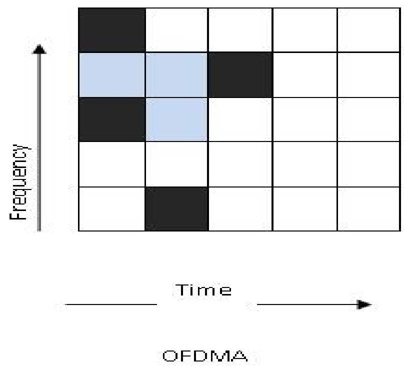

Figure 8: IEEE 802.16 PHYs: OFDMA

Adaptive Antenna System (AAS) is used in WiMAX to specify the beam-forming techniques where a set of antennas are used at the BS to increase the gain to the SSs at the same time while reducing interference to and from other SSs. AAS can also be used to enable Spatial Division Multiple Access (SDMA) so that multiple SSs which are located in different spaces can receive and transmit on the same sub-channel simultaneously.

2.6 IEEE 802.16 MAC Layer

This section discusses the upper two layers, the convergence and MAC layers, of the 802.16 standard that is presented in Figure 6. These two layers form the fundamental parts of the air interface which governs how the limited radio resources are shared by base stations and subscriber stations.

2.6.1 Convergence Sublayer

The Convergence Sublayer (CS) performs two main tasks, Packet Classification and PHS. The IEEE 802.16-2004 (IEEE Standard 802.16-2004, 2004) contains two specifications for convergence layer, ATM and packet CS. This section covers the packet CS, but much of the principles applies also to ATM CS.

The process when the CS receives a higher layer packet and maps this packet to a service flow, which is a connection with a set of QoS parameters, is called Packet Classification. Since the delivery of the packets needs appropriate QoS constraints, each service flow is classified and associated with specific QoS parameters. This classification is made based on different criterion, such as destination or source IP-addresses. If a packet matches a criteria it is delivered to a MAC connection which has been matched to that criteria. In other words the classification results in an appropriate connection identifier (CID) for a connection, as seen in Figure 9 (IEEE Standard 802.16-2004, 2004). Several classifiers may exist for the same service flow, and since they can overlap with each other they are explicitly ordered.

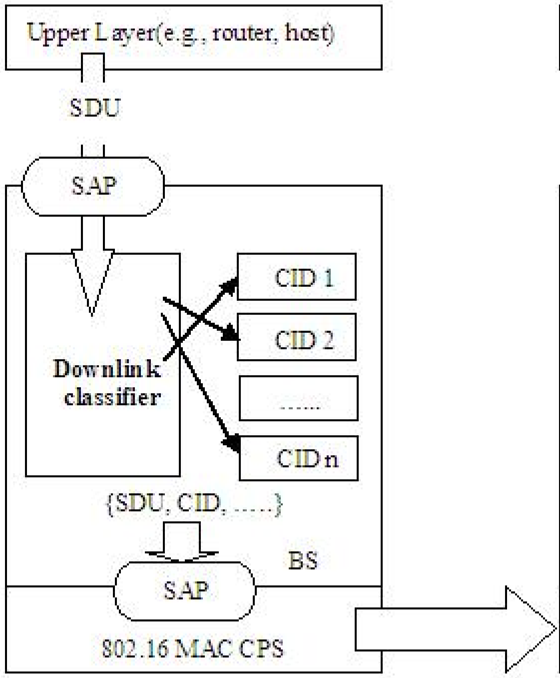

Figure 9: Classification and CID mapping (BS to SS)

An optional capability called PHS is used to remove repetitive or redundant information from higher layer packets headers. When packets are classified, they can also be mapped to PHS Rules. The packet header information is compared with a Payload Header Suppression Field (PHSF). If there is a match between the header bits and the PHSF, some of these bits can be masked. Such bits that are desirable to mask can be higher layer static fields, such as IP-address. Dynamic fields can be left as it is without change by using the PHS Mask, which specifies the bits that are not to be suppressed. In the other hand, when receiving a packet applied to PHS, the receiver unmasks the appropriate bits and reassembles the packet headers before delivering the packet to the higher layers. As a result the information needed for PHS is needed on both receiving and sending entity.

As a reference to the appropriate PHSF a Payload Header Suppression Index (PHSI) is added to the packet on the receiving side. Rules of PHS are created dynamically through management messages like Dynamic Service Change (DSC) or Dynamic Service Addition (DSA) (IEEE Standard 802.16-2004, 2004). These rules can be created over time by making some fields available after being unknown at creation. The new available fields are added to the rule at later time. Then after passing through the CS the packet will be delivered to the appropriate service flow in the MAC-layer with the format shown in Figure 10. This format is called a MAC SDU in the standard. and it contains the PHSI and the higher layer PDU. The receiver uses the PHSI to select the appropriate PHSF for unmasking the header. Now the MAC SDU will be in the MAC-layer which is responsible for delivering the packet to the receiver over the air interface (IEEE Standard 802.16-2004, 2004).

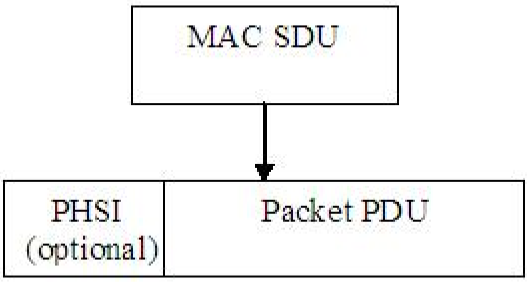

Figure 10: MAC SDU Format

2.6.2 Medium Access Control

While the PHY layer handles specific radio functions like modulation and physical frequency allocations, the MAC-layer distributes the limited radio spectrum resources between the base stations and subscriber stations. These Modulation and Coding Schemes (MCS) are adaptive based on Signal to Noise Ratio (SNR). SNR of the receiver is then transferred to the BS via MAC management messages. It is important to distribute the radio bandwidth efficiently since the medium is limited by the radio bandwidth. Some of the functions performed by the MAC layer are:

- QoS
- Connection management
- Scheduling of data
- Handovers, Idle/sleep mode

This section covers MAC in PMP mode, but much of the section is applicable to Mesh mode operation as well. The 802.16 MAC-layer is connection oriented and all data communication is associated with a connection. The QoS parameters together with the connection make up a service flow, which is an essential term in the standard. QoS is maintained through five different QoS classes which are different in their characteristics with each other.

- **Framing**

The MAC-layer support both TDD and FDD framing, where TDD separates both uplink and downlink by time while FDD separates them by frequency. The frame size can be varied according to the different physical profiles. Between the uplink and downlink there are guard spaces called Transmit/receive Transition Gap (TTG) and Receive/transition Gap (RTG) to allow switching the radio between transmitting and receiving and modes. According to (WiMAX Forum, 2006) Figure 11 shows an OFDMA frame structure.

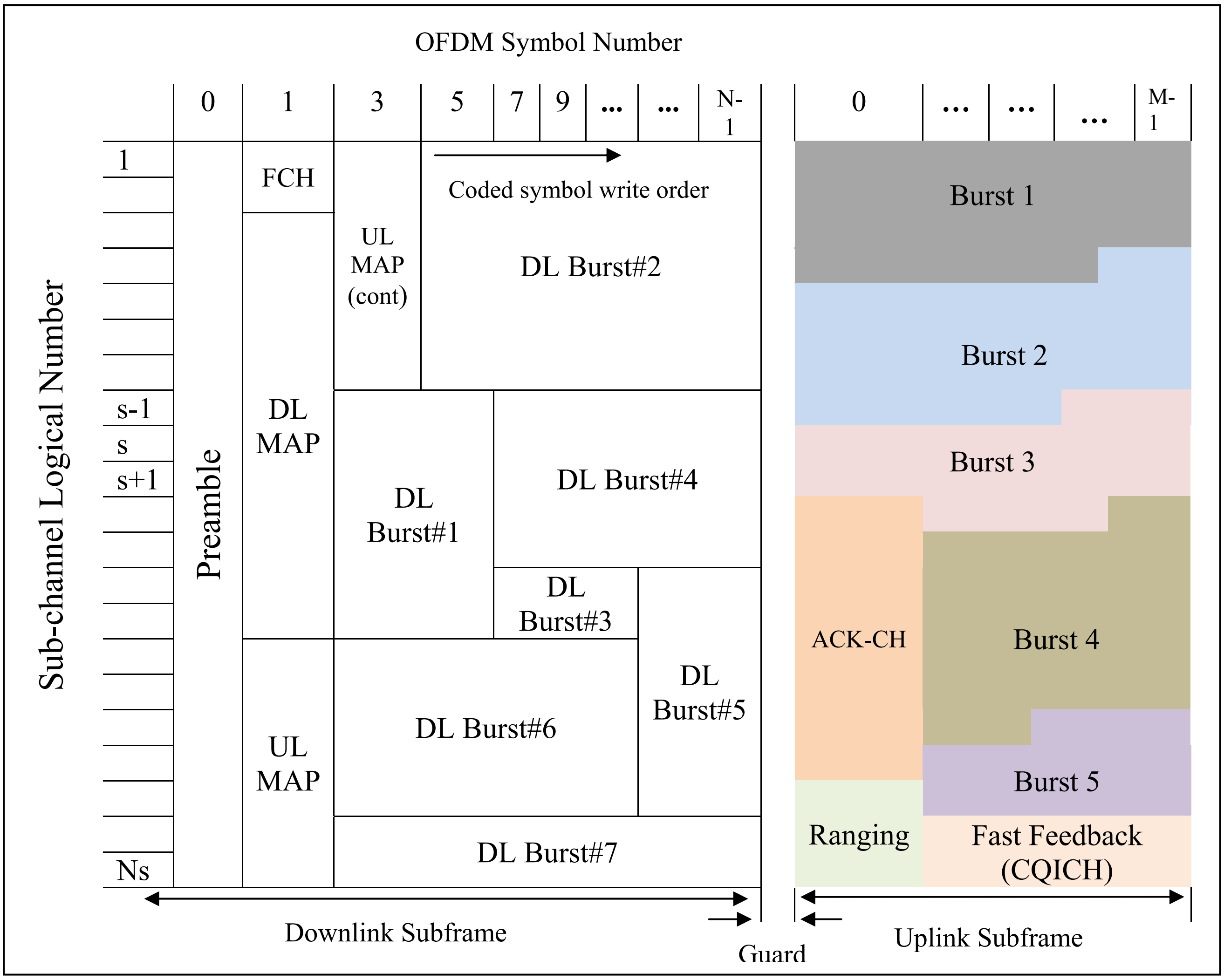

Figure 11: WiMAX OFDMA Frame Structure

The DL consist of several physical bursts of different modulation/coding. These bursts are addressed to different CID which identify burst addressed to individual SS. The DL-MAP, is always comes first. It describes the content of the downlink which contains information elements (IE) regarding the downlink data bursts, i.e transmissions from the BS to different SSs or MSs. After the DL-MAP it comes the UL-MAP which functions in much the same way as the DL-MAP except with the description of the uplink bursts. These burst have been allocated by the BS for SSs or MSs to send their uplink data. The IEs in the UL-MAP describe the type of burst and in which time the allocated burst will start.

In Uplink channel descriptor (UCD) and Downlink channel descriptor (DCD) messages detailed information for the uplink and downlink bursts are supported. While the IEs in the UL-MAP and DL-MAP contain references to what type each burst is, UCD and DCD messages contain the details on how each type of burst should be decoded.

- **Scheduling**

The scheduling algorithms can be managed through the DSA and DSC management messages by associating a connection with each scheduling service so that the BS can anticipate the needs and behavior of each connection. The scheduling algorithm defines which mechanism will be used for each connection and how to send grants to SSs.

- **Bandwidth Requests**

Before sending data an SS may request bandwidth either by sending a Bandwidth Request message or by an optional Piggyback Request in a Grant Management subheader. Each type of service classes send its bandwidth requests independently. These requests can be sent in any of the uplink bursts, except during the initial ranging interval. These requests can be either incremental or aggregate. In incremental request the new requests adds the request bandwidth to the amount already allocated to the SS while an aggregate request replaces the current value with the new value of the request. Piggyback requests can only be incremental. When using piggyback request, the current bandwidths were requested via piggybacking in previous data frames. For example a bandwidth request for packet n can be piggybacked on the previous packet number n−1.

By considering a packet with an interval shorter than the WiMAX frame size, only the bandwidth for the first packet of the data stream will be requested during the contention phase and all other bandwidth requests can be piggybacked.

- **Polling**

There are three polling types, unicast, multicast, and broadcast according to the bandwidth availability. Unicast polling can only be used if the bandwidth is enough for polling all SSs individually. However, some SSs may be polled by multicast polling or broadcast polling. The BS send unicast polling with bursts directed at the SS's Basic CID and send broadcast or multicast polling with contention interval bursts that has standard defined CIDs.

2.6.3 Requirement of QoS

The WiMAX technology has to handle the requirements for different types of traffic. It handles very high data rate applications, such as VoIP and video or audio streaming, as well as low data rate applications, such as Hypertext Transfer Protocol (HTTP). WiMAX needs to handle all of these traffic at the same time and considering that some applications cannot work without quality of service. For those applications that need quality of service , some delay may be acceptable, but too much can make the application unusable. Scheduling algorithms in WiMAX have been designed to allow a variety of QoS requirements. The 802.16 must be flexible and efficient for the different traffic requirements because of the varied bandwidth and latency of the applications of end users.

2.6.4 Service classes supported in WiMAX

WiMAX support five different data traffic types for VOIP and video-audio streaming and others.

- **Unsolicited Grant Service (UGS)**

Real-time CBR applications are supported in UGS. These applications generate fixed size data packets on a periodic basis. The base station (BS) assigns unsolicited fixed bandwidth grants at periodic intervals for these packets based on their maximum sustained traffic rate, therefore scheduling is not required for this class. This class supports VoIP without silence suppression (IEEE Standard 802.16-2004, 2004).

- **Real-Time Polling Service (rtPS)**

This service is appropriate to real-time applications which produce variable-size data packets like Moving Picture Experts Group (MPEG). The BS allows the SSs to issue bandwidth requests on a periodic basis by means of a polling mechanism (IEEE Standard 802.16-2004, 2004).

- **Extended Real-Time Polling Service ertPS**

This service is similar to rtPS service except that the BS can ensure a default bandwidth according to the maximum sustained traffic rate as in UGS and dynamically provide additional resources, therefore scheduling is not required here. This service support VOIP with silence suppression (IEEE Standard 802.16e/D9, 2005).

- **Non-Real-Time Polling Service (nrtPS)**

This service is appropriate for the delay tolerant applications that generate variable size data packets and require a minimum data rate. The BS typically polls its flows to issue bandwidth requests on a regular basis if it is possible. It can use contention slots to ask for bandwidth grants. The polling interval depend on network traffic load so it is not guaranteed. nrtPS supports non real-time connection, like File Transfer Protocol (FTP) (IEEE Standard 802.16-2004, 2004), (Ghazal et al., 2008).

- **Best Effort (BE)**

This service class is appropriate for traffic with weak QoS requirements. BE Flows can only use contention slots to deliver their bandwidth requests. This class provides services for best effort traffic like HTTP (IEEE Standard 802.16-2004, 2004), (Ghazal et al., 2008).

2.6.5 QOS specification for different service classes

Some important QoS metrics are latency, jitter and maximum sustain traffic rate which different service classes use to improve their performance in the network. These metrics are:

- **Latency**

The granularity of the physical layer chain is causing the end-to-end packet transmission time. Latency is also affected by various QoS protocols, how packet queued, and user characterizations. Maximum Latency is defined as the maximum interval between the reception of a packet by the BS or the SS and the forwarding of the packet to its Radio Frequency (RF) Interface (IEEE Standard 802.16-2004, 2004).

- **Jitter**

The Jitter is caused when packets arrive at different times due to different routes taken in the path or due to queuing. A memory buffer is used to typically address the Jitter by storing early arriving packets, and then concatenates later arriving packets. Thus smoothes the voice arriving at the receiver. The Tolerated Jitter defines the maximum delay variation for the connection.

- **Maximum Sustained traffic Rate**

The Maximum Sustained Traffic Rate defines the maximum rate of the service. In UGS, the Maximum Sustained Traffic Rate is also the minimum rate reserved for the service flow. With the fixed grant size, this parameter determines the intervals at which BS issues periodic data grant. For the fixed length data carried by the service flow these grant sizes must be large enough to contain them (IEEE Standard 802.16e/D9, 2005).

The following table summarize all classes of WiMAX with their QoS parameters and examples for the applications they support (WiMAX Forum, 2006), (Ghosh et al., 2008).

Table 1: Service classes in WiMAX

Class	Application	QoS parameters
UGS Unsolicited Grant Service	VoIP, E1; fixed-size packets on periodic basis	• Maximum Sustained Rate • Maximum Latency Tolerance • Jitter Tolerance
rtPS Real-Time Polling Service	Streaming Audio or Video	• Minimum Reserved Rate • Maximum Sustained Rate • Maximum Latency Tolerance • Traffic Priority
ertPS Extended Real-Time Polling Service	Voice with activity Detection (VoIP)	• Minimum Reserved Rate • Maximum Sustained Rate • Maximum Latency Tolerance • Jitter Tolerance • Traffic Priority
nrtPS Non-Real-Time Polling Service	File Transfer Protocol (FTP)	• Minimum Reserved Rate • Maximum Sustained Rate • Traffic Priority
BE Best-Effort Service	Data Transfer, Web Browsing, etc.	• Maximum Sustained Rate • Traffic Priority

CHAPTER 3: LITERATURE REVIEW ON RELATED WORK

3 LITERATURE REVIEW ON RELATED WORK

Researches in PMP WiMAX networks can be classified into two main classifications (Sekercioglu et al., 2009):

- Admission control and packet scheduling which focuses on implementing the uplink and downlink schedulers at the BS and SS scheduler.
- Signaling and Internetworking which focuses on improving signaling and Internetworking between WiMAX and other networks like Wi-Fi. This research focuses on Admission control and packet scheduling.

In (Nagaraju and Sarkar, 2009) the authors have used Earliest Deadline First (EDF) algorithm for enhancing rtPS. This algorithm tends to starve BE traffic. Their proposed algorithm tried to solve this starvation via fragmenting the original packets of rtPS traffic in order to fit the time slots of the frames. As a result, the last fragment of each packet of rtPS traffic will have some empty spaces. The authors tried to use these empty spaces for sending BE traffic. The disadvantages of their method are as follows:

- It cannot serve much BE traffic by just using the last fragments of the packets.
- Additional buffer at BS.
- Additional computation at BS and SS.
- Extra header information.

Although the authors tried to justify these drawbacks, their proposed method still generate more processing time for serving very little BE traffic.

The research in (Monteiro et al., 2009) made an enhancement to an old algorithm which is called Round Robin. It is trying to differentiate the traffic inside each QoS class category via prioritizing connections from terminals. The priority is given to the highest received power signal commonly called Received Signal Strength Indication (RSSI). According to this rule, terminals with equal service class connections and traffic priorities will be served in such a way that privileges the ones with better radio conditions, or are closer to the transmitting antenna. The disadvantage of this algorithm is that it concentrates on the strength of the signal regardless of the traffic type.

In (Fantacci et al., 2009) three different resource allocation algorithms are proposed by taking into account the QoS constraints and the user's channel conditions. They considered simple strategy for choosing the most suitable MCS according to the channel conditions of the assigned subcarriers. In this study they only focus on the overall traffic load and throughput which means, they did not study the behavior of each traffic type separately.

As UGS and rtPS traffic have unicast polls slots granted from BS to request their bandwidth, nrtPS has unicast and contention poll slots. But in BE traffic only contention based polling is allowed which most of the time is responsible for causing a starvation while requesting the bandwidth from the BS. Because of that, the authors in (Kim et al., 2009) developed three bandwidth request and grant schemes for BE traffic. The first one is Request Per Frame (RPF). In RPF, an SS attempts to send bandwidth request for every frame when it has packets to send. It is simple and easy to implement. This scheme is efficient when the number of request slots is enough for all SSs. If these slots are not enough, collisions will occur and this causes unnecessarily delay.

The second scheme is Request and Wait (RW). In this scheme the SS waits several slots before sending its bandwidth request in order to avoid collisions. The number of waited slots is approximated from the history of the previous collisions.

The third scheme is the Grant Without Request (GWR). In this third scheme a base station allocates the bandwidth for each SS without the bandwidth requests. The allocated bandwidth in GWR is estimated by the BS from the sending rate of the SS. This scheme has two advantages the first one is that the SS does not need a function for bandwidth request. The second advantage is that it is suitable for large scale networks since no need for bandwidth request slots. The major drawback of this scheme is wasting bandwidth when the sending rate is highly fluctuated so that it is hard to estimate the required bandwidth.

As found in (Chen et al., 2009) the authors proposed two mechanisms for studying the VBR video traffic transmission. They divided the uplink bandwidth into several intervals and each interval represents a traffic state. The bandwidth request process is incurred only when the traffic state is changed. They also used two reserved bits in the generic MAC header of IEEE 802.16 BWA systems as piggyback bits so that the information of video traffic state transition can be sent to the BS without extra overhead. But they did not examine the effects of their method in nrtPS and BE traffic services and whether it caused starvation to them or not.

In Deficit Fair Priority Queue (DFPQ) scheduling algorithm (Chen et al., 2005), counter sizes are set with different quanta which are fixed for all service classes to differentiate service classes. The quantum of rtPS is larger than that of other classes because rtPS has higher priority. Moreover, the quantum of the BE class is the smallest since BE has the lowest priority. Every service class quantum is assigned to its Deficit Counter (DC). After setting the counters, service classes start transmitting data based on their counters by turns. When every service class uses up its counter in one round, the deficit counters are added by their quantum in the second round until the frame is over.

Although DFPQ gives rtPS more transmission opportunities to deliver its data packets, it results in the delay for lots of packets based on their delay requirement. That is because if there are packets need to be served urgently and the quantum for rtPS traffic has been finished, these packets will stay in queue for next frames and therefore will cause delay for these packets.

To reduce the delay of rtPS in DFPQ, the authors in (Safa et al., 2007) proposed a scheduling scheme known as Preemptive Deficit Fair Priority Queue (PDFPQ) which is based on DFPQ. In contrast, PDFPQ uses an extra quantum called quantum critical (Q_{crit}) for rtPS in order to decrease the delay of rtPS. Q_{crit} is a percentage of the original quantum of rtPS which is set at the beginning of every frame. PDFPQ will monitor the data packets of rtPS as soon as it arrives. The authors put three different scenarios that are an enhancement for (Chen et al., 2005). One of the problems in their study is the many calculations made. But the major problem of their study is that it was performed in four frames duration which is too low to measure the starvation of nrtPS and BE data traffic and even difficult to measure the behavior of the network.

The authors on (Yu et al., 2008) tried to avoid the overhead of monitoring done in (Safa et al., 2007) and to enhance the rtPS traffic. They proposed Adaptive Deficit

Priority Queue (ADPQ) method which is based on one QoS parameter called (Maximum Latency) defined in the standard for the rtPS traffic. First, they monitor all the packets in the rtPS queue and then send them all if they are critical (reached the maximum latency). They continue sending rtPS traffic if critical until they reach the threshold they specify which is 2/3 of the total bandwidth. Their algorithm is as Figure 12:

```
Set rtPS_threshold
Set DC [i] = Q[i] for each service class
Set Maximum_latency of rtPS

If(queue type is rtPS)
        While(Tf + (Tnow - Tarrival) > Tlatency)
                set DC.rtPS = DC.rtPS + Packet_size[i]
        If(DC.rtPS > rtPS.threshold)
                set DC.rtPS = rtPS.threshold
        If(DC.rtPS – Q.rtPS > Q.rtPS)
                set DC.rtPS = DC.rtPS – Q.rtPS

        While(DC.rtPS > 0 and rtPS not empty)
                Send rtPS packets;
        While(DC.nrtPS > 0 and nrtPS not empty)
                Send nrtPS packets;
        While(DC.BE > 0 and BE not empty)
                Send BE packets;
        Adjust all DC counters
        Next scheduling round
```

Figure 12: ADPQ algorithm

All parameters they used are explained as follows. Tf means frame duration, Tnow is the current time when the algorithm checks the packet, Tarrival is the time when the packet arrives to the rtPS queue, and Tlatency is the maximum latency of rtPS which is the maximum time that the packet can remain in the queue. The DC is the deficit counter (Chen et al., 2005) which is the original quantum (Q) but it is adjustable for rtPS traffic only. Finally, the rtPS_Threshold indicates the maximum value of rtPS's counter and it is set to 2/3 of the allocated bandwidth for the SS. Because their method focuses on rtPS by serving rtPS traffic heavily in all cases, they cause big starvation for nrtPS and BE in cases of heavy loaded traffic of both. They used three SSs only to send data which is not enough to measure a heavy loaded environment. Moreover, the number of applications they used are not enough to judge that their scheduling algorithm is suitable. For example what will happen if there is no BE traffic is sent or a heavy loaded traffic of BE is sent? What if data rates for all traffic types are sent equally?

Random Early Detection based Deficit Fair Priority Queue (RED-based DFPQ) (Ting et al., 2009) is an uplink scheduler that uses DCs for each rtPS, nrtPS, and BE service class. The deficit counter for rtPS service class is adaptively changes according to the queue length as illustrated in Figure 13. At the beginning of every frame (round) the scheduler checks the rtPS queue and sets its deficit counter. If the current length of the rtPS queue (QLcurrent) is less than the low threshold of rtPS queue (QLthreshold1), the DC value will be set to the minimum DC (DCmin). In the second case, if the QLcurrent is more than QLthreshold1 but less than the high threshold of rtPS queue (QLthreshold2), DC will be equal to the dynamic DC (DCdynamic). The DCdynamic can be calculated by the following equation:

$$DCdynamic = QrtPS + (QLcurrent - QLthreshold1 \, / \, QLthreshold2 - QLthreshold1) * QrtPS$$

where QrtPS is the original fixed quantum of the rtPS service class. In the last case, if the QLcurrent is more than QLtreshold2, then the DC for rtPS will be set to the maximum queue length of rtPS (DCmax) which is two times DCmin.

```
set DC [i] = Q[i] for each service class
set QLthreshold1 as the low threshold percentage of the total queue length of rtPS
set QLthreshold2 as the high threshold percentage of the total queue length of rtPS
set DCmin = DC.rtPS
set DCmax = 2DC.rtPS
set DCdynamic = DC.rtPS + (QLcurrent – QLthreshold1 / QLthreshold2 –
QLthreshold1) * DC.rtPS
If(queue type is rtPS)
        While (0 <= QLcurrent <= QLthreshold1 )
                set DC.rtPS = DCmin
        If (QLthreshold1 < QLcurrent < QLthreshold2 )
                set DC.rtPS = DCdynamic
        If (QLthreshold2 <= QLcurrent <= QLmax)
                set DC.rtPS =DCmax

        while(DC.rtPS > 0 and rtPS not empty)
                Send rtPS packets;
        while(DC.nrtPS > 0 and nrtPS not empty)
                Send nrtPS packets;
        while(DC.BE > 0 and BE not empty)
                Send BE packets;
        adjust all DC counters
        next scheduling round
```

Figure 13: RED-based DFPQ algorithm

In the RED-based DFPQ the DCs are used to decide on how many packets to be transmitted in each frame. It transmits rtPS packets and then transmits nrtPS packets, and finally the BE packets. This technique uses the number of packets in the rtPS queue of an SS, which is not suitable to be used in a BS uplink scheduler. That is because in reality packets will vary in size from application to another. And in other cases where rtPS queue is full because of heavy loaded network it is difficult to just judge of number of packets without considering the latency of these packets. This technique focuses on rtPS only and causing starvation in other data traffic types. They used three SSs only to send data which is not enough to measure a heavy loaded environment. Moreover, the number of applications they used are not enough to judge that their scheduling algorithm is suitable.

The mentioned scheduling algorithms do not solve the starvation of nrtPS and BE data traffic. They only focused in their algorithms on rtPS since it is an important data traffic. Another problem in their work is the low number of applications used. Moreover, they also had been tested their work in very limited scenarios which is most of the time was only one scenario. This raises many questions such as; what will happen if there is no BE traffic is sent or a heavy loaded traffic of BE is sent? What if data rates for all traffic types are sent equally? Because of that a scheduling algorithm is proposed in this research to solve the starvation of nrtPS and BE data traffic without affecting the rtPS traffic while taking into consideration all scenarios that could take place in the real life.

CHAPTER 4: EXTENDED ADAPTIVE DEFICIT PRIORITY QUEUE (EADPQ)

4 Extended Adaptive Deficit Priority Queue (EADPQ)

This chapter discusses the proposed algorithm and the simulation environment used in this thesis. A scheduling algorithm named Extended Adaptive Deficit Priority Queue (EADPQ) is proposed which is an extended work from ADPQ (Chia-Yu et al., 2008). This algorithm is divided into two parts to enhance QoS in WiMAX networks. The first part of the proposed idea focuses on polling BE and nrtPS data traffic with some restrictions, while keeping the rtPS polling as in the default (polled every frame). The second part focuses on distributing bandwidth dynamically and fairly between rtPS, nrtPS and BE data traffic.

4.1 Polling in EADPQ:

This section is divided into two parts, the BE and nrtPS polling. Many experiments are held to select the best percentages and numbers in EADPQ polling. The first subsection will be on polling BE.

4.1.1 Polling BE:

In this method, the BS polls BE stations to let them send their bandwidth requests instead of using contention based method. However, this method could waste the bandwidth if it is used without restrictions. Therefore, two restrictions are used for polling BE traffic after checking that BE queue is active as shown in Figure 14:

1- Check the last bandwidth requested (lastBwRequested) of each BE service flow (sFlow). If it is greater than or equal to 39% of the available bandwidth (bandwidthinbits), then give it a unicast poll (numUcastPolls) because BE has a queued data from the last request which has not been served yet.

2- If the last request was less than 39% , it means that the BE queue is either empty or has very little data. Then, in this case 80 milliseconds has to be waited before polling BE traffic. And the waited time is being known by checking the time difference (timeDiff) between the last allocation time (lastAllocTime) and the current time (currentTime).

```
While (sFlow = BE && sFlowQueue != null)
{
        timeDiff = currentTime - lastAllocTime
        if (lastBwRequested >= (bandwidthinbits * 0.39 ) or timeDiff >= 80 *
                MILLI_SECOND
        {
            numUcastPolls ++
            lastAllocTime = currentTime
        }
          sFlow = next sFlow
}
```

Figure 14: Proposed EADPQ BE polling

4.1.2 Polling nrtPS:

Some restrictions are specified for polling nrtPS traffic by checking that nrtPS queue is active and by executing the code shown in Figure 15:

1- Check the last bandwidth request of nrtPS. If it is greater than or equal to 55% of the bandwidth available, then give it a poll because nrtPS has a queued data from the last request which has not been served yet.

2- If the last request was less than 55%, it means that nrtPS queue is either empty or has very little data. In this case 40 milliseconds must be waited before polling nrtPS traffic.

```
While (sFlow = nrtPS && sFlowQueue != null)
{
        timeDiff = currentTime - lastAllocTime
        if (lastBwRequested >= (bandwidthinbits * 0.55 ) or timeDiff >= 40 *
                MILLI_SECOND
        {
              numUcastPolls ++
              lastAllocTime = currentTime
        }
           sFlow = next sFlow
}
```

Figure 15: Proposed EADPQ nrtPS polling

For rtPS, it is left as the default which is, if rtPS traffic queue is active and has data, it is polled in every frame.

4.2 Scheduling algorithm in EADPQ:

The second part of the proposed work concentrates on distributing bandwidth fairly to all three data traffic studied without causing starvation. It distributes bandwidth fairly and accurately depending on latency and size of bandwidth requested for rtPS and depending on size of bandwidth requested for both nrtPS and BE. Because previous researches focused on improving rtPS traffic, they cause starvation in nrtPS and BE traffic. Thus, the proposed algorithm distributes the bandwidth fairly between rtPS, nrtPS and BE without causing starvation to any data traffic.

These are the abbreviations used in the proposed scheduling algorithm. Tcur is the current time when checking any queue. Talloc is the last time that the BS allocated bandwidth to this data traffic. Tlatency is the maximum time a packet can remain in the queue. Breq is the bandwidth requested by data traffic. Bavail is the available bandwidth in BS that is allowable for an SS to take. 'temp' is a temporary variable. Finally, rtPSQ, nrtPSQ, and BEQ are the quantums allocated to rtPS, nrtPS, and BE data traffic respectively. Figures 16,17,18 illustrate the algorithms of allocating bandwidth for rtPS, nrtPS and BE. Many experiments are held to select the best percentages in EADPQ.

4.2.1 Distributing bandwidth for rtPS:

First, the algorithm of rtPS checks the latency of the packets to be sent and judge as shown in Figure 16:

- If the data reached its allowed latency (urgent), i.e. it will be expired if waited to next frame, a quantum is given to rtPS SS and this quantum is allowed to up to 62% of the available bandwidth in order to send its data. But, in one case which is when the rtPS requested bandwidth is greater than or equal to two times the available bandwidth, an up to 75% of the available bandwidth is given to rtPS SS to send its data.
- If the data has not reached its allowed latency, three cases will be taken in consideration:

  1- If the bandwidth requested is greater than or equal to two times the available bandwidth, a 40% of the available bandwidth is allocated to the rtPS SS.

2- If the bandwidth requested is greater than or equal to 55% of the available bandwidth, a 36% of the available bandwidth is allocated to the rtPS SS.

3- Otherwise, a 33%, i.e. one third, of the available bandwidth is allocated to rtPS SS, so that other data traffic will be allowed to get a chance to send their data if they are starving.

```
If(queue type is rtPS)
{
        If(Tcur - Talloc) >= Tlatency
        {
                temp = Bavail * 0.43
                If(Breq <= temp)
                        rtPSQ = Breq
                else
                {
                        If(Breq >= Bavail * 2)
                                temp = Bavail * 0.75
                        else
                                temp = Bavail * 0.62
                        If(Breq <= temp)
                                rtPSQ = Breq
                        else
                                rtPSQ = temp
                }
        }
        else
        {
                If(Breq >= Bavail * 2)
                        temp = Bavail * 0.40
                else
                {
                        If(Breq >= Bavail * 55)
                                temp = Bavail * 0.36
                        else
                                temp = Bavail * 0.33
                }
                If(Breq <= temp)
                        rtPSQ = Breq
                else
                        rtPSQ = temp
        }
}
```

Figure 16: Distributing bandwidth for rtPS in EADPQ

4.2.2 Distributing bandwidth for nrtPS:

Secondly, nrtPS queue will be checked. In this situation, three cases will be taken into consideration as shown in Figure 17:

1- If the bandwidth requested is greater than or equal to 65% of the available bandwidth, 46% of the available bandwidth is allocated to the nrtPS SS.

2- If the bandwidth requested is greater than or equal to 55% of the available bandwidth, 40% of the available bandwidth is allocated to the nrtPS SS.

3- Otherwise, 36% of the available bandwidth is allocated to nrtPS SS which is a fair quantum for this data type according to its priority, so that other data types are allowed to send their data.

```
If(queue type is nrtPS)
        If(Breq >= Bavail * 0.65)
                temp = Bavail * 0.46

        else
        {
                If(Breq >= Bavail * 0.55)
                        temp = Bavail * 0.40
                else
                        temp = Bavail * 0.36
        }
        If(Breq <= temp)
                nrtPSQ = Breq
        else
                nrtPSQ = temp
```

Figure 17: Distributing bandwidth for nrtPS in EADPQ

4.2.3 Distributing bandwidth for BE:

Third, BE queue will be checked. In this situation, three cases will be taken into consideration as shown in Figure 18:

1- If the bandwidth requested is greater than or equal to 55% of the available bandwidth, a 31% of the available bandwidth is allocated to the BE SS.
2- If the bandwidth requested is greater than or equal to 39% of the available bandwidth, a 27% of the available bandwidth is allocated to the BE SS.
3- Otherwise, a 21% of the available bandwidth is allocated to the BE SS which is a fair quantum for this data type traffic according to its low priority.

```
If(queue type is BE)
        If (Breq >= Bavail * 0.55)
                temp = Bavail * 0.31

        else
        {
                If(Breq >= Bavail * 0.39)
                        temp = Bavail * 0.27

                else
                        temp = Bavail * 0.21
        }

        If(Breq <= temp)
                BEQ = Breq

        else
                BEQ = temp
```

Figure 18: Distributing bandwidth for BE in EADPQ

The EADPQ is tested using the network simulation and experiments as shown in the following section.

4.3 The simulation environment and experiments

The experiments are simulated using Qualnet version 5.0.1 (Scalable Network Technologies, 2010). In order to emphasize the importance of the simulation experiments conducted in this thesis, five different scenarios are used. In the first scenario the applications of rtPS, nrtPS, and BE are equally distributed between SSs by applying BE in 10 SSs, nrtPS in 10 SSs and rtPS in 10 SSs. Secondly, 20 BE SSs, 5 nrtPS SSs, and 5 rtPS SSs are used for scenario 2. For scenario 3, 5 BE SSs, 20 nrtPS SSs, and 5 rtPS SSs are used. Then in scenario 4, 5 BE SSs, 5 nrtPS SSs, and 20 rtPS SSs are used. Finally, in scenario 5, 15 nrtPS SSs and 15 rtPS SSs are used.

All simulation experiments are done in a 1000x1000 meter^2 terrain. The network has one BS and 30 SSs. All of these SSs and BS are distributed uniformly in the terrain, i.e., using a uniform random number generator. The network mode that is worked on is PMP. Each configuration was run 20 times, each time with a different random position for the nodes. This way each run produces different results. At the end, the averages of all outputs of the same configuration are taken. Although the original experiments of ADPQ and RED-based DFPQ were executed on 50 seconds of simulation time, this research executes the experiments on five different durations, 20, 40, 60, 80, and 100 seconds. Moreover, to be more confident about the behavior of the proposed scheduling algorithm, some of the experiments are repeated for longer durations that reach 150 seconds and 200 seconds. In all experiments the data rate used for rtPS, nrtPS, and BE are 1 Mbps, 512 Kbps, and 256 Kbps respectively. Table 2 shows a summary of the simulation parameters used in the simulation experiments conducted in this thesis.

Table 2: Simulation parameters summary

Parameter	Value(s)
Simulator	Qualnet version 5.0.1
Number of runs for each configuration	20
Applications	rtPS, nrtPS, BE
Data rate for rtPS	8 Mbps
Data rate for nrtPS	4 Mbps
Data rate for BE	2 Mbps
Latency	600 milliseconds
Transmission power in dBm	60
Simulation time	20, 40, 60, 80, 100 seconds
Terrain size	1000x1000 meter^2
Number of nodes	31
Mobility	Static
Channel bandwidth	20 MHz
Sampling factor	8/7
Scenario 1	10 BE, 10 nrtPS, 10 rtPS
Scenario 2	20 BE, 5 nrtPS, 5 rtPS
Scenario 3	5 BE, 20 nrtPS, 5 rtPS
Scenario 4	5 BE, 5 nrtPS, 20 rtPS
Scenario 5	15 nrtPS, 15 rtPS

CHAPTER 5: SIMULATION RESULTS AND ANALYSIS

5 Simulation results and analysis

In this chapter the results of the simulation experiments are shown with detailed description to all of them. All figures do not show accurate results when using 20 and 40 seconds for the simulation time that is because the SSs need nearly 20 seconds or little more to be initialized to the BS.

- **Scenario 1:**

In this scenario, the applications of rtPS, nrtPS, and BE are equally distributed between SSs by applying each data type of BE, nrtPS and rtPS in 10 SSs.

As seen in Figure 19 the rtPS data traffic's throughput of EADPQ is nearly the same as ADPQ and RED-based DFPQ. That is because no load of rtPS, nrtPS and BE data traffic.

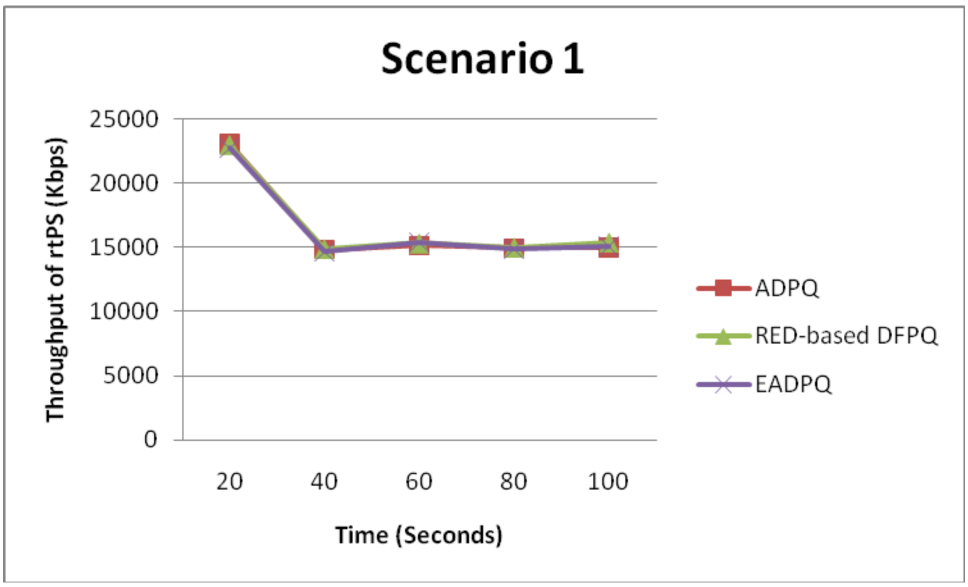

Figure 19: Throughput of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 1

Figure 20 shows that the nrtPS data traffic's throughput of EADPQ is the worst when running the simulation with little time but it becomes better by extending the time until it reaches an enhancement of nearly about 5% than RED-based DFPQ and nearly the same as ADPQ. That is because of keeping the none urgent data of rtPS in their queues and serving more data of nrtPS.

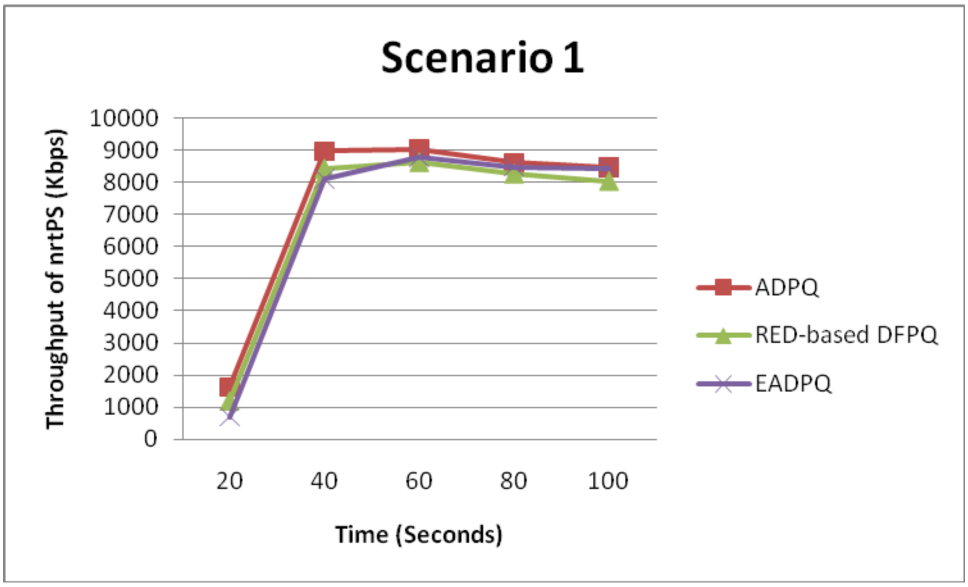

Figure 20: Throughput of nrtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 1

Figure 21 shows that BE in RED-based DFPQ is better than the proposed EADPQ but this enhancement is decreased until it reaches 4%. The reason of the decreasing in RED-based DFPQ is because their technique checks the queue length of rtPS but not the latency. If it does not reach a specific threashold, RED-based serve BE in a specific percentage, this percentage is decreased when the rtPS queue becomes large by time.

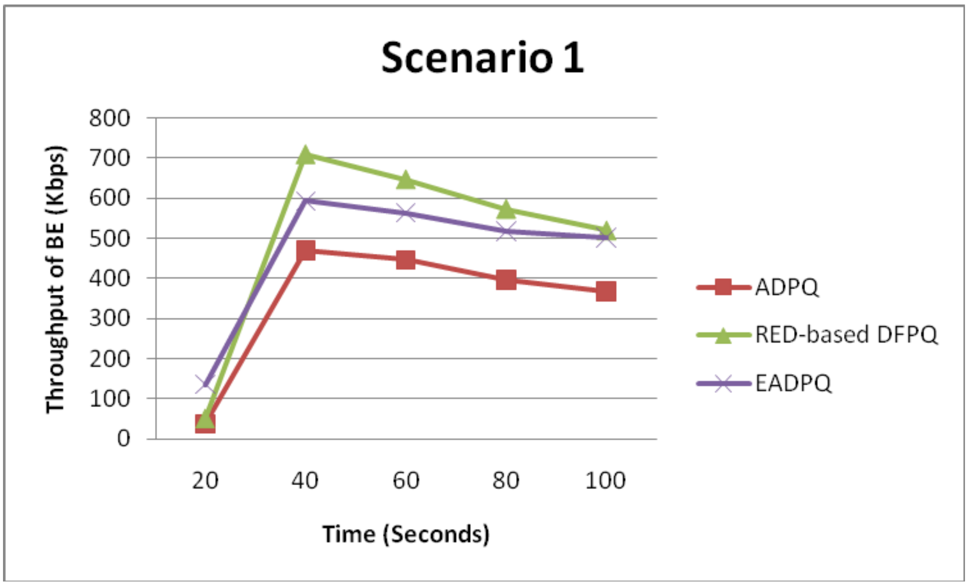

Figure 21: Throughput of BE for ADPQ, RED-based DFPQ and EADPQ in scenario 1

According to the proposed technique, the rtPS traffic in EADPQ is given nearly third of the bandwidth or up to 40% of the bandwidth and kept the remaining traffic in the queue if it is within the allowable latency. In spite of this, the delay of rtPS is nearly the same as ADPQ and RED-based DFPQ as shown in Figure 22.

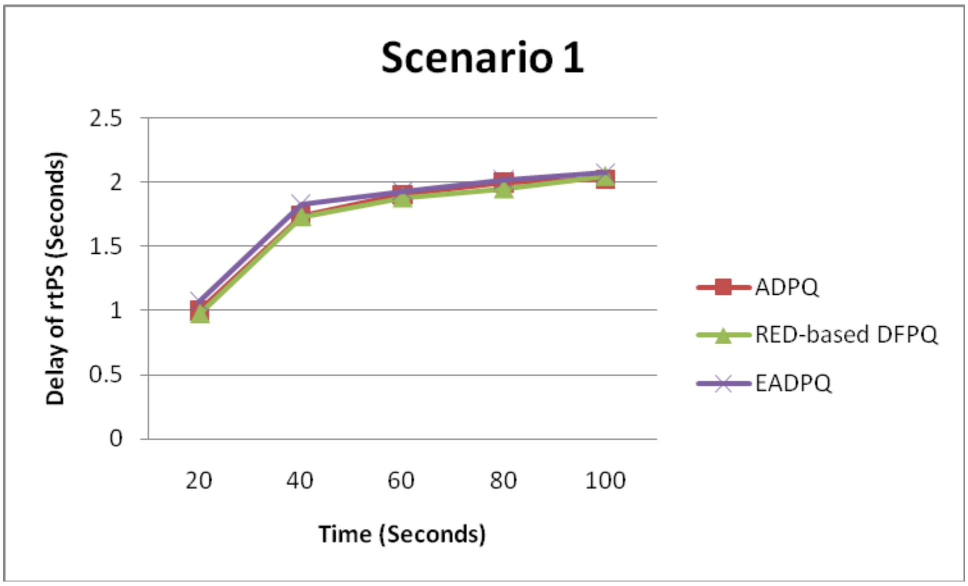

Figure 22: Delay comparison of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 1

Since the proposed algorithm intends to keep the none urgent data in the queue and since the delay in EADPQ is nearly the same as ADPQ and RED-based DFPQ, the data in EADPQ will have slight different in jitter than ADPQ and RED-based DFPQ as shown in Figure 23.

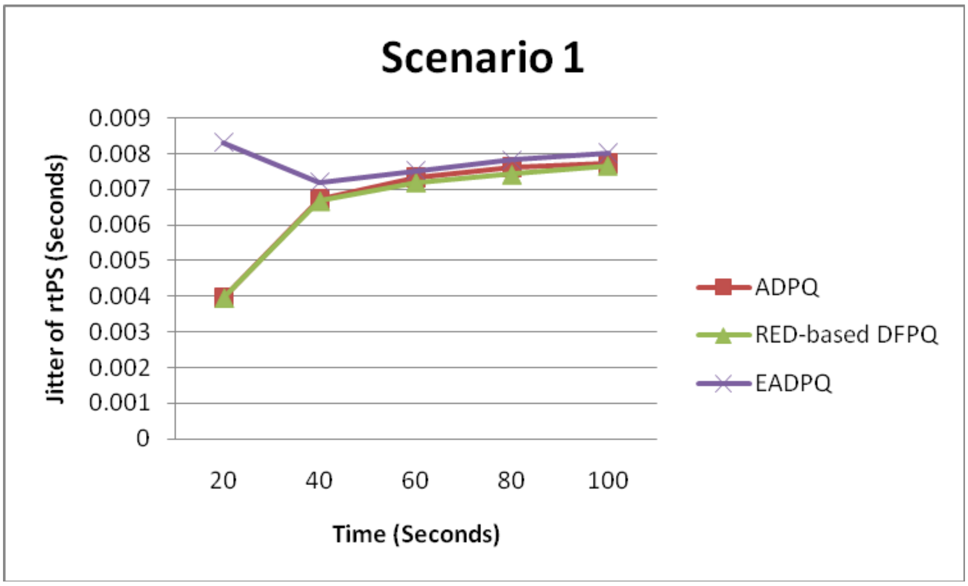

Figure 23: Jitter comparison of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 1

In Figure 24, it is shown that EADPQ has better delay of about 6% than ADPQ and a better delay of about 12% than RED-based when running the simulation for 100 seconds. That is because the proposed algorithm checks the queue size of nrtPS and judge about the needed bandwidth allocated for it. The BS finds some available bandwidth because it serves third of the bandwidth to rtPS in cases of none urgent data and consequently the delay will be reduced.

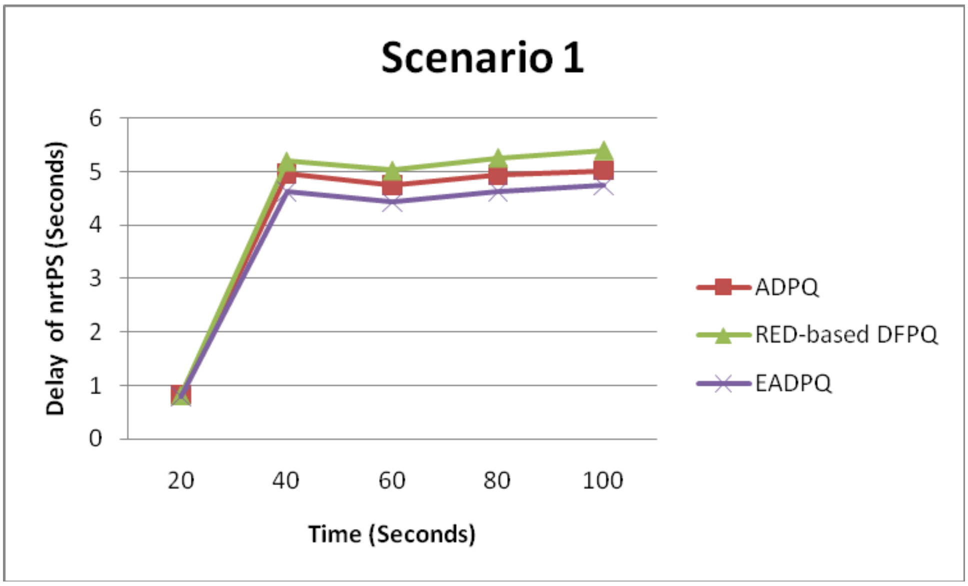

Figure 24: Delay comparison of nrtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 1

As it can be seen in Figure 25, the EADPQ has better delay than ADPQ in up to 6% and worse than RED-based DFPQ in nearly about 1%. That is because in ADPQ they give fixed quantum for BE even if it needs more. On the other hand, RED-based does the same as ADPQ in giving fixed quantum for BE, but here it does not show the shortcoming because in cases of heavy loaded rtPS traffic, RED-based gets portion of the BE bandwidth to rtPS. However, this network does not have loaded rtPS traffic.

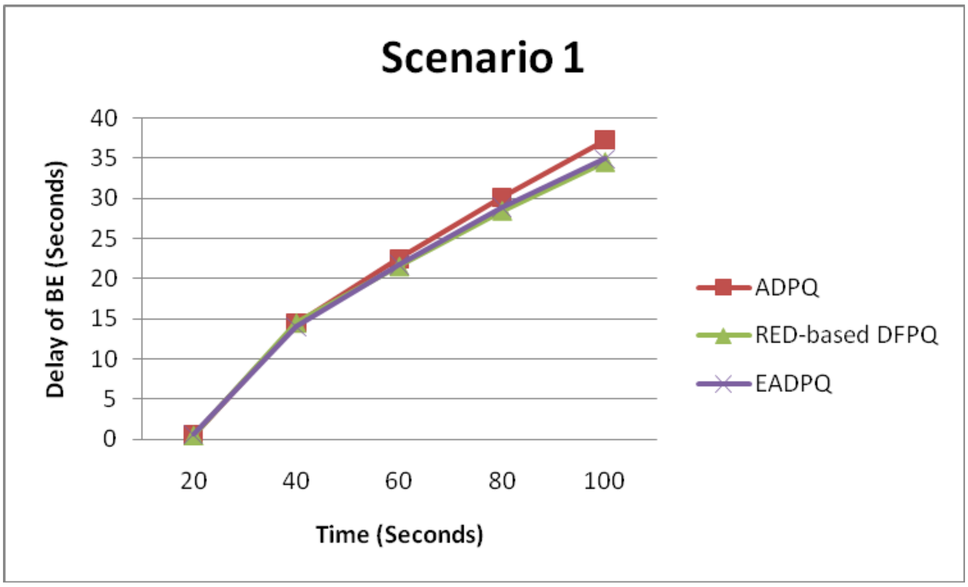

Figure 25: Delay comparison of BE for ADPQ, RED-based DFPQ and EADPQ in scenario 1

- **Scenario 2:**

In this scenario, BE is applied in 20 SSs, nrtPS in 5 SSs and rtPS in 5 SSs.

Although the BE traffic runs on 20 SSs in this scenario, the rtPS traffic in the proposed method, EADPQ, is not affected by that. By increasing the simulation time to 100 seconds, the throughput of EADPQ shown in Figure 26 is increased up to 1% than RED-based and up to 6% than ADPQ. This is because in EADPQ there is a maximum threshold for BE quantum in order not to waste bandwidth of other rtPS traffic.

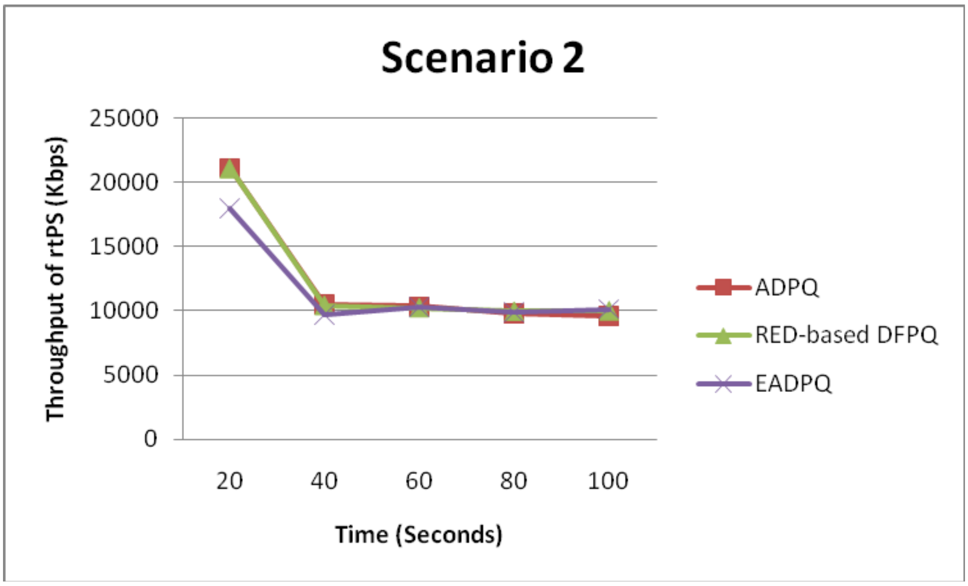

Figure 26: Throughput of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 2

Because the BE traffic is applied on 20 SSs in this scenario, the nrtPS traffic in the proposed method, EADPQ, is affected when the simulation is run for short periods but this gap in throughput is decreased when increasing the simulation time. Consequently, when the simulation time is increased to 100 seconds the EADPQ is performed better than ADPQ and RED-based as shown in Figure 27. It increases by 6%

than RED-based DFPQ and the gap between EADPQ and ADPQ is decreased until it reaches 9%.

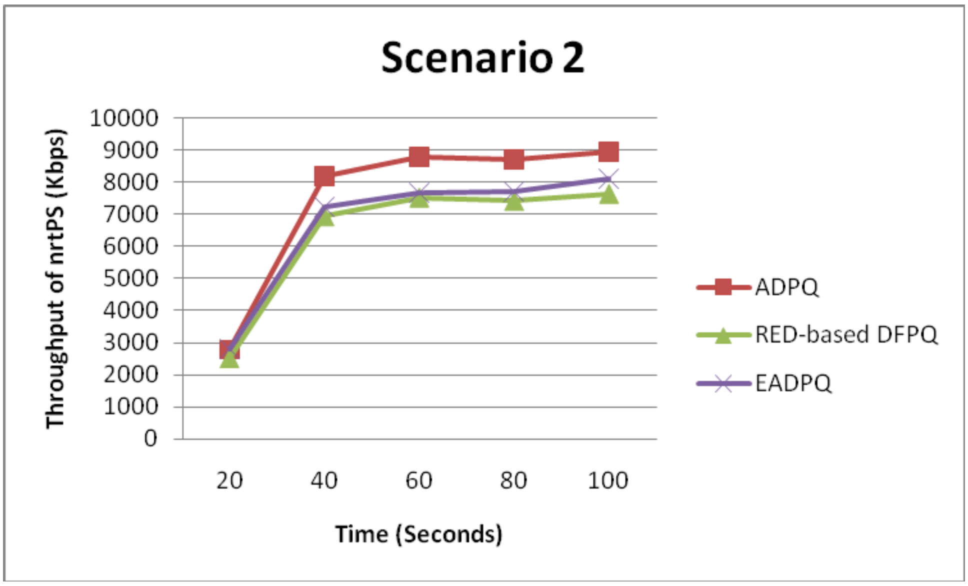

Figure 27: Throughput of nrtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 2

In Figure 28 the BE shows better throughput in EADPQ than RED-based DFPQ by nearly 3% and by 15% than ADPQ. This is because in the proposed method a dynamic quantum allocation is used for BE depending on the requested bandwidth.

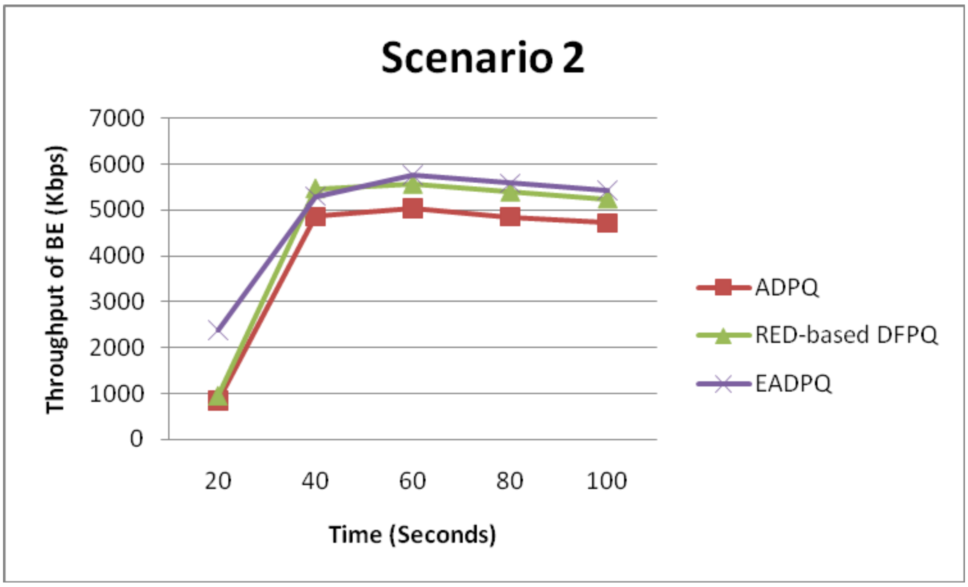

Figure 28: Throughput of BE for ADPQ, RED-based DFPQ and EADPQ in scenario 2

The rtPS traffic in EADPQ is given almost third of the bandwidth or up to 40% of the bandwidth while keeping the remaining traffic in the queue if it is within the allowable latency. In spite of this, the delay of rtPS is nearly the same as ADPQ and RED-based DFPQ as shown in Figure 29. The ADPQ is showing high delay when it reaches 60 seconds because its algorithm gives a fixed quantum to serve rtPS traffic if it reaches its allowable latency. However, if large rtPS data traffic needs to be served, it is given up to two thirds of bandwidth. At that time the data were queued for long time since only 5 SSs are sending rtPS traffic, in addition to the long time needed to reach that specific queue size.

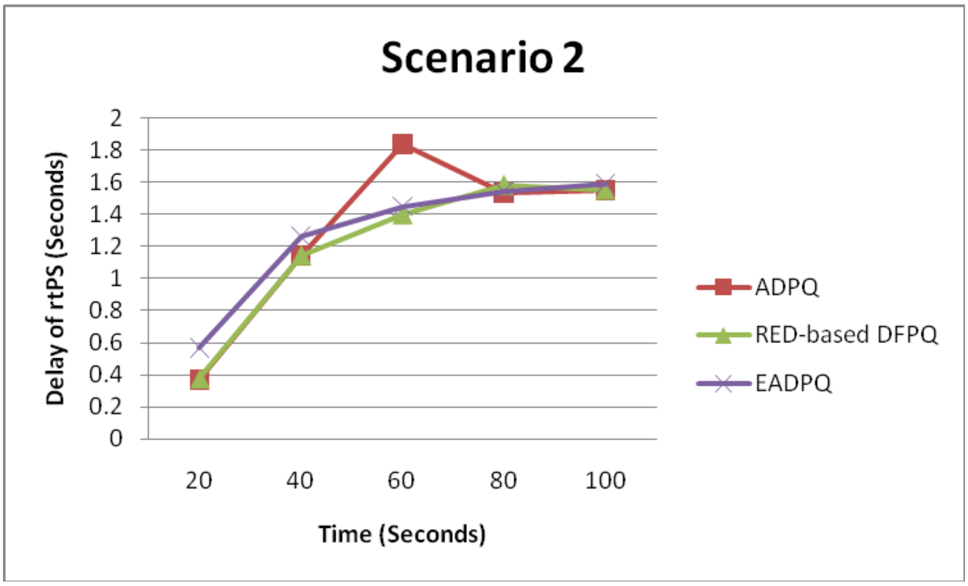

Figure 29: Delay comparison of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 2

Since the proposed algorithm intends to keep the none urgent data in the queue and since the delay in EADPQ is close to ADPQ and RED-based DFPQ delays, the data in EADPQ will have slight different in jitter than ADPQ and RED-based DFPQ as it is shown in Figure 30.

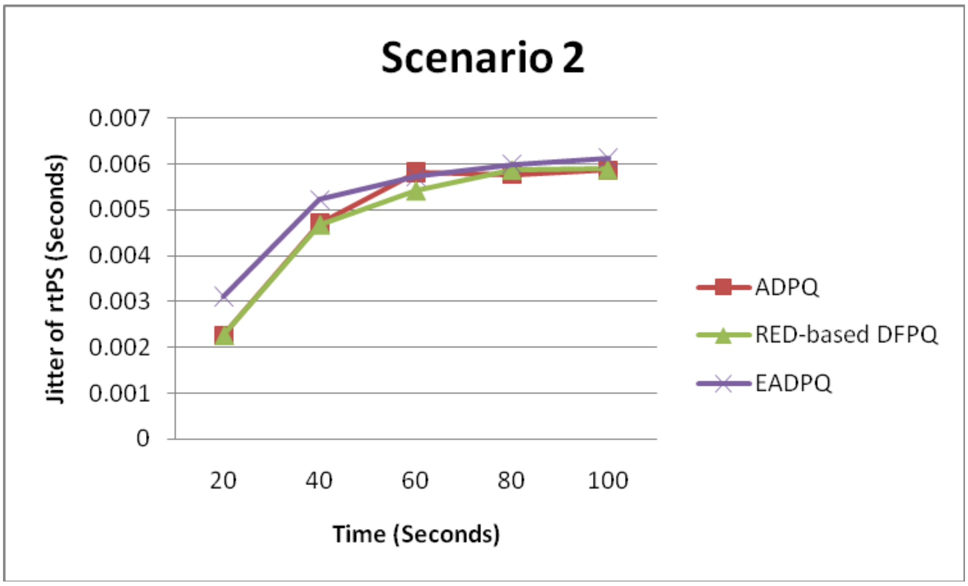

Figure 30: Jitter comparison of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 2

Figure 31 shows that the delay of nrtPS in EADPQ is stable. When the simulation time is run for 100 seconds the EADPQ's nrtPS delay is less than ADPQ in about 14% while it is less than RED-based DFPQ in about 22%. This is because the proposed method tried to dynamically serve BE load of this scenario without affecting the nrtPS traffic.

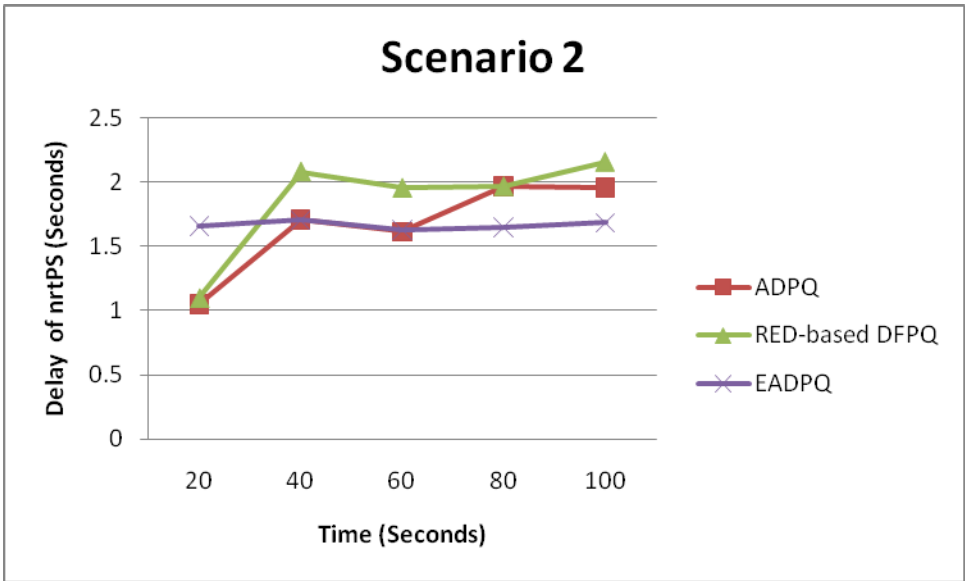

Figure 31: Delay comparison of nrtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 2

Because of the big load of BE in this scenario and because so many packets have to be queued for a long time, the delay is increased in all methods. However, EADPQ has higher delay because some of the served throughput of BE was queued for long time. Figure 32 shows that the EADPQ has a worse delay in about 5% than ADPQ and 14% than RED-based DFPQ.

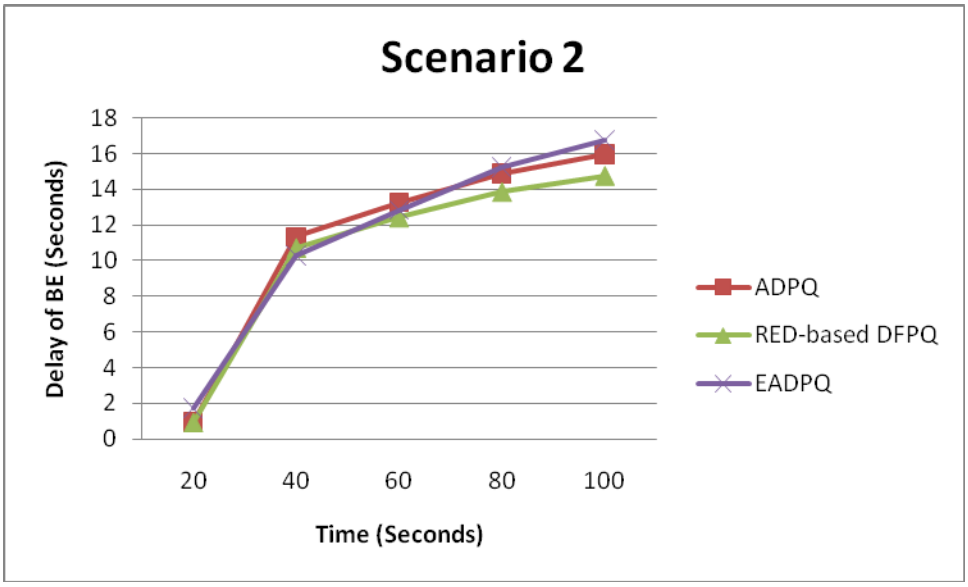

Figure 32: Delay comparison of BE for ADPQ, RED-based DFPQ and EADPQ in scenario 2

- **Scenario 3:**

In this scenario, nrtPS is applied in 20 SSs, rtPS in 5 SSs and BE in 5 SSs.

Although the nrtPS traffic runs on 20 SSs in this scenario, the rtPS traffic in the proposed method, EADPQ, is not affected by that. By increasing the simulation time to 100 seconds, the throughput of EADPQ is increased up to 1% than RED-based and up to 2% than ADPQ as shown in Figure 33.

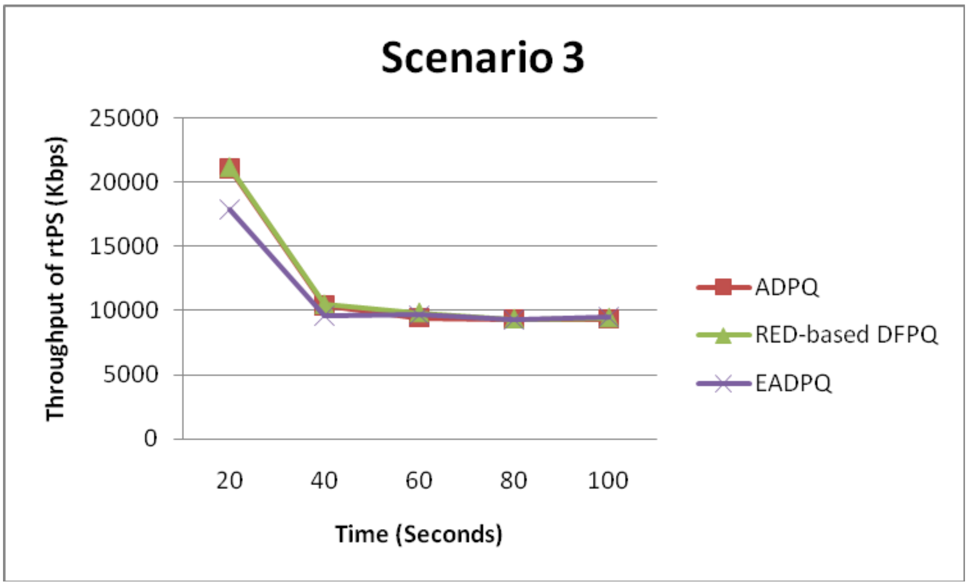

Figure 33: Throughput of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 3

Figure 34 shows that all techniques are nearly having the same throughput in case of nrtPS traffic.

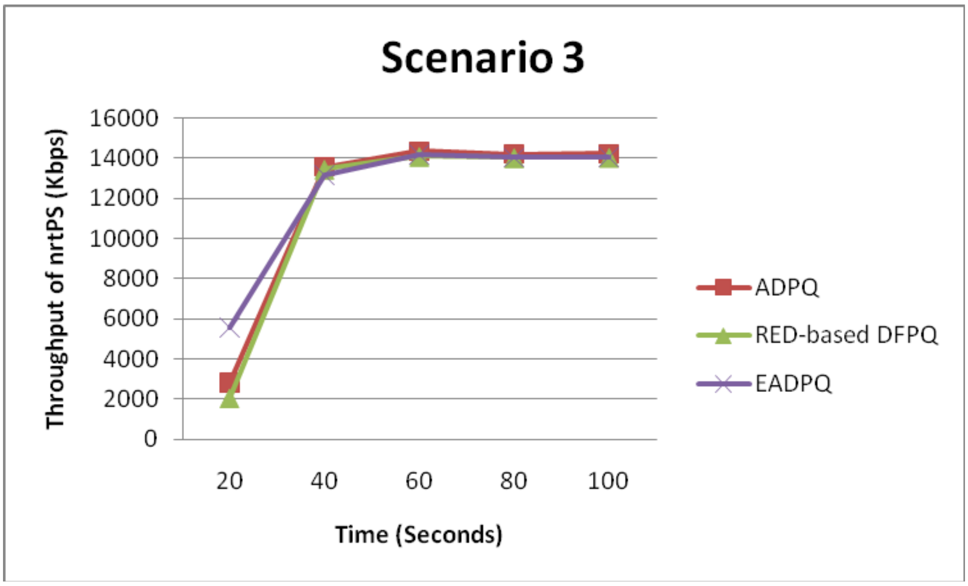

Figure 34: Throughput of nrtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 3

Since 20 SSs in this scenario are running nrtPS traffic and 5 SSs only are running rtPS traffic, EADPQ finds enough bandwidth for rtPS and extra bandwidth for nrtPS and BE. However, in Figure 35 the BE got large quantum of bandwidth at low simulation times because the SSs of BE have been initialized first.

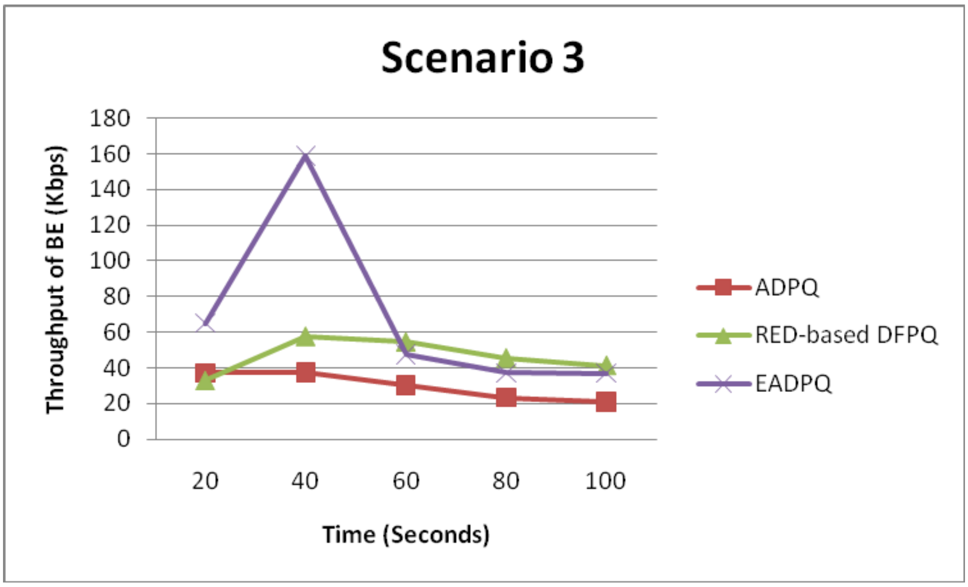

Figure 35: Throughput of BE for ADPQ, RED-based DFPQ and EADPQ in scenario 3

According to the proposed technique, the rtPS traffic in EADPQ is given about third of the bandwidth or up to 40% of the bandwidth while keeping the remaining traffic in the queue if it is within the allowable latency. In spite of this, the delay of rtPS is almost the same as ADPQ and RED-based DFPQ as shown in Figure 36.

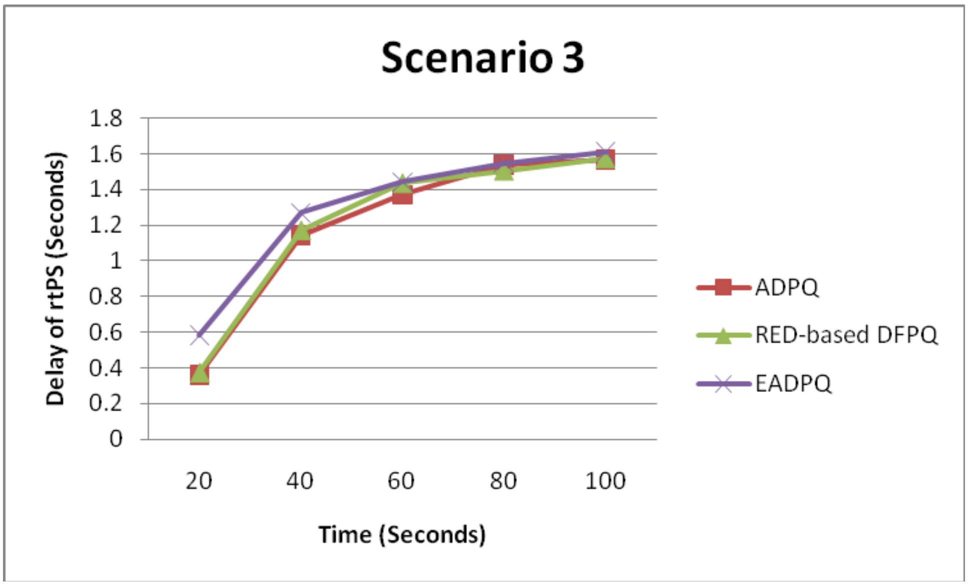

Figure 36: Delay comparison of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 3

Since the proposed algorithm intends to keep the none urgent data in the queue and since the delay in EADPQ is nearly the same as ADPQ and RED-based DFPQ, the data in EADPQ will have slight different in jitter than ADPQ and RED-based DFPQ as shown in Figure 37.

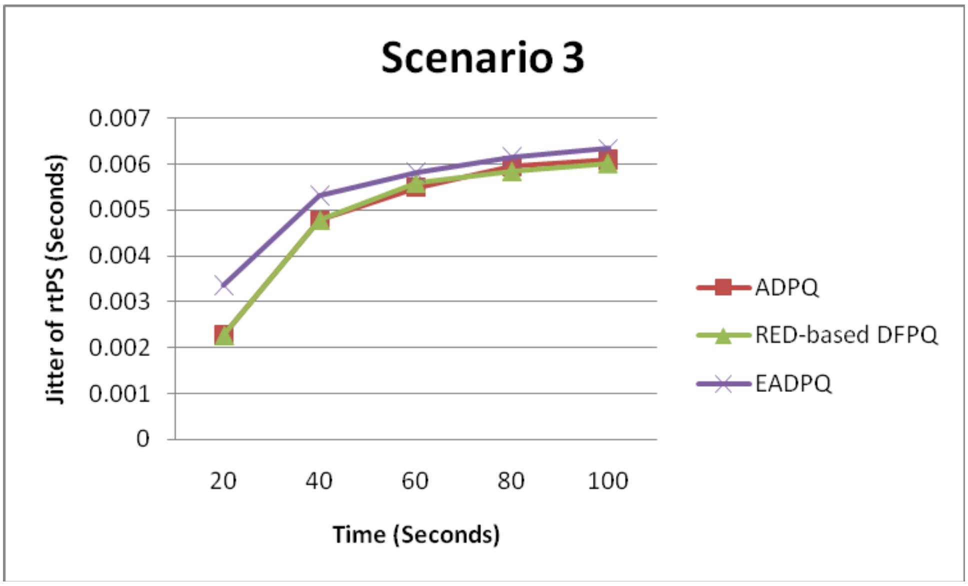

Figure 37: Jitter comparison of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 3

Figure 38 shows that EADPQ has a slight increasing delay than ADPQ and RED-based DFPQ with no more than 2%. That is because the nrtPS packets are queued little time before serving them.

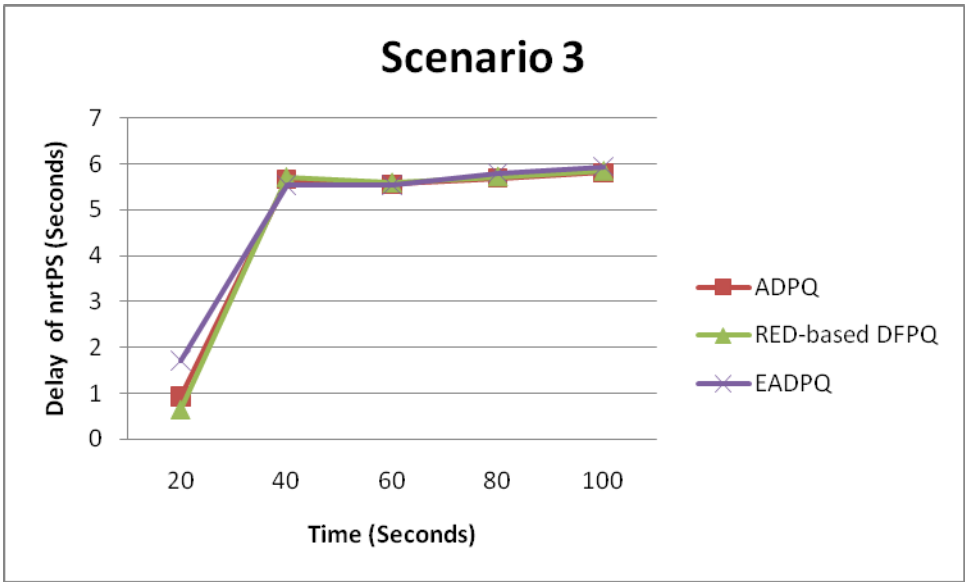

Figure 38: Delay comparison of nrtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 3

When running the simulation for 100 seconds the delay of BE traffic as shown in Figure 39 is very similar in EADPQ and ADPQ while it is higher than RED-based DFPQ in about 7%. That is because the load and priority of nrtPS data traffic are higher than BE and consequently, will take more of BE traffic chances.

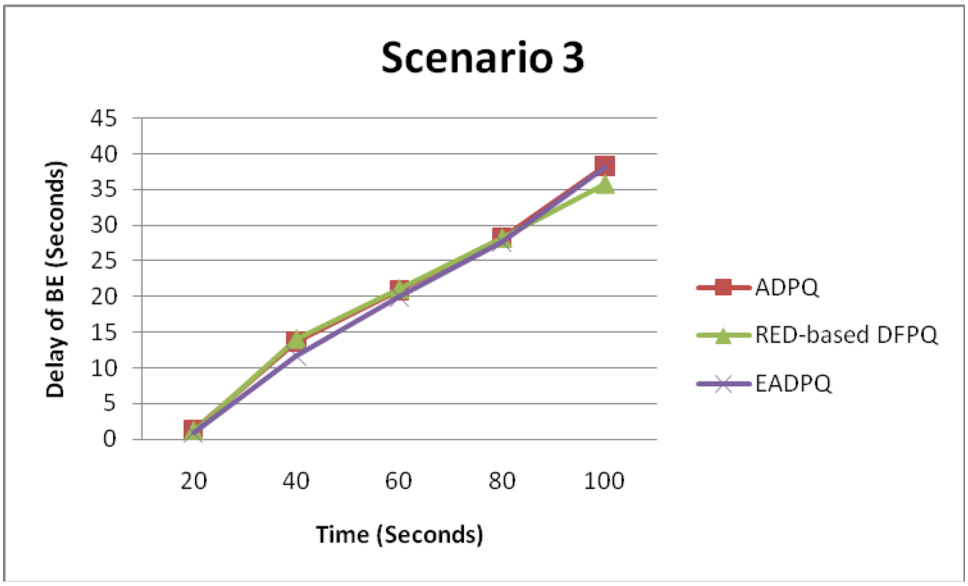

Figure 39: Delay comparison of BE for ADPQ, RED-based DFPQ and EADPQ in scenario 3

- **Scenario 4:**

Now, in this scenario, rtPS traffic is applied in 20 SSs, nrtPS in 5 SSs and BE in 5 SSs.

The ADPQ and RED-based DFPQ have reached optimal throughput for rtPS traffic in their techniques. However, in this scenario the proposed EADPQ has better throughput than RED-based DFPQ in about 1% when running the simulation for 100 seconds and better throughput than ADPQ in about 2% within the same simulation time as shown in Figure 40.

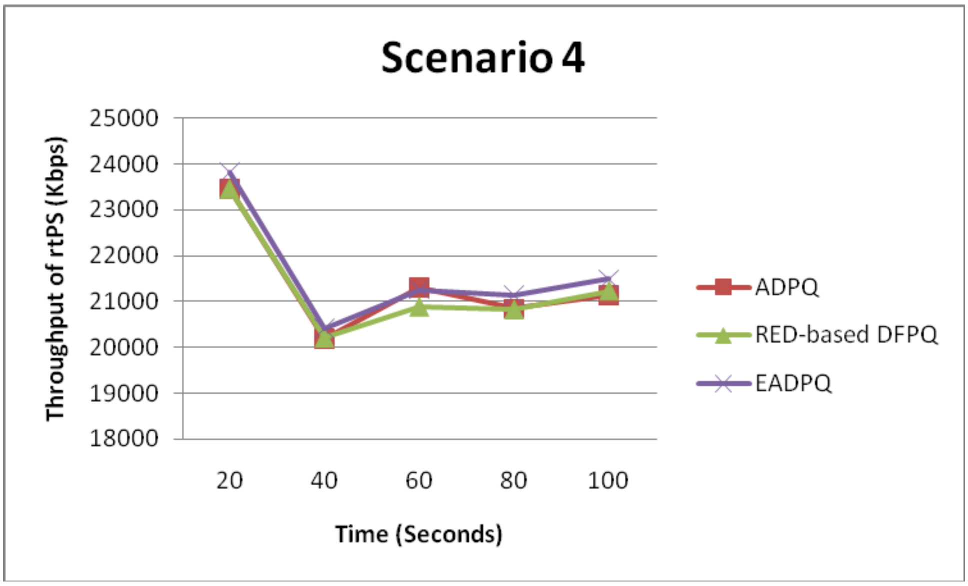

Figure 40: Throughput of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 4

Figure 41 shows a better throughput for nrtPS when using EADPQ. It is better than ADPQ in about 6% when running the simulation for 100 seconds while it is better than RED-based DFPQ in about 13%. This increased throughput of nrtPS in EADPQ is coming from the reduced load of 'bandwidth request' packets and the reduced number of polls.

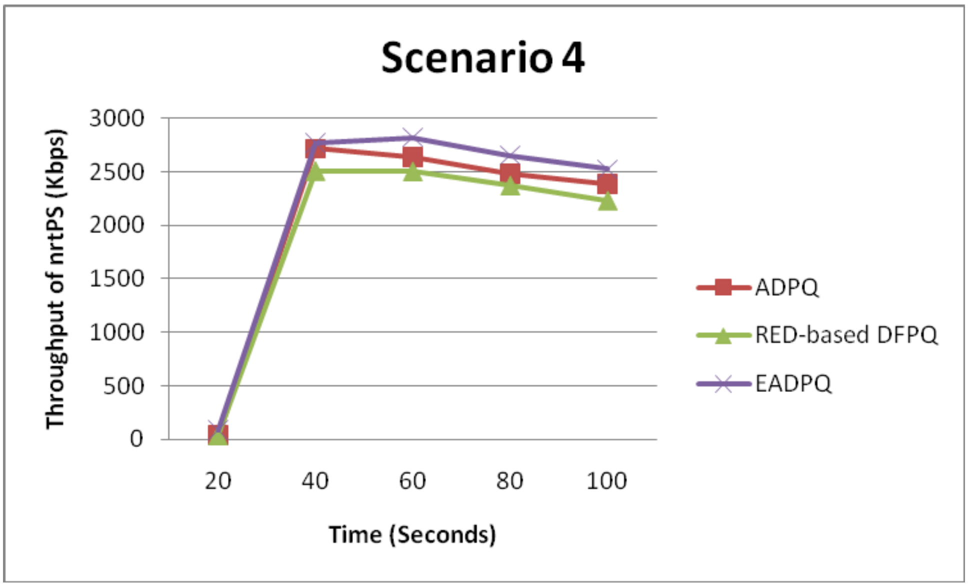

Figure 41: Throughput of nrtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 4

Figure 42 shows a good throughput for BE when using EADPQ. It is better than ADPQ in about 44% when running the simulation for 100 seconds while it is better than RED-based DFPQ in about 36%. This increased throughput of BE in EADPQ is coming from the reduced load of 'bandwidth request' packets and the reduced number of polls.

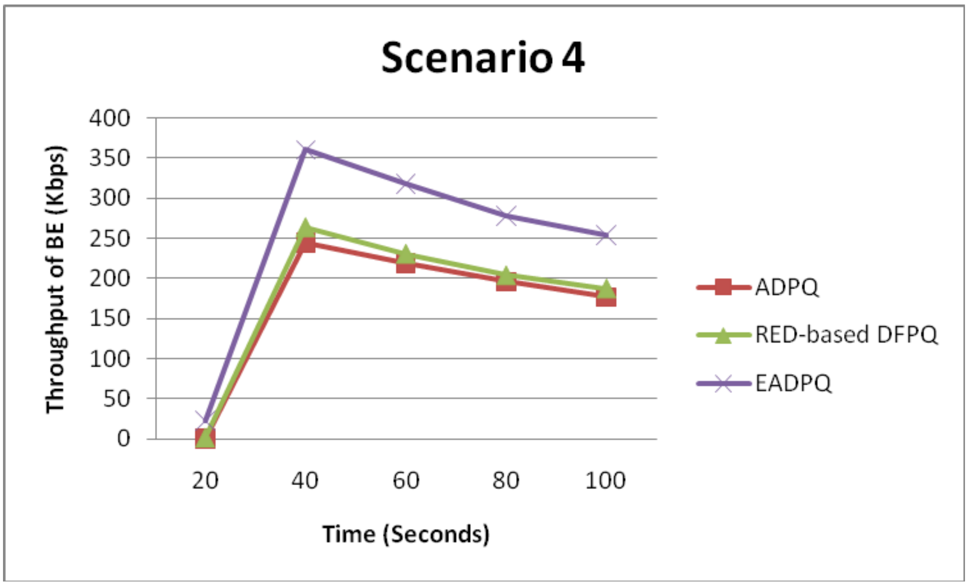

Figure 42: Throughput of BE for ADPQ, RED-based DFPQ and EADPQ in scenario 4

According to the proposed technique, the rtPS traffic in EADPQ is given almost third of the bandwidth or up to 40% of the bandwidth while keeping the remaining traffic in the queue if it is within the allowable latency. In spite of this, the delay of rtPS is nearly the same as ADPQ and RED-based DFPQ as shown in Figure 43.

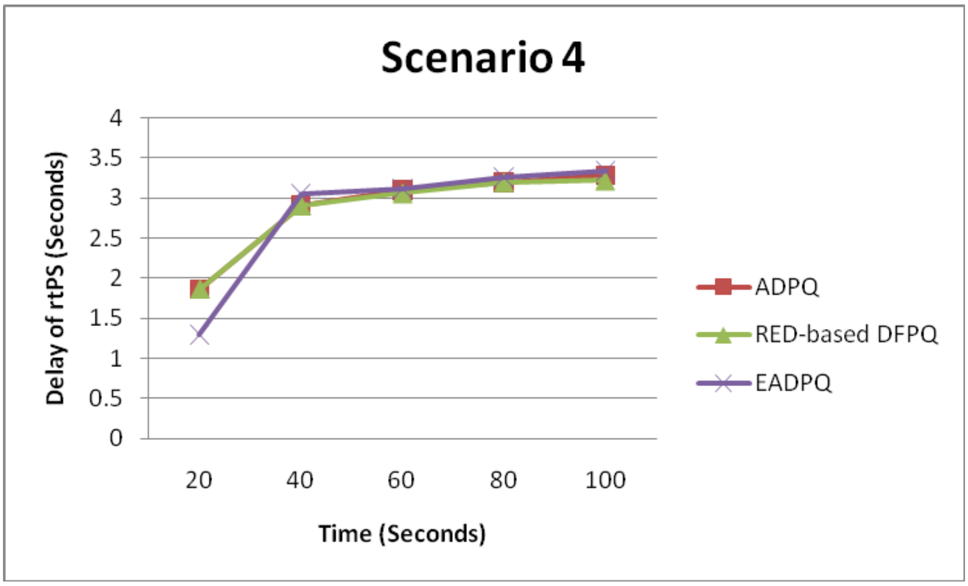

Figure 43: Delay comparison of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 4

The proposed algorithm allocates 62% of the available bandwidth when the data of rtPS reaches the allowed latency time. But when the load of the queue is increased, specifically from the 60 seconds in simulation time, the BS in EADPQ starts to allocate 75% of the available bandwidth to rtPS traffic. Because of that, some data has been queued for long time and accordingly the jitter starts to increase as it is shown in Figure 44.

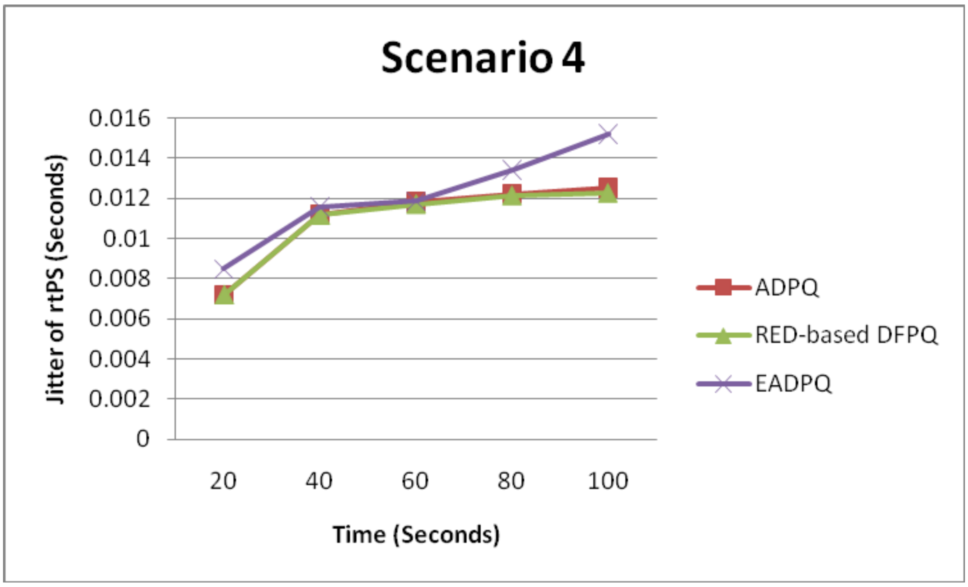

Figure 44: Jitter comparison of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 4

The proposed EADPQ has better delay for nrtPS than ADPQ and RED-based DFPQ for about 5 and 10% respectively, as shown in Figure 45. That is because the nrtPS data in this scenario is not loaded. As a result, it is being served immediately and not needed to be queued for a long time.

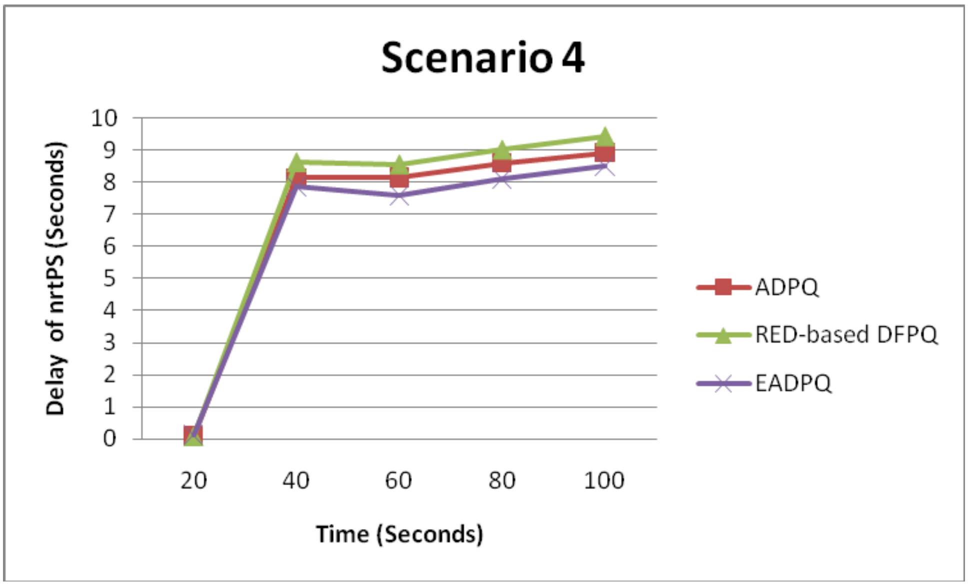

Figure 45: Delay comparison of nrtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 4

The proposed EADPQ has better delay for BE than ADPQ and RED-based DFPQ for about 8% and 4% respectively, as shown in Figure 46. That is because the BE data in this scenario is not loaded. As a result, it is being served immediately and is not needed to be queued for a long time.

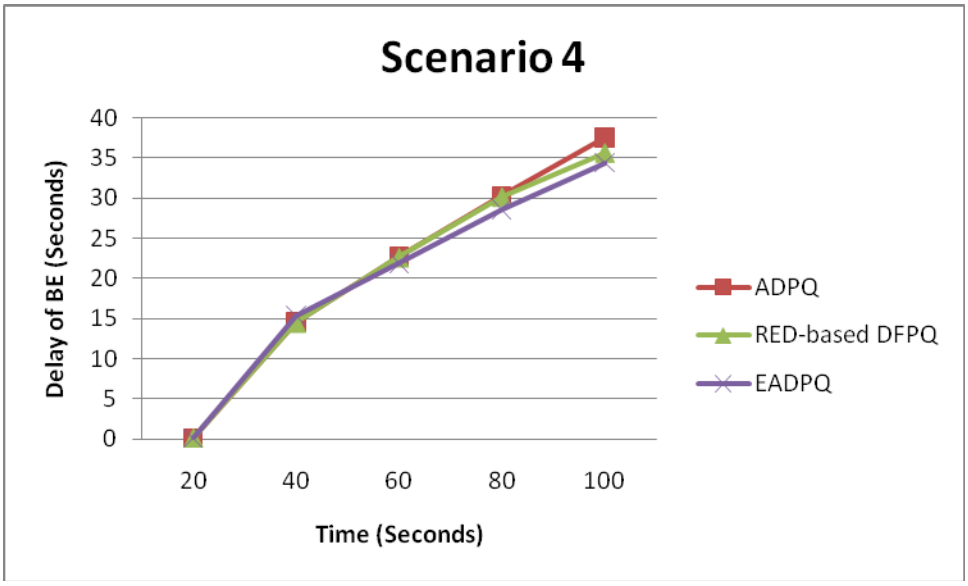

Figure 46: Delay comparison of BE for ADPQ, RED-based DFPQ and EADPQ in scenario 4

- **Scenario 5:**

Finally , no BE traffic is used in this scenario. The rtPS traffic is applied in 15 SSs, and nrtPS traffic in 5 SSs.

Figure 47 shows that EADPQ has nearly the same throughput of ADPQ and RED-based DFPQ since no BE traffic and no loaded rtPS traffic in this scenario.

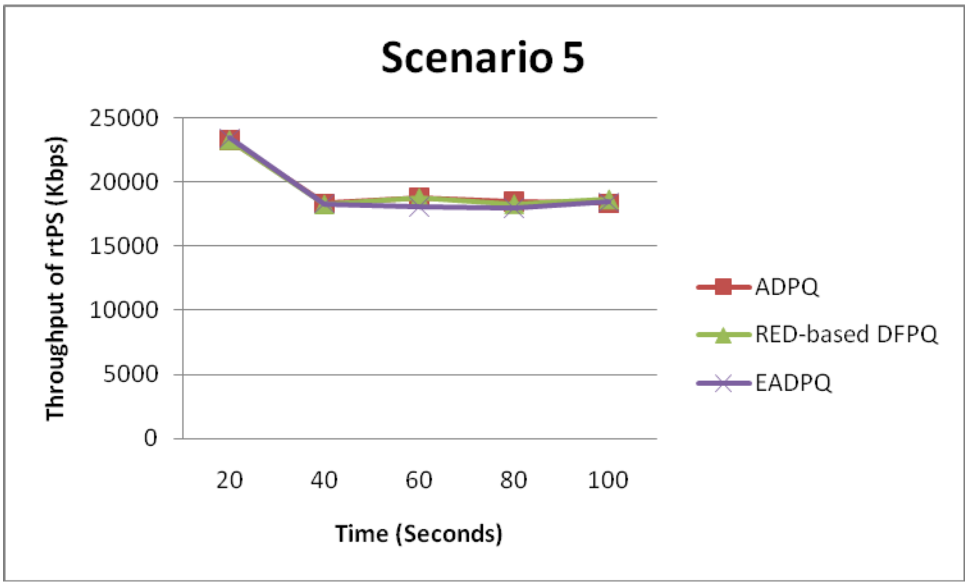

Figure 47: Throughput of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 5

Figure 48 shows that the throughput of nrtPS using EADPQ is better than ADPQ and RED-based DFPQ by almost 8% and 12% , respectively, when running the simulation for 100 seconds. This is because the proposed algorithm is using dynamic quantum to allocate nrtPS traffic depending on the requested bandwidth. Moreover, the EADPQ finds available bandwidth to allocate nrtPS since no BE traffic in this scenario.

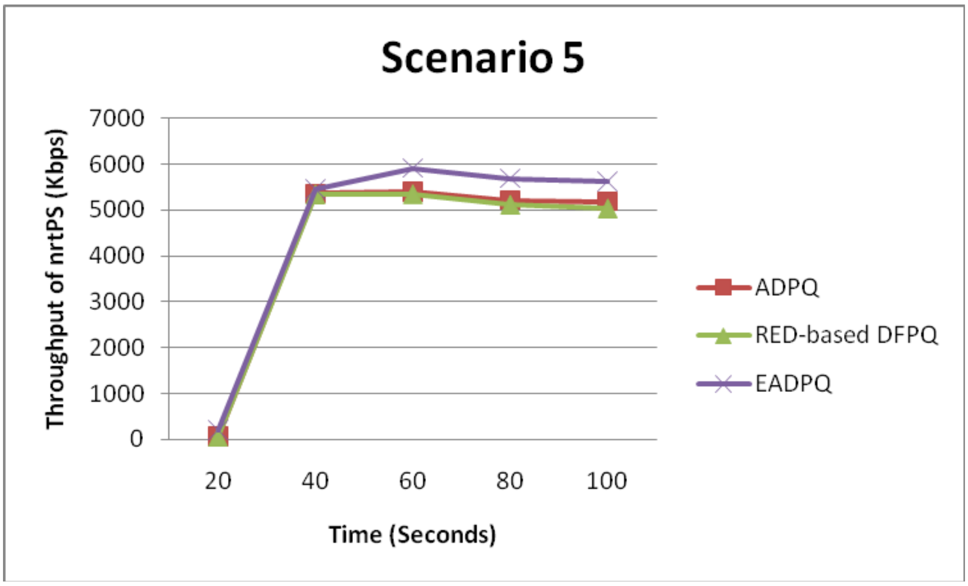

Figure 48: Throughput of nrtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 5

According to the proposed technique, the rtPS traffic in EADPQ is given about third of the bandwidth or up to 40% of the bandwidth while keeping the remaining traffic in the queue if it is within the allowable latency. The delay of rtPS is larger than ADPQ and RED-based DFPQ in nearly about 6% and 5% respectivly as shown in Figure 49. In spite of that, it is still at the beginning of the allowable limit of latency.

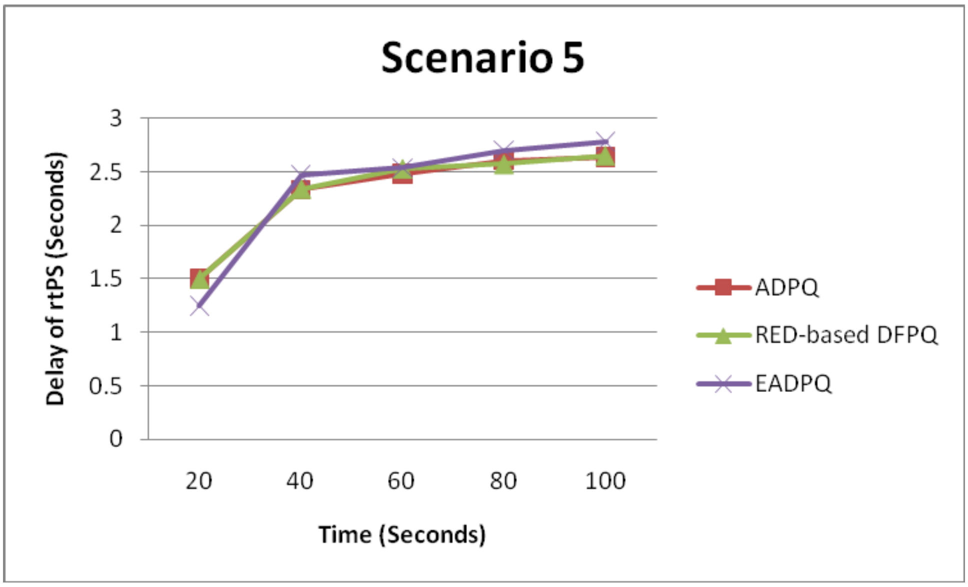

Figure 49: Delay comparison of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 5

The proposed algorithm allocates 62% of the available bandwidth when the data of rtPS reaches the allowed latency time. But when the load of the queue is increased, specifically when running the simulation for 60 seconds the BS starts to allocte 75% of the available bandwidth to rtPS traffic. Because of that, some data has been queued for long time and accordingly the jitter starts to increase as shown in Figure 50.

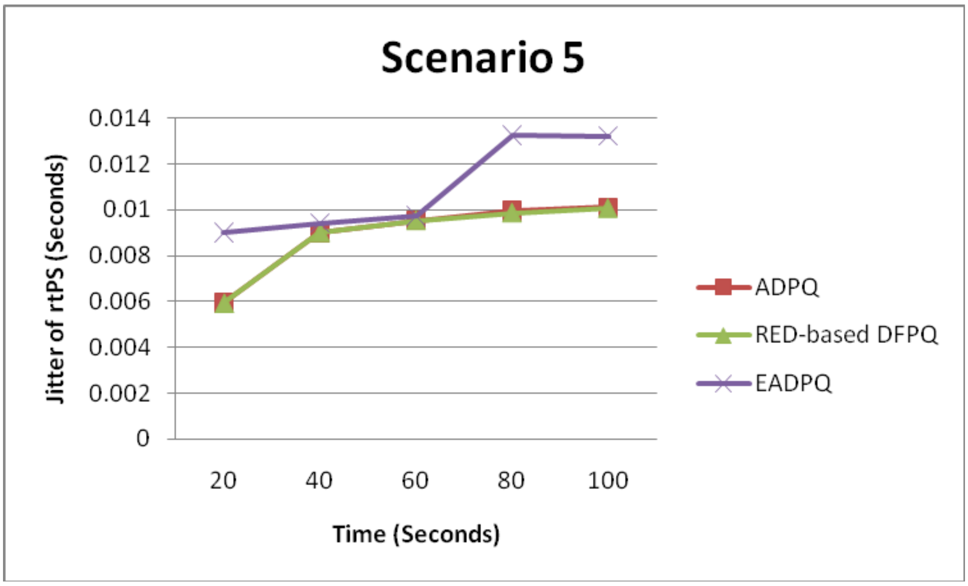

Figure 50: Jitter comparison of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 5

Figure 51 shows that the delay of nrtPS is good when using EADPQ. It is better than ADPQ and RED-based DFPQ in a nearly of 12% and 8% when running the simulation 100 seconds. This is because nrtPS traffic is served immediately since no load in this scenario and not much data of nrtPS traffic. Moreover, because no BE traffic in the network, the time for packets queued is reduced.

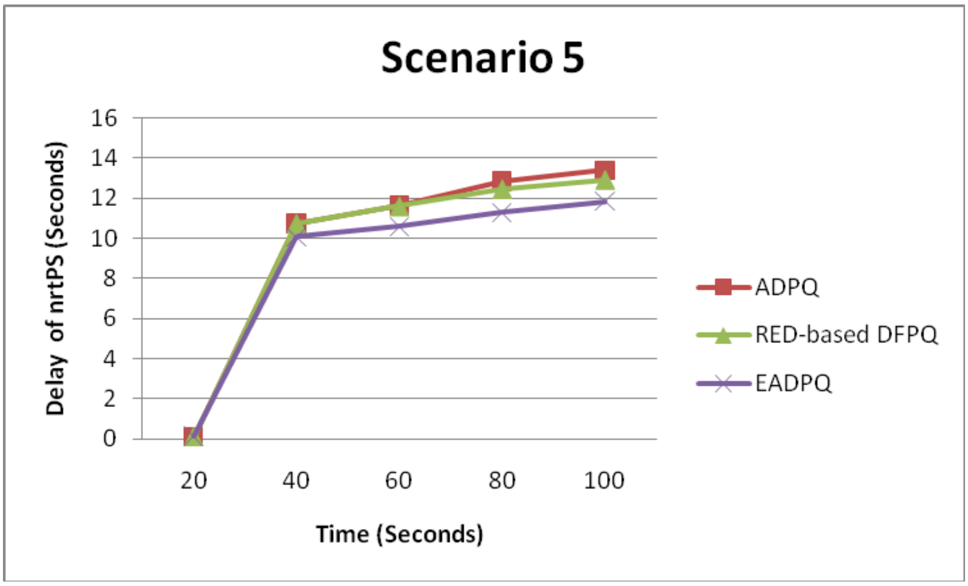

Figure 51: Delay comparison of nrtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 5

- **More experiments at 200 seconds of simulation time:**

To be more confident about some of the scenarios experiments and to make sure that the behavior of the proposed algorithm does not change by increasing simulation time, more experiments have been conducted for some of the scenarios with 200 seconds simulation time.

Figure 52 shows that if the time of the simulation is increased the throughput of EADPQ is becoming better. As the simulation time increases to 200 seconds, the throughput is enhanced in nearly between 3 to 8% than RED-based DFPQ and ADPQ. That is because lots of requests done by BE and nrtPS are reduced by limiting the requests to specific thresholds.

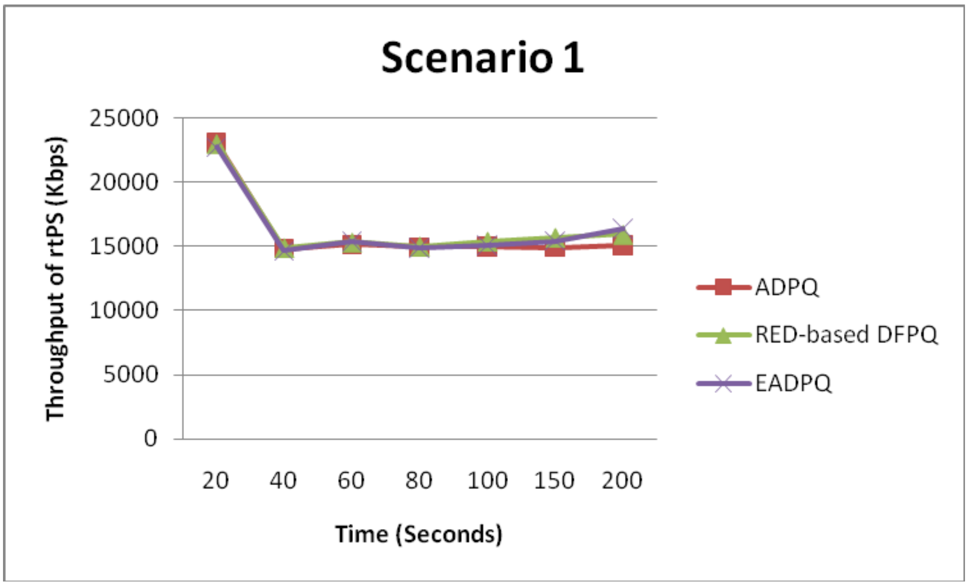

Figure 52: Throughput of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 1

To proof the decreasing of RED-based DFPQ happened in Figure 21, the simulation time is extended in this case to 200 seconds and it is found in Figure 53 that the proposed technique is becoming better than RED-based DFPQ in about 2% and better than ADPQ in about 26%. The reason of the decreasing in RED-based DFPQ is because their technique checks the queue length of rtPS but not the latency. If it does not reach a specific threshold, they serve BE in a specific percentage, this percentage is decreased when the rtPS queue becomes large by time.

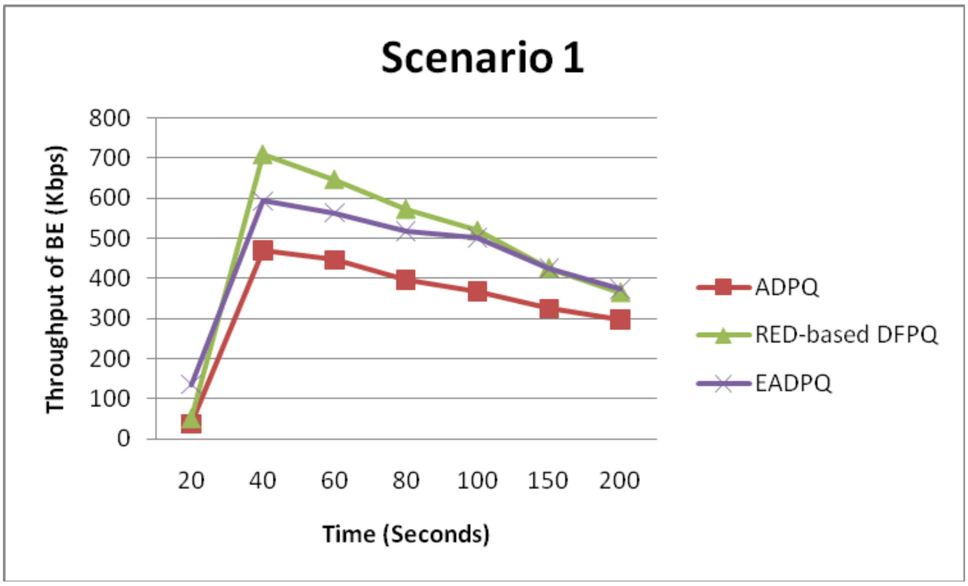

Figure 53: Throughput of BE for ADPQ, RED-based DFPQ and EADPQ in scenario 1

The proposed algorithm allocates 62% of the available bandwidth when the data of rtPS reaches the allowed latency time. But when the load of the queue is increased, specifically from the 60 seconds simulation time, the BS in EADPQ starts to allocate 75% of the available bandwidth to rtPS traffic. Because of that, some data has been queued for long time and accordingly the jitter starts to increase. Moreover, to assure this, the simulation time is increased in this case to 200 seconds as shown in Figure 54 and it is shown that the jitter starts to decrease when it reached 150 seconds which means that the allocated bandwidth started to serve the new incoming data traffic and has nearly finished the old queued packets.

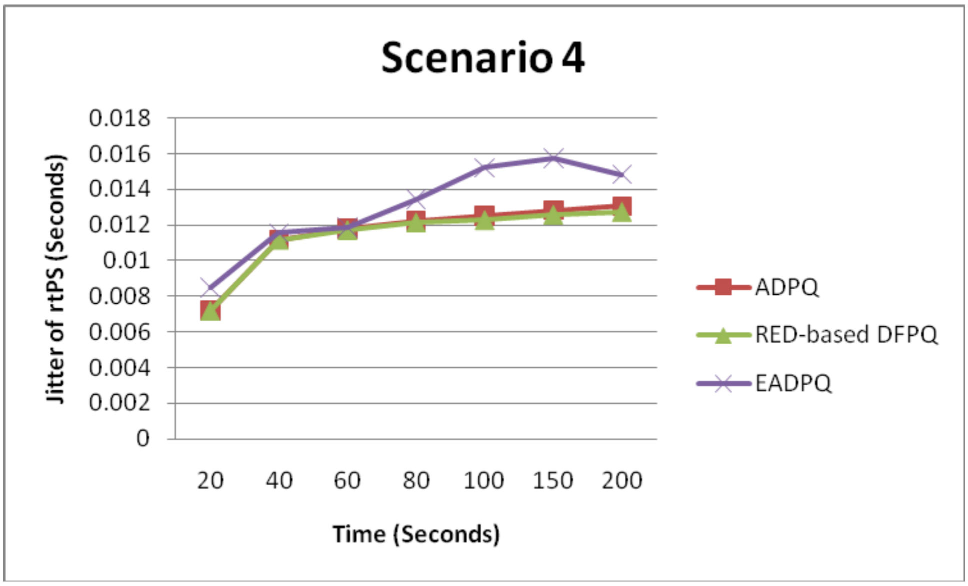

Figure 54: Jitter comparison of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 4

The proposed algorithm allocates 62% of the available bandwidth when the data of rtPS reaches the allowed latency time. But when the load of the queue is increased, specifically when running the simulation for 60 seconds the BS starts to allocte 75% of the available bandwidth to rtPS traffic. Because of that, some data has been queued for long time and accordingly the jitter starts to increase. Moreover, to assure this, the simulation time is increased in this case to 200 seconds as shown in Figure 55 and it is shown that the jitter starts to decrease when it reaches 150 seconds which means that the allocated bandwidth started to serve the new incoming data traffic and has nearly finished the old queued packets.

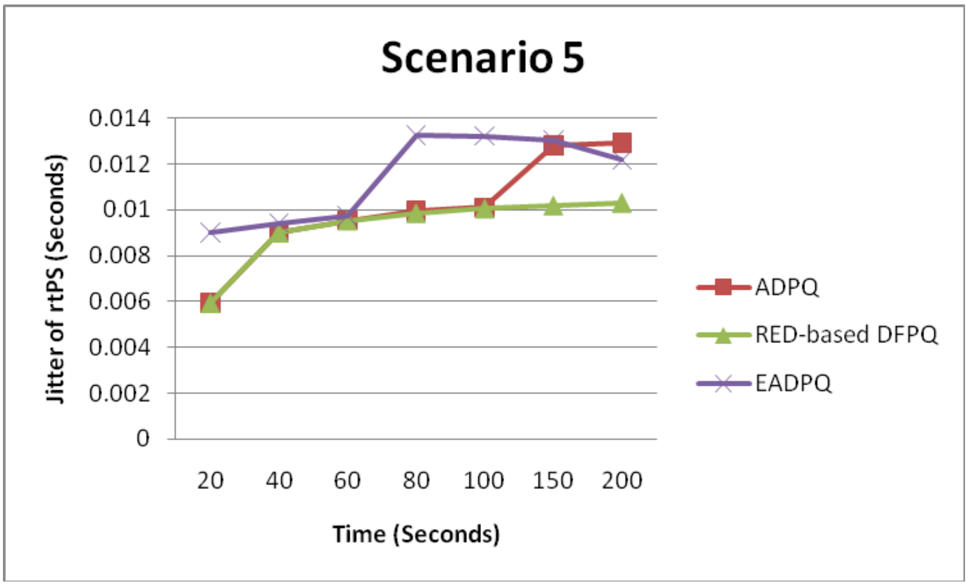

Figure 55: Jitter comparison of rtPS for ADPQ, RED-based DFPQ and EADPQ in scenario 5

CHAPTER 6: CONCLUSIONS AND FUTURE WORK

6 Conclusions and future work

This thesis started by over viewing the WiMAX background. It discussed then some recent researches in the area of scheduling in WiMAX and discuss their shortcomings.

In this thesis, a scheduling algorithm called EADPQ is proposed. The EADPQ technique is divided into two parts, the first one is modifying the bandwidth request method of nrtPS and BE to improve the QoS of WiMAX rtPS, nrtPS and BE data traffic. The second part is distributing the bandwidth fairly between the different data traffic. Five different scenarios with different number of applications are used in each scenario, so that the proposed technique can be examined in different situations. Changing the way that BE requests its bandwidth from contention based to polling based and making the polling for both nrtPS and BE restricted with conditions, have increased the overall bandwidth of the network. This increase is due to the reducing of the number of polls, number of bandwidth request packets and the collisions happened in BE traffic when sending bandwidth requests. As a result, in the first scenario the overall bandwidth of the network(rtPS, nrtPS and BE throughputs) has been increased with nearly 5% when increasing the simulation time but in the second scenario where there exists 20 BE SS, the overall bandwidth have been increased up to 3%. Moreover, the third scenario where 20 nrtPS SSs exists, the bandwidth is nearly the same for all related algorithms. For the fourth scenario the increase was about 3%. Lastly, in the fifth scenario where no BE traffic exists, 2% is increased in the bandwidth.

By applying the five different scenarios it can be concluded that the behavior of all data types have the best results in terms of throughput and delay. For rtPS, the throughput is the same or better than other related scheduling algorithms that are presented in the literature. It reaches its best throughput in the first scenario by reaching an enhanced throughput of 8%. In the other side, the delay of rtPS in all scenarios are also within the allowable latency.

The nrtPS data traffic has also got better throughput or equal to other methods. It reaches in the fourth scenario an enhancement of 13%. Moreover, the delay in the third scenario has a little bit increase in delay for about 2% but all other four scenarios have better results of delay and reached 22% of improvement in the second scenario.

The third data traffic, the BE data traffic, has a better throughput in all scenarios and reached an improvement of up to 70% in the third scenario. However, the delay in the second scenario has increased by no more than 15%. But has a better delay results in all other three scenarios by up to 8% in the fourth scenario.

Finally it is concluded that the proposed scheduling algorithm has made a new enhanced step to rtPS than other methods. Furthermore, EADPQ has ignored the huge starvation of nrtPS and BE data traffic. As a result, EADPQ can be used in real life networks which have different variations of data traffic types and different loads.

As a future work, the proposed scheduling algorithm could be tested on mesh networks. It is also a good work to study the effects of some management packets on the proposed algorithm. Finally, using piggybacking instead of polling will be a good work to study.

REFERENCES

Abu Ali Najah, Dhrona Pratik, and Hassanein Hossam (2009), A performance study of uplink scheduling algorithms in point-to-multipoint WiMAX networks, **Elsevier, Computer Communications**, 32(3): 511-521.

Benefits of WiMAX, **http://nislab.bu.edu/nislab/education/sc441/JustinKen/JustinKen/Networking%20 Webpage/index_files/Page477.htm**, Last accessed: 27 October 2010.

Chen J., Jiao W., and Wang H. (2005), A service flow management strategy for IEEE 802.16 broadband wireless access systems in TDD mode, **IEEE International Conference on Communications (ICC'05)**, Seoul, Korea, VOL. 5, May, 2005, 3422-3426.

Chen Yeong-Sheng, Deng Der-Jiunn, Hsu Yu-Ming, and Wang Sheng-De (2009), An Enhanced Uplink Scheduling Algorithm for Video Traffic Transmission in IEEE 802.16 BWA Systems, **ACM, Wireless network applications**, June, 2009, 1330-1334.

Dhrona Pratik, Abu Ali Najah, and Hassanein Hossam (2008), A Performance Study of Scheduling Algorithms in Point-to-Multipoint WiMAX Networks, **33rd IEEE Conference on Local Computer Networks**, October, 2008, 843 - 850.

Fantacci Romano, Marabissi Dania, and Tarchi Daniele (2009), Adaptive scheduling algorithms for multimedia traffic in wireless OFDMA systems, **Elsevier, Physical communication**, 2(3): 228-234.

Ghazal Sahar, Mokdad Lynda, and Ben-Othman Jalel (2008), Performance Analysis of UGS, rtPS, nrtPS Admission Control in WiMAX Networks. **IEEE International Conference on Communications, 2008. ICC '08.**, 19-23 May, 2008, 2696 – 2701.

Ghosh Debalina, Gupta Ashima, and Mohapatra Prasant (2008), Scheduling in Multihop WiMAX Networks, **ACM SIGMOBILE Mobile Computing and Communications Review**, 12(2): 1-11.

IEEE Standard 802.16-2004 (2004), Part 16: Air Interface for Fixed Broadband Wireless Access Systems. **IEEE Standard for Local and metropolitan area networks**. Technical report, June, 2004.

IEEE Standard 802.16e/D9 (2005), Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems - Amendment for Physical and Medium Access Control Layers for Combined Fixed and Mobile Operation in Licensed Bands. **Draft IEEE Standard for Local and metropolitan area networks**. Technical report, June, 2005.

Katevenis Manolis, Sidiropoulos Stefanos, and Courcoubetis Costas, (1991), Weighted Round-Robin cell Multiplexing in a General-Purpose ATM Switch chip, **IEEE Journal on selected areas in communications**. 9(8): 1265 - 1279.

Kim Seungwoon, Lee Minwook, and Yeom Ikjun (2009), Impact of bandwidth request schemes for Best-Effort traffic in IEEE 802.16 networks, **Elsevier, Computer communications**, 32(2): 235-245.

Lin Jin-Cherng, Chou Chun-Lun, and Liu Cheng-Hsiung (2008), Performance Evaluation For Scheduling Algorithms In WiMAX Network**, 22nd International Conference on Advanced Information Networking and Applications – Workshops, IEEE**, March, 2008, 68-74.

Monteiro João, Sargento Susana, Gomes Álvaro, Fontes Francisco, and Neves Pedro (2009), IEEE 802.16 Packet Scheduling with Traffic Prioritization and Cross-Layer Optimization, **MOBILIGHT 2009. ICST Institute for Computer Sciences, Social-Informatics and Telecommunication Engineering**, 13: 272–281.

Nagaraju Chirayu and Sarkar Mahasweta (2009), A Packet Scheduling To Enhance Quality of Service in IEEE 802.16, **Proceedings of the World Congress on Engineering and Computer Science**, Vol I WCECS 2009, San Francisco, USA, October, 2009.

OFDM Variants 2–11 GHz (2009), 8 December 2009, **http://wimax-made-simple.blogspot.com/2009/12/ofdm-variants-211-ghz.html**, Last accessed: 27 October 2010.

Safa H., Artail H., Karam M., Soudah R., and Khayat S. (2007), New scheduling architecture for IEEE 802.16 wireless metropolitan area network, **Proceedings of the IEEE/ACS International Conference on Computer Systems and Applications (AICCSA'07)**, 2007, 203-210.

Sayenko Alexander, Alanen Olli, and Hamalainen Timo (2008), Scheduling solution for the IEEE 802.16 base station, **Elsevier, Computer Networks: The International Journal of Computer and Telecommunications Networking**, 52(1): 96-115.

Scalable Network Technologies (SNT), 2010, Qualnet version 5.0.1, **http://www.scalable-networks.com/products/qualnet/**, Last accessed: 27 October 2010.

Sekercioglu Y. Ahmet, Ivanovich Milosh, and Yegin Alper (2009), A survey of MAC based QoS implementations for WiMAX networks, **Elsevier, ACM, Computer Networks: The International Journal of Computer and Telecommunications Networking**, 53(14): 2517-2536.

Shreedhar M. and Varghese George (1996), Efficient Fair Queuing Using Deficit Round-Robin**, IEEE/ACM Transactions on networking**, 4(3): 375 – 385.

So-In Chakchai, Jain Raj, and Tamimi Abdel-Karim (2009), Scheduling in IEEE 802.16e Mobile WiMAX Networks: Key Issues and a Survey. **IEEE JOURNAL ON SELECTED AREAS IN COMMUNICATIONS**, 27(2): 156 – 171.

Ting Po-Chun, Yu Chia-Yu, Chilamkurti Naveen, Tung-Hsien Wang, and Shieh Ce-Kuen (2009), A Proposed RED-based Scheduling Scheme for QoS in WiMAX Networks, **IEEE, 4th International Symposium on Wireless Pervasive Computing. ISWPC 2009**, February, 2009, 1 – 5.

WiMAX Forum (2006), Mobile WiMAX – Part I: A Technical Overview and Performance Evaluation, August, 2006, **http://www.wimaxforum.org**, Last accessed: 27 October 2010.

WiMax.com Broadband Solutions, Inc. (2010), Objections to WiMAX, **http://www.wimax.com/education/wimax/objections**, Last accessed: 27 October 2010.

WiMax.com Broadband Solutions, Inc. (2010), Subscriber Stations, **http://www.wimax.com/education/wimax/subscriber_station**, Last accessed: 27 October 2010.

WiMax.com Broadband Solutions, Inc. (2010), WiMAX Antennas, **http://www.wimax.com/education/wimax/antennas**, Last accessed: 27 October 2010.

WiMax.com Broadband Solutions, Inc. (2010), WiMAX radios, **http://www.wimax.com/education/wimax/radios**, Last accessed: 27 October 2010.

WiMax.com Broadband Solutions, Inc. (2010), Wireless architectures, **http://www.wimax.com/education/wimax/wireless_architectures**, Last accessed: 27 October 2010.

Wolnicki Jakub (2005), The IEEE 802.16 WiMAX Broadband Wireless Access; Physical Layer (PHY), Medium Access Control (MAC) layer, Radio Resource Management (RRM), **Seminar on Topics in Communications Engineering**, January, 2005.

Yu Chia-Yu, Zeadally Sherali, Chilamkurti Naveen, and Shieh Ce-Kuen (2008), An Enhanced Uplink Scheduling Scheme for IEEE 802.16 Metropolitan Area Networks, **ACM, The International Conference on Mobile Technology, Applications & Systems 2008 (Mobility Conference)**, Ilan, Taiwan ,September, 2008.

جدولة نقل البيانات في شبكات IEEE 802.16 مع الإهتمام بجودة الخدمة

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ملخص

WiMAX هي واحدة من التكنولوجيا اللاسلكية الحديثة. وتهدف هذه التقنية لدعم جودة عالية لحركة مرور بياناتها المختلفة. وعلى الرغم من وجود فئات مختلفة من أنواع البيانات التي يحددها معيار IEEE 802.16 فإنه ترك الخيار لمزودي هذه التكنولوجيا لتحديد نوعية جدولة البيانات. وبما أن خوارزميات جدولة البيانات الموجودة حاليا لم تحل أزمة بيانات الـ nrtPS و BE فإنه تم إقتراح خوارزمية لجدولة البيانات في هذه الرسالة وتدعى EADPQ وينقسم الـ EADPQ الى قسمين لدعم جودة الخدمة في شبكات الـ WiMAX. فالقسم الأول يبدأ عن طريق إعطاء فرصة لبيانات الـ BE لإرسال حاجتها من المساحة لإرسال بياناتها وبالتالي تجنب الدخول في حالة التنافس لطلب المساحة. بالإضافة الى ذلك، إذا كانت بيانات الـ nrtPS و BE في قائمة الإنتظار لمدة تقل عن مستوى معين، فإن البيانات ستستمر في قائمة الإنتظار حتى وقت محدد. أما الجزء الثاني من العمل المقترح فهو توزيع نطاق المساحة المطلوبة لإرسال البيانات بشكل متغير و عادل بين أنواع البيانات المختلفة. وقد تم عقد مقارنة بين الخوارزمية المقترحة مع غيرها من

خوارزميات الجدولة الحديثة. عن طريق إجراء العديد من السيناريوهات المختلفة تم الخلاص إلى أن الـ EADPQ زادت الإنتاجية الإجمالية للشبكة(rtPS وnrtPS وBE) بنسبة تصل إلى 5%. ورغم أن التركيز كان على بيانات الـ nrtPS و BE فإن ثمة تحسن لإنتاجية بيانات rtPS بنسبة 8%. وعلاوة على ذلك، فإن EADPQ تجنبت الأزمة الضخمة لبيانات الـ nrtPS و BE من خلال زيادة الطاقة الإنتاجية وخفض التأخير. بالنسبة لبيانات الـ nrtPS فقد وصلت إنتاجيتها إلى 13%، بينما تناقص التأخير بنسبة 22%. بالإضافة إلى ذلك تم تحسين إنتاجية بيانات الـ BE بنسبة 70% وانخفض التأخير بنسبة 8%. وأخيرا، تم الإستنتاج بأن EADPQ يمكن استخدامها في الشبكات الحقيقية التي فيها أشكال مخلفة من البيانات والأحمال المختلفة.