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A Study on Uplink Video Transmission in Broadband Wireless Access Systems

조선대학교 대학원

컴퓨터공학과

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이 논문을 공학석사학위신청 논문으로 제출함.

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ABSTRACT

A Study on Uplink Video Transmission in Broadband Wireless Access Systems

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Real time VBR video traffic generated from diverse multimedia applications such as video conferencing, video streaming, VOD and IP broadcasting is expected to be the major portion of traffic in broadband wireless access systems. VBR applications, however, oblige stringent requirements in terms of QoS, which requires an efficient traffic control and sophisticated network management functions.

IEEE 802.16 doesn't allow contention based BW-REQ mechanism for video applications belonging to rtPS. However, we introduced the feasibility of contention BW-REQ by proposing GTAP-BRM. With the results obtained, we believe that this can be a novel alternative for the contention free polling. In this work we made provision for contention based BW-REQ with broadcasted access probability for rtPS by employing *game theory* as a tool to prioritize their access resulting almost similar performance as polling based BW-REQ. This implies that the adoption of this scheme will simply increase the bandwidth efficiency since the bandwidth that could have been wasted for unicasting is saved.

For the VBR video application belonging to rtPS, resource allocation should be done in such a way that their delay obligation should be met with the least possible PLR performance. In any transmission systems packet loss could occur

due to imperfect channel or if they missed their deadlines and are dropped by the system. PLR because of the expiration of the deadline can be reduced significantly if we can save the signaling time that occur in the system to control their transmission. In IEEE 802.16, for polling based signaling for bandwidth request and grant, we proposed a prediction mechanism that can be used for adaptive bandwidth request in which request is made for *future-requirement* up to the grant time, unlike *present- requirement* at request time in the legacy schemes. This yields the immediate benefit of reduction of waiting time by at least one NPI (Nominal Polling Interval) for the packets that arrive in the system between the interval of request time and grant time. Since the allocation is based on the prediction, it should be maintained that prediction is accurate enough. Our proposed prediction model has the accuracy of almost 98 %.

Because of the streaming and conferencing applications, we could expect huge video traffic in the uplink even though the proportion will be comparatively low than the downlink traffic. Hence study on uplink video scheduling is getting wide interests these days. Video is usually encoded before transmission for better compression and to mitigate the error propagation. Because of encoding of original video content, video content will be interpreted in different way than originally it was meant. There could possibly rearrangement of the content with some fraction containing much information while other carries less information. In such condition, in wireless domain it is wise to save part of the encoded video which carries more information if any loss to occur. So we proposed an opportunistic scheduler which saves the important frames in the expense of less important frames.

The proposed schemes, novel signaling mechanism for contention based BW-REQ, improvised scheme to save signaling time in polling based bandwidth *request-grant* mechanism and scheduling scheme with least possible distortion, for video applications in IEEE 802.16 are the remarkable improvements which we believe are the strong basis for the better service provisioning.

ABSTRACT

광 대역 무선 접속 시스템에서의 상향링크 비디오 전송에 대한 연구

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광대역 무선 통신 시스템의 하나인 IEEE 802.16 MAN 에서의 패킷 전송은 연결형 MAC 프로토콜 (connection oriented MAC protocol) 특성에 따라 전송 전에 기지국과 단말 간의 연결 설정이 이루어져야 한다. 이러한 연결 설정을 통해 단말이 전송하고자 하는 상향링크로의 모든 어플리케이션에 대한 QoS 관리, 대역폭 관리 등을 기지국 주도로 수행하게 된다. 모든 단말의 상향링크 패킷 전송을 위한 기지국의 자원 할당은 기본적으로 request-grant 메커니즘과 상향링크 스케줄링 메커니즘에 의해 수행된다. 본 논문에서는 이러한 상향링크 전송 메커니즘을 고려하여 비디오 트래픽과 같은 실시간 트래픽의 효과적인 전송 알고리즘들을 제시하고 그 성능을 분석하였다.

우선, GTAP-BRM (Game Theoretic Access Probability based Bandwidth Request Mechanism) 알고리즘이 WiMAX PMP 네트워크에서 실시간 트래픽의

효율적 전송방식으로 제안되었다. IEEE 802.16 에서 rtPS(Real Time Polling Service) QoS 클래스에 속한 연결은 비 경쟁 방식의 BW-REQ(Bandwidth Request)를 수행하도록 되어있다. 반면에, nrtPS(Non Real Time Polling Service)는 경쟁 기반 방식과 비경쟁 방식의 두 가지 모두를 수행할 수 있도록 되어있다. 제안된 GTAP-BRM 알고리즘에서는 두 개의 클래스 모두 경쟁 기반 방식으로 BW-REQ 를 수행할 수 있도록 하였다. 엄격한 지연 시간 제약을 가지는 실시간 트래픽과 같은 QoS 클래스는 매우 높은 평균 패킷 전송 성공 확률과 매우 낮은 평균 접속 지연 시간을 가져야 한다. 따라서, 제안된 GTAP-BRM 은 경쟁 기반 전송의 장점과 더불어 rtPS 연결에 대해서도 1 프레임 길이의 시간동안 거의 100%의 평균 성공 확률과 낮은 평균 접속 지연 시간을 가지도록 설계되어 이러한 실시간 트래픽의 요구조건을 만족함을 보여 주었다.

두 번째로, polling 기반의 전송 기법에서 실시간 트래픽 예측 메커니즘이 제안되었다. 제안된 알고리즘에서는 FMLP (Feed Forward Multi Layer Perceptron) 기반 시스템의 ANFIS(Adaptive Neuro Fuzzy Inference System) 이 VBR coded MPEG 프레임의 첫 번째 프레임의 길이를 예측하기 위해서 사용되었다. 트래픽 예측 알고리즘을 수행하는데 있어서 프레임의 길이는 chaotic time series 으로 모델링 되었고, step-ahead prediction 이 ANFIS 을 사용하기 위해서 수행되었다. 성능 평가 결과로부터 하나의 MPEG 소스로부터의 training time series 와 다른 MPEG 소스로부터의 training time series 를 위한 예측된 프레임의 RMSE(Root Mean Square Error)는 각각 2.08%과 2.14%가 확인되었다.

예측된 통계치는 WiMAX 의 rtPS 클래스에서 look-ahead 자원 할당을 위하여 사용될 수 있으며 이를 통해 MPEG 프레임을 위한 지연 시간을 줄이고 PLR(Packet Loss Ratio)를 향상 시킬 수 있다.

마지막으로 본 논문에서는 IEEE 802.16 기반의 광 대역 무선 접속 시스템에서 부호화된 MPEG 비디오 트래픽을 위한 상향링크 패킷 스케줄링 알고리즘이 제안되었다. 제안된 스케줄러에서는 GOP 의 개별적인 프레임을 우선순위화 시켰다. 제시된 우선순위화 알고리즘에서는 splitting video flow 을 다른 레이어로 나누고, 스케줄링을 위해서 각각의 프레임에 우선순위를 부여하고, 우선 순위에 따라 선택적으로 packet drop 을 수행한다. MPEG 프레임의 우선순위화 된 inter frame dependency 는 비디오 프레임로부터 분할된 비디오 패킷 전송의 새로운 순서를 만든다. 제안된 스케줄링 알고리즘의 성능평가로부터 I 와 P 와 같은 중요한 프레임은 낮은 PLR 을 갖게 되었고, 반면에 가장 작은 중요성을 가지는 B 프레임은 높은 PLR 을 가짐을 확인하였다.

본 논문에서 수행된 연구결과는 IEEE 802.16 MAN 시스템 뿐만 아니라 기지국 기반의 모든 무선 전송 시스템에서 상향링크 패킷 전송 메커니즘으로 활용될 수 있을 것으로 기대된다.

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Publications from Thesis

1. **Subodh Pudasaini** and Seokjoo Shin, "***Game Theoretic Access Probability based Bandwidth Request Mechanism for WiMAX Point to Multipoint Networks***", Submitted to **IEEE TENCON, India, 2008.**

2. **Subodh Pudasaini,** Moonsoo Kang and Seokjoo Shin**, "*Uplink Scheduling Mechanism for MPEG Encoded Video IEEE 802.16 MAN*",** Submitted to **IEEE WiMOB, France, 2008.**

3. **Subodh Pudasaini** and Seokjoo Shin**, "*FMLP based Traffic Prediction for Adaptive Resource Allocation in WiMAX VBR Applications*", KICS, Korea 2008**.

Acronyms

AAS	Adaptive Antenna System
ADAP	Announced Dynamic Access Probability
ANFIS	Adaptive Neuro Fuzzy Inference System
ATM	Asynchronous Transfer Mode
BE	Best Effort
BS	Base Station
BWA	Broad-band Wireless Access
BW-REQ	Bandwidth Request
CAC	Call Admission Control
CBR	Constant Bit Rate
COV	Coefficient of Variance
CRC	Cyclic Redundancy Check
CS	Convergence Sublayer
DCCP	Dynamic Congestion Control Protocol
DL-MAP	Down Link MAP
DOCSIS	Data Over Cable Specification Interface System
DSL	Digital Subscriber Links
EDF	Earliest Deadline First
ertPS	Extended Real Time Polling Services
FIS	Fuzzy Inference System
FMLP	Feed Forward Multi Layer Perceptron
FTP	File Transfer Protocol
GOB	Group of Blocks
GOP	Group of Pictures
GPC	Grant Per Connection
GPSS	Grant Per Subscriber Station
GTAP-BRM	Game Theoretic Access Probability based Bandwidth Request Mechanism

HARQ	Hybrid Automatic Repeat Request
HDTV	High Definition Television
HOL	Head of Line
IE	Information Element
IP	Internet Protocol
IPCS	IP Convergence Sublayer
MAC	Medium Access Control
MAN	Metropolitan Area Network
MPEG	Moving Picture Expert Group
MRL-NN	Multi Resolution Learning based Neural Networks
MTU	Maximum Transfer Unit
NE	Nash Equilibrium
NPI	Nominal Polling Interval
nrtPS	Non Real Time Polling Service
PDU	Protocol Data Unit
PLR	Packet Loss Ratio
PMP	Point to Multipoint
QoS	Quality of Service
Q-rtPS	Quasi Real Time Polling Service
RGPSS	Requests and Grant Per Subscriber Station
RLC	Radio Link Control
RMSE	Root Mean Square Error
RTG	Receive/Transmit Transition Gap
RTP	Real-time Transport Protocol
rtPS	Real Time Polling Service
SS	Subscriber Station
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TLBA	Two Level Bandwidth Allocation
TTG	Transmit/Receive Transition Gap
UDP	User Datagram Protocol

UGS	Unsolicited Grant Service
UL	Uplink
UL-MAP	Uplink MAP
UMTS	Universal Mobile Telecommunication System
VBR	Variable Bit Rate
VOD	Video on Demand
WiMAX	Worldwide Interoperability of Microwave Access

I. Introduction

A. Research Overview

Real time VBR video traffic generated from diverse multimedia applications such as video conferencing, video streaming, VOD (Video on Demand) and IP broadcasting is expected to be the major portion of traffic in future integrated services networks. For transmission over networks, video is typically encoded to reduce bandwidth requirement. For encoding there could be two approaches: in the first approach CBR (Constant Bit Rate) encoding is adopted while in another VBR encoding is adopted. In CBR encoding, the quantization scale is dynamically adjusted to keep the bit rate at a prescribed level while in VBR encoding the quantization scale is kept constant, resulting in essentially constant video quality but highly varying bit rate. Since VBR encoding provides consistent video quality and can achieve more efficient compression, it is the focus of much of the multimedia applications. Efficient transmission of the VBR video depends on the parameters like encoding mechanism applied at source and QoS provisioning architecture of the network as shown in Fig. 1-1.

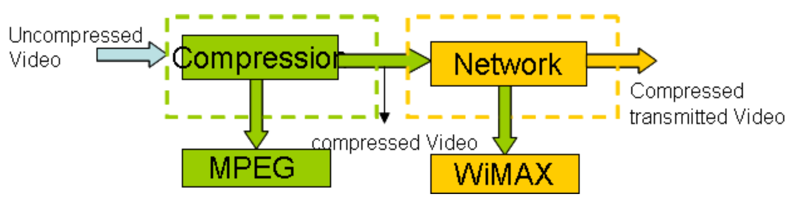


Figure 1-1: Generic representative model for video transmission

Even compressed video, however, requires large bandwidth on the order of 100 kbps or even few Mbps. In addition compressed video streams typically exhibit highly variable bit rates. Aforementioned characteristic in conjunction with the stringent quality of service requirements (loss and delay) of video traffic makes the transport of video traffic over communication networks a challenging problem.

As consequences, networking research on all aspects of video transport has exploded in the recent years. Video traffic modeling and enhancing protocols/mechanism for the efficient transport of video traffic have received a great deal of interest among network researchers and operators.

B. Research Objective

The diverse application domains like broadband fixed wireless access, WiFi backhauling, nomadic internet access and high data rate access with seamless sessions, WiMAX seems undisputed in leading the present BWA (Broad-band Wireless Access) systems [1]. It arises like an alternative to fiber optic links, coaxial systems using cable modem system as in DOCSIS (Data over Cable Specification Interface System) and DSL (Digital Subscriber Links). Since the BWA technology is now offering higher bandwidth, many new and multimedia applications are emerging. These emerging applications exercise ever-increasing demand for QoS guarantees. To enhance the reliability of the performance of the user applications when they are being transported to the broadband network, characteristics of the applications should be handled well to offer better service. The major objective of this carried research is to study the UL (Uplink) video transmission and to suggest some techniques and algorithms which could improve the performance in terms of PLR (Packet Loss Ratio) and delay.

In WiMAX PMP, two BW-REQ mechanisms are provisioned, namely, contention based BW-REQ and contention-free based BW-REQ (polling). Signaling delay in polling based access is predictable. So, unicast polling is preferred for the real time applications. Unicast polling avoids the request collision. The price paid for collision avoidance in unicasting is wastage of bandwidth in general and long polling delay when network is saturated with highly densed users. Solution for the aforementioned drawback could be multicasting, adaptive multicasting and broadcasting. Broadcasting is similar to contention based access while adaptive multi casting needs some modification in the standard with the definition of new QoS class namely Q-rtPS (Quasi Real Time Polling Service) and additional signaling as in [2]. In this work, however, our objective is to suggest a simple solution which needs no additional signaling and no modification in the standard with the broadcasted access probability based contending BW-REQ mechanism applicable to delay constraint rtPS class.

For the VBR video application belonging to rtPS, resource allocation should be done in such a way that their delay obligation should be met with the least possible PLR performance. If the SS (Subscriber Station) is polled to issue a bandwidth request at t_r, grant response will be issued by BS at t_g implying $t_g > t_r$. It indicates that this request grant procedures always adds delay even for the packets that already arrived in the queue. The data packets that will arrive in the queue between the interval (t_r, t_g) will have to wait up to NPI (Next Polling Interval), t_{npi} for making request which results longer delay resulting increase in PLR since the packets with expired deadlines will simply be dropped. The long delay suffered, $t_{npi} - t_g$, can be reduced if we can make request for the video packets at t_r not only for the packets that are already suited in queue at t_r but also for packets that might arrive in the queue up to t_g. For this aforementioned problem, our objective is to propose an online traffic prediction model.

In WiMAX, SSs notify the BS about the amount of bytes to be sent by a connection through specific MAC headers. While bandwidth is requested by a SS per connection, the BS grants uplink bandwidth to a SS in a lump sum manner. This process results a necessity of TLBA (Two Level Bandwidth Allocation). Due to the hybrid nature of the request grant mechanism (i.e. requests per connection, grants per SS), an SS also has to implement locally a scheduling algorithm to redistribute the granted capacity to all of its connections. Every connection is classified according to their QoS requirements. So for every QoS classes there should be class based scheduling algorithm which can maintain required QoS profiles efficiently. At this point, our objective is to develop an opportunistic scheduler for rtPS at SS which could offer better performance for MPEG encoded video in terms of PLR, than the scheduler with sequential forwarding currently being practiced or studied. At this point we anticipate least possible degrade in video quality will be observed if degrade is sure to happen due to resource constraint in the system.

C. Research layout

The research layout we followed was in accordance with the objective of the carried research. We structured our works in two parts. First part deals with the bandwidth request mechanisms while second part covers the scheduling mechanisms. The characteristics focuses of the work segmented on two parts are presented diagrammatically in Fig. 1-2.

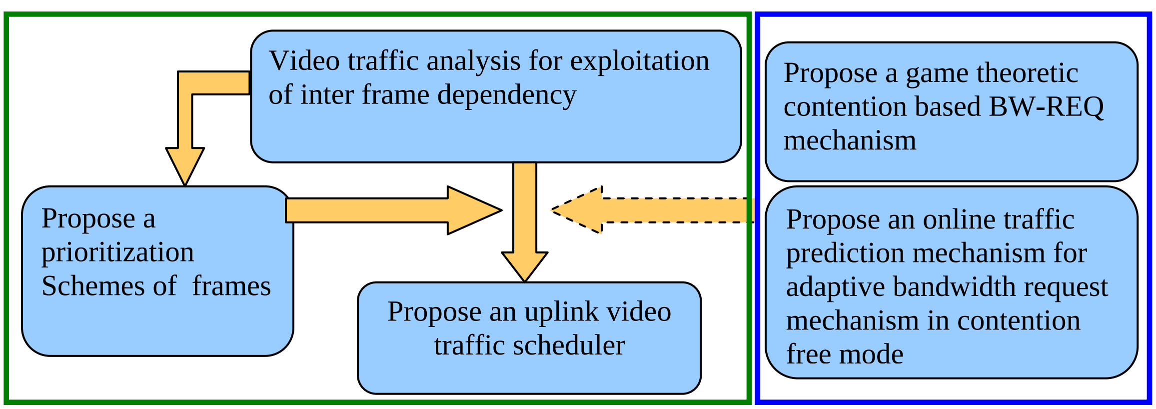


Figure 1-2: Diagrammatic representation of the carried research

D. Thesis Contribution

The characteristics parts of the carried research work are summarized under the title of thesis contribution. They are as follows

- Proposition of game theoretic access probability based BW-REQ mechanism as a novel alternative to polling based BW-REQ mechanism.
- Proposition of traffic prediction model for that can be used in polling based bandwidth request grant mechanism for dynamic bandwidth allocation. Applicability condition for prediction based request grant is also suggested in terms of nominal polling interval and MAC frame length.
- Proposition of prioritization scheme for MPEG frames subjected to be transmitted in BWA systems according to the importance of the frames in the visual quality. In other words, inter frame dependency of the MPEG is used as the reference for the prioritization. With reference of this prioritization scheme an uplink scheduling algorithm is also proposed and performance study is carried.

E. Thesis organization

The content of this thesis is organized in modular chapters. Chapter 2 is devoted to brief overview of bandwidth request mechanism and scheduling architecture of BWA system, namely, WiMAX. Related works are also included in this chapter. In chapter 3, proposed GTAP-BRM is presented. Why online traffic prediction is needed and how it can be realized is presented in chapter 4. Chapter 5 contains the proposed prioritization scheme and scheduler with the detail performance evaluation. The necessity of such scheduler is also presented in the same chapter. This thesis is concluded in the last chapter with the wrapping text for summary of carried research and possible future works.

II. Bandwidth Request and Scheduling Mechanisms in BWA Systems

A. Request Grant Mechanism: An Overview

In IEEE 802.16, bandwidth request grant mechanism is employed to control medium access. Bandwidth request is made by connections affiliated to any SS and grant is given by BS. Request grant schemes can be classified as shown in Fig. 2-1. The purpose of the request-grant procedure is to allow every user; if possible, to have the suitable QoS required for his application. BS uses two control messages, DL-MAP and UL-MAP for this purpose. DL-MAP is for organizing the downlink while UL-MAP is for organizing the uplink transmission. The UL-MAP message notifies each SS about the start and end time of the uplink grants.

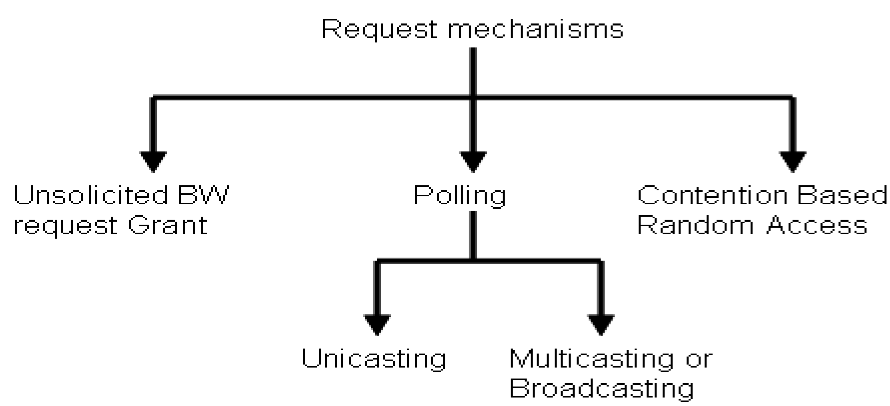


Figure 2-1: Classification of bandwidth request grant mechanism in WiMAX

There are five different QoS classes namely UGS, ertPS, rtPS, nrtPS and BE. Every connections of each QoS class are supposed to follow the predefined request grant mechanism. For the UGS user there is no need to follow the request grant procedure. Because when they are accepted in the system they will be given unsolicited constant grant periodically. However, for the rtPS and ertPS, they should request the required bandwidth when they are polled. Usually they are polled in unicast fashion to issue BW-REQ. Usually, request is acknowledged with bandwidth grant. That bandwidth grant message is contained in UL-MAP of next downlink sub frame. Uplink IE (Information Element) is also included in the UL-MAP. This IE is provisioned to point the beginning and the ending time of the grant and the SSs that the grant is addressed to. This can be interpreted as: even though the resources are scheduled per connection, all grants in sub frame addressed to different connections belonging to the same SS are characterized by IE. Thus, there is a need of implementation of a local scheduler at SS to select which of its connection will transmit in the assigned grant. And with the information of UL-MAP, locally placed scheduler at SS can transmit data belonging to multiple connections in the assigned slots.

1. Request Mechanisms

In TDD/ TDMA frame structure of IEEE 802.16, uplink sub frame consists of request slots and data slots. How these request slots are used to make BW-REQ makes the distinction in request mechanisms.

QoS Classes	**Access Mechanism**			
	Polling		**Contention based Access**	
	Unicasting	**Multicasting**		
UGS	**NN**	**NN**	**NN**	
ertPS	**A**	**A**	**NA**	
rtPS	**A**	**EA**	**NA**	**(EA)**
nrtPS	**A**	**A**	**A**	
BE	**NA**	**NA**	**A**	

NN: No Need
A: Allowed
EA: Extended to be Allowed
NA: Not Allowed

Our proposal

Table 2-1: Request mechanisms for IEEE 802.16

The request mechanisms that are allowed and restricted for individual QoS classes are tabulated with legends A and NA in the table 2-1. EA are the latest suggested works not covered in the standard.

a. Contention based BW Request Mechanism

Contention based access implies random access vulnerable to contention. Bandwidth request with random access is mainly allowed for application with non timing constraints for QoS classes like nrtPS and BE. However, even for real time applications also random access based request grant mechanism is studied in [4]. The contention based access produce some delay if contention occurs. Sum of access delay and queue delay (during scheduling after request is acknowledged with grant) will be the mean delay for any application. Increase in access delay will definitely elongate mean delay which is unfavorable for time critical real time applications. However, if we can guarantee high level of success probability with less access delay and if that is comparatively very low than the queuing delay, in such case contention based bandwidth request grant can be used for real time applications as well.

b. Contention-free (Polling) based BW Request Mechanism

In BWA systems, polling corresponds to selecting a particular SS allowing it to ask for bandwidth request. Polling based request grant mechanism is specially designed for time critical applications. Polling based access is applied to all QoS classes except BE. This can be noted in table 2-1. In rtPS, polling is regular and periodic. So there should be regular polling without violating the TPG (Tolerable Polling Jitter) however in nrtPS polling interval is long and can be non periodic as well.

The importance of polling based bandwidth request is particularly important for AAS (Adaptive Antenna System) subscribers since they might not be able to request bandwidth using the usual random access contention mechanism. This happens because the adaptive array may not have a beam directed at the SS when it is requesting bandwidth, and the bandwidth request will be lost. In such circumstances polling is the best way to adopt for the bandwidth request [4].

2. Grant Mechanisms

a. Grant per Subscriber Station (GPSS)

In GPSS mode, the BS grants bandwidth grant per SS basis even though the request was made per connection basis. SS is responsible to distribute the granted bandwidth among its connections, maintaining QoS requirements and priority agreements. Hence in the GPSS mode, there is a need of implementation of local scheduler at SS. It is more suitable for many connections per SS with real time applications.

b. Grant per Connection (GPC)

In GPC mode, SS receives grants only for specific connections. It is mainly suitable for SS with low number of connections. Additionally, in GPC mode, SS must request additional bandwidth to meet its unexpected Radio Link Control (RLC) requirements. For this reason GPC mode is less efficient than GPSS mode. In this mode there is no need of scheduler at SS. GPC is no more existing in the current updated standard of IEEE 802.16.

B. Scheduling in BWA: An Overview

In the BWA systems, where there is already the provision of differentiated QoS service classes according to the specific requirement of each type of service, there should be differentiation in the way how the bandwidth is allocated to them and how that allocated bandwidth is utilized in terms of either better utilization or higher throughput. These issues are covered in the scheduling architecture.

Basically scheduling architecture is considered especially to endorse all the parameters that could be the matter of interest for every standard's protocol stack. The scheduling architecture defined for the BWA system is simply to make confident provision for delivering specified QoS to the users. As a prominent candidate of BWA, WiMAX also defined its scheduling architecture which make scheduling possible only after two way signaling termed as bandwidth request grant procedure. The scheduling architecture considered in WiMAX is discussed in subsection 1.

1. Scheduling Architecture in WiMAX

The entities in the WiMAX systems are BS and SSs. In centralized architecture, different SS are served by a single BS. The scheduling architecture for SS is presented in Fig. 2-2.

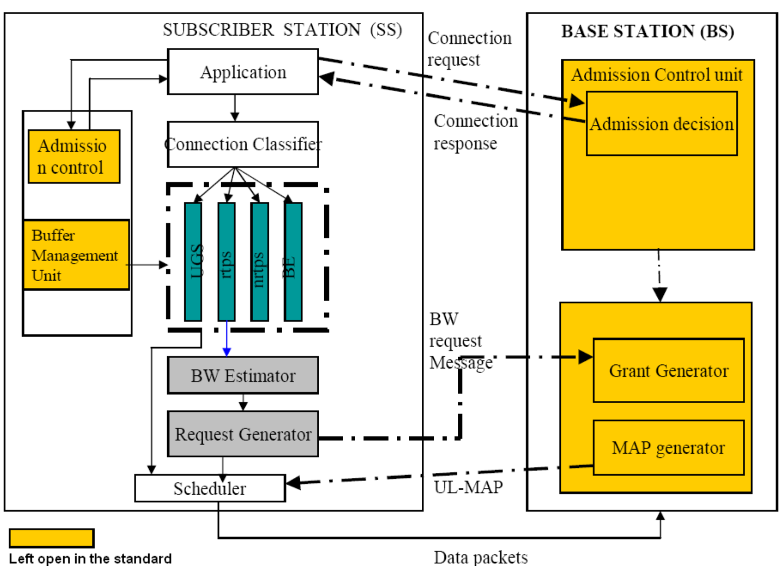


Figure 2-2: Scheduling architecture in WiMAX

This architecture covers the signaling mechanism that must be performed for the resource negotiation. Scheduling should be followed in accordance with the granted resource resulting from the negotiation. However, the way of grant generation, way of scheduling at SS, admission control and buffer management are left unspecified in the IEEE 802.16.

2. Hierarchical scheduling in WiMAX

In IEEE 802.16 bandwidth allocation is hierarchical in nature [3] as shown in Fig 2-3. This figure typically shows the TLBA (Two Layer Bandwidth Allocation) which dynamically allocates bandwidth according to instantaneous demand on SS. Among the total available bandwidth, bandwidth available for the uplink is distributed to the all active SS. This allocation is the BS-Level allocation. In every SS there could be k number of connections. The allocated bandwidth to the SS is to be distributed to these multiple connections. The distribution at this stage is called SS-Level allocation. Collectively, these two layer allocation results TLBA. This hierarchical bandwidth scheduling is only the option for WiMAX to comply GPSS mode provisioned in IEEE 802.16.

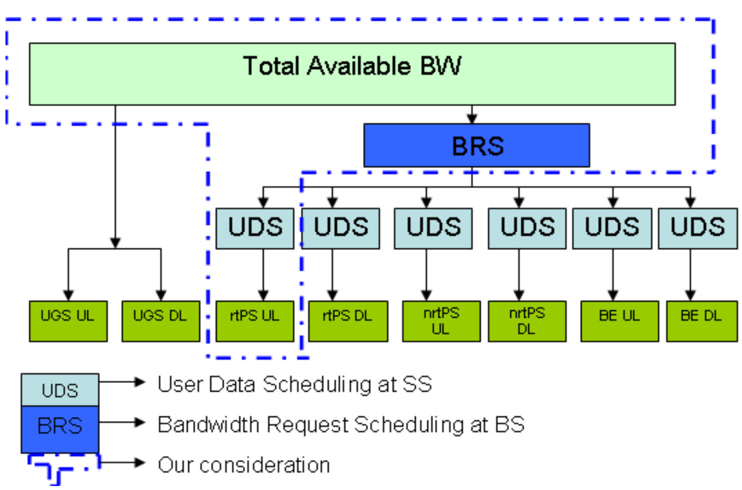


Figure 2-3: TLBA scheduling architecture for WiMAX

Let us denote the total available uplink bandwidth in the system be BW_{UL}. In the network UGS, rtPS, nrtPS and BE users occur simultaneously. Among these classes UGS has the highest priority and the priority goes on decreasing while moving towards right. So when the system is loaded heavily, there could be the probability that lower priority class may suffer from starvation of resource. So not to let that situation happen, assigning certain weighting factor for bandwidth to each QoS class is widely accepted technique. Again, let us assume that W_{rtPS} be the weight for the rtPS QoS class. Hence it can be said that, available bandwidth for the rtPS is

$$BW_{rtPS}^{total} = BW_{UL} \times W_{rtPS} \tag{2-1}$$

BW_{rtPS}^{total} can be distributed to the SSs in different possible ways. The easiest way is to divide equally to all SSs.

$$G\_BW_{rtPS}^{i} = \frac{BW_{rtPS}^{total}}{S} \tag{2-2}$$

Where $G\_BW_{rtPS}^{i}$ is the granted bandwidth to ith SS when there are S number of SS in the system. Unfortunately, this approach could lead to starvation of resource to some SS and excess of resource to other remaining SSs, if the number of connection per SS varies widely. So to handle this type of hindrance, available resource can be distributed in the following way

$$G\_BW_{i}^{rtps} = \frac{R\_BW_{i}^{rtps}}{\sum_{i=1}^{S} R\_BW_{i}^{rtPS}} \times BW_{rtPS}^{total} \tag{2-3}$$

Where,

$R\_BW_{i}^{rtps}$ is the requested bandwidth size per SS for rtPS class. So number of the granted slot will be

$$\# slots_i^{rtps} = \left\lceil \frac{G\_BW_i^{rtps}}{S_i} \times f_l \right\rceil \qquad (2\text{-}4)$$

Where,

f_l is the frame length. This distribution of the slots to connections belonging to the i^{th} SS is as per the scheduler locally installed at SS. Information of grant is broadcasted in UL-MAP in DL subframe once every scheduling period. Thus one map frame conveys bandwidth allocation information with regard to one scheduling period. For example if one scheduling period is 10ms and one mini slot is 6.25 μs, one map manages allocation of 1600 mini slots. So in this case one MAP will broadcast information of 1600 mini slots is available in total. It's the scheduler at SS who will distribute it to different connections affiliated with it.

C. Related Works

Our work basically consists three parts as stated in the research objective. First part covers the contention based bandwidth request for rtPS; second part aims to make improvement in the contention free access with dynamic bandwidth request capability with online prediction of future video frames and third is the scheduling mechanism with opportunistic behavior to minimize distortion that could arise because of frame loss. So, related works for all three parts are presented separately in the subsequent paragraphs respectively.

From the widely available literature, some remarkable contributions can be noted for contention based BW-REQ mechanism. In [4-5], contention based access for WiMAX is explored well. In [4], it is compared with contention free polling and reported that random access request mechanism is more efficient than polling when the request rate is low. In [5], the authors reported the signaling overhead for the contention based access increases with the increase in the contention period and suggested a new RGPSS (Requests and Grants Per SS) as a modification of GPSS. In both [4] and [5] access is random. In [6], A-DAP (Announced Dynamic Access Probability) based access is proposed for next generation wireless networks. The proposed GTAP-BRM is related with the concept of A-DAP with the game theoretic solution for the access probabilities to be broadcasted.

For closely realizing the VBR traffic many analytical models like renewal models, markovian models and self similar traffic models are presented in [7]. Non linear model based on FMLP for accurately capturing the burstiness of the VBR traffic is more common these days in research community. Most accurate prediction based on cubic spline interpolation for MPEG VBR video is proposed in [8]. ANFIS implementation of the pre encoded input output pair is used to test existence of packet lost in different rate of bandwidth usage in [9]. Even though the tool used for the prediction is common to [9], our prediction is different since we are using chaotic time series represented training information set constructed from trace. Our primary reference is [10] which introduced the concept of dynamic bandwidth allocation with online traffic prediction with MRL-NN (Multi Resolution Learning based Neural Networks).

For video applications, MPEG (Moving Picture Expert Group) is the preferred encoding mechanism since it offers high compressibility. Compressibility enjoyed with MPEG encoding, however, add inter frame dependency between the frames [11]. So, in order to mitigate adverse performance that could occur because of inter frame dependency during transmission, different opportunistic approaches are suggested in [12-15]. In [12] selective frame discard in DCCP (Dynamic Congestion Control Protocol) with only I and P frames is introduced to achieve the lowest possible degrade on video quality whenever congestion occurs. Buffer management with data differentiation for proactive B-dropping is suggested in [13]. In [14], distortion minimizing network approach for scheduling video streaming data in UMTS (Universal Mobile Telecommunication System) for single user case is presented. In [15], selection of substream from the nested substreams of a video stream for transmission is presented. The selection is carried according to the importance value of the substream with reference to effective bandwidth. Our approach, however, is priority based approach which exploits the inter frame dependent information of frames within GOP to add priority weight to each frame and correspondingly to each video packet to be scheduled.

III. Game Theoretic Access Probability based Bandwidth Request Mechanism (GTAP-BRM)

A. Introduction

In WiMAX PMP, two bandwidth request mechanisms are provisioned, namely, contention based BW-REQ and contention-free based BW-REQ (polling). Signaling delay in polling based access is predictable. So, unicast polling is preferred for the real time applications. Unicast polling avoids the request collision. The price paid for collision avoidance in unicasting is wastage of bandwidth in general and long polling delay when network is saturated with highly densed users. Solution for the aforementioned drawback could be multicasting, adaptive multicasting and broadcasting. Broadcasting is similar to contention based access while adaptive multi casting needs some more further modification in the standard with the definition of new QoS class namely Q-rtPS an additional signaling as in [2]. In this carried research, however, we tried a simple solution which needs no additional signaling and no modification in the standard with the broadcasted access probability based contending BW-REQ mechanism applicable to delay constraint rtPS class.

The importance of contention free polling based bandwidth request is particularly important for AAS subscribers since they might not be able to request bandwidth using the usual random access contention mechanism. This happens because the adaptive array may not have a beam directed at the SS when it is requesting bandwidth, and the bandwidth request will be lost [2]. In such circumstances polling is the best way to adopt for the bandwidth request. In this work, we didn't confine ourselves towards the enabled optional AAS. Hence, AAS which restrict the bandwidth request grant procedure for rtPS to be polling only doesn't oppose our idea of making contention based access for bandwidth request even for real time applications.

In the contention based access, some delay is experienced if contention exists. Access delay and scheduling delay will be the dominant factors for calculating mean delay for any application. Increase in access delay will definitely elongate mean delay which is unfavorable for time critical real time applications. In contention based access, however, if we can assure high level of average success probability with less access delay and if that is comparatively very low than the scheduling delay, in such case contention based bandwidth request grant can be used for real time applications as well.

B. Proposed GTAP-BRM

IEEE 802.16 doesn't allow contention based BW-REQ mechanism for rtPS. However, this GTAP-BRM is the contention based BW-REQ mechanism. With the results we obtained, we believe that this can be the novel alternative for the contention free polling. In this work we made provision for contention based BW-REQ with broadcasted access probability for rtPS by employing *game theory* as a tool to prioritize their access resulting almost similar performance as polling based BW-REQ. This implies that the adoption of this scheme will simply increase the bandwidth efficiency since the bandwidth that could have been wasted for unicasting is totally saved. We formulated a *game* to realize GTAP-BRM, in which access probability is determined by solving that game to obtain an equilibrium access probability with which they will access for BW-REQ. AS there are three tuples in the conventional game, in GTAP-BRM, players are connections belonging to either delay constraint rtPS and delay intolerant nrtPS QoS classes, strategies of players are probable access probabilities and players' payoffs are the difference of utility gain from successful access and the cost that is incurred for the unsuccessful access.

1. System Model for GTAP-BRM

In this work we considered a WiMAX PMP environment operating in TDD (Time Division Duplex) mode. The frame structure for this mode is as shown in Fig. 3-1.

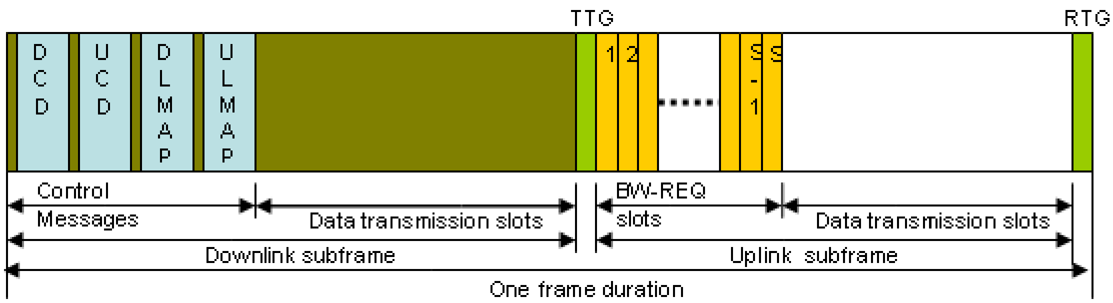


Figure 3-1: Frame structure of WiMAX under TDD mode

A single frame is partitioned into a DL subframe and UL subframe. At the end of each DL (UL) frame there is a provision of time, TTG (RTG), enough to switch from reception (transmission) to transmission (reception). The UL subframe is portioned to BW-REQ slots and the data transmission slots. For the successful BW-REQ made using BW-REQ slots, grant information is broadcasted in the UL MAP. With the information of the grant received, data belonging to that connection which receives grant can transmit its data content in the specified data transmission slots. In the standard random access mechanism is left unspecified. So we formulated an access probability based BW-REQ access mechanism in which access probability assignment is carried using game theory as detailed in subsection B of section II. With the access probability assignment, we can treat connections belonging to the different QoS classes differently. In this proposed mechanism, access probability for connections is broadcasted in UL MAP in downlink sub frame. The connections belonging to multiple SSs to which the broadcasted access probability belongs will access any of the BW-REQ slots among the S available slots with that assigned probability. WiMAX follows a self-correcting protocol rather than an acknowledge protocol [16]. So, there is no provision of the dedicated acknowledgement for the access made about the success or failure. If the grant is received in the next UL-MAP then it is perceived as the successful access, otherwise it should join the random access process up to the timeout appropriate for the QoS of the connection.

2. Game Theoretic Access Probability Assignment

Making the probability assignment formulations simple, we made the following two assumptions:

- The connections are of only two types, either rtPS or nrtPS. They are referenced with notations G_{rtPS} and G_{nrtPS} respectively.
- G_{rtPS} Connections are assigned probable access probability first. The complement of probable access probability assigned for G_{rtPS} will be assigned to G_{nrtPS}. This assumption will introduce prioritization among connections of two considered QoS classes.

For assigning the access probability ap_i to the J connections belonging to the i different QoS classes, we formulate a game. As per the conventional game theory, the game, G can be represented with three tuples as

$$G = (J, ap_i, U_i), ap_i \in [0,1] \tag{3-1}$$

Where, i carries the group identification information as stated below.

$$i = \begin{cases} 0 & : rtPS \\ 1 & : nrtPS \end{cases} \tag{3-2}$$

U_i is called payoff of the game. This payoff is simply the payoff of ith QoS class's connections for playing the game G with strategy ap_i. Among the J connections, some connections belongs to G_{rtPS} and rest in G_{nrtPS}. Let ϕ_{rtPS} be the loading factor for G_{rtPS} and ϕ_{nrtPS} be the loading factor for G_{nrtPS} .The number of connections belonging to each class is given by the following equation.

$$J_{rtPS} = \lceil \phi_{rtPS} \times J \rceil \qquad (3\text{-}3a)$$

$$J_{nrtPS} = J - J_{rtPS} \qquad (3\text{-}3b)$$

$\lceil x \rceil$ is the smallest integer not less than x. If the connections are looking for deterministic access, there could happen only two cases. Either the access is denied or the connection gain access. In game theory domain, it is termed as pure strategy. These two deterministic strategies can be thought as the deterministic probability of 0 and 1. In our game, strategy is mixed because we assign probability to pure strategy. It is possible to assign infinite different probability values for pure strategy. But, only n different values are assigned so as to make the game a finite-horizon game. So set of strategies for connections belonging to G_{rtPS} will be of dimension. These available strategies are the available access probabilities. Similarly the strategy for G_{nrtPS} is $n \times 1$. By multiplying the two strategy spaces, we can get a $n \times n$ matrix which we called matrix of probable access probabilities, AP^{prob} .

$$AP_0 = \begin{bmatrix} \frac{1}{n^2} & \frac{2}{n^2} & . & . & \frac{1}{n} \\ \frac{n+1}{n^2} & \frac{n+2}{n^2} & . & . & \frac{2n}{n^2} \\ . & . & . & . & . \\ . & . & & & . \\ \frac{((n^2-n)+1)}{n^2} & \frac{((n^2-n)+2)}{n^2} & . & . & 1 \end{bmatrix} \qquad (3\text{-}4)$$

Hence, the probable access probabilities for connections belonging to different QoS classes can be stated as

$$\left(ap_{i=0}^{prob}\right)_{k,l} = \left(AP^{prob}\right)_{k,l} \quad \forall \;\; k,l = 1 \; to \; n \qquad (3\text{-}5)$$

$$\left(ap_{i=1}^{prob}\right)_{k,l} = \left(1-\left(AP^{prob}\right)_{k,l}\right) \ \forall \ k,l = 1 \ to \ n \tag{3-6}$$

Let G_{rtPS} and G_{nrtPS} connections decide to access with $(ap_{i=0}^{prob})_{k,l}$ and $(ap_{i=1}^{prob})_{k,l}$ respectively. This means they will not access with the probability of $1-\left(ap_{i=0}^{prob}\right)_{k,l}$ and $1-\left(ap_{i=1}^{prob}\right)_{k,l}$. Hence, the payoff for the successful access can be calculated as

$$\begin{aligned}\left(U_{i=0}\right)_{k,l} &= \left(ap_{i=0}^{prob}\right)_{k,l}\left\{1-\left(ap_{i=1}^{prob}\right)_{k,l}\right\}(1-C) \\ &\quad -\left(ap_{i=0}^{prob}\right)_{k,l}\left(ap_{i=1}^{prob}\right)_{k,l} C\end{aligned} \tag{3-7a}$$

$$\begin{aligned}\left(U_{i=0}\right)_{k,l} &= \left(ap_{i=0}^{prob}\right)_{k,l}\left\{1-C-\left(ap_{i=1}^{prob}\right)_{k,l}\right\} \\ &\quad \forall \ k,l = 1 \ to \ n\end{aligned} \tag{3-7b}$$

Similarly the payoff for another QoS class will be

$$\begin{aligned}\left(U_{i=1}\right)_{k,l} &= \left(ap_{i=1}^{prob}\right)_{k,l}\left\{1-C-\left(ap_{i=0}^{prob}\right)_{k,l}\right\} \\ &\quad \forall \ k,l = 1 \ to \ n\end{aligned} \tag{3-8}$$

$0 \le C \le 1$ is the cost function. If access is successful, some positive value is deducted as cost that is incurred for that successful access. The provisioning of this C make the players doesn't always look for the highest strategy value which could be more than the required strategy value. The payoff matrix will then be generated. Every element in the payoff matrix will have the entry as

$$U_{k,l} = \left\langle \left(U_{i=0}\right)_{k,l}, \left(U_{i=1}\right)_{k,l} \right\rangle \quad \forall \ k,l \ 1 \ to \ n \tag{3-9}$$

By solving the payoff matrix (detail of different solution methods for matrix game can be obtained in [17], we can get the equilibrium payoff $\left(U_{i=o}^{eq}, U_{i=1}^{eq}\right)$. The equilibrium payoff is the NE. NE is the consistent prediction of the outcome of the game. In this equilibrium, no players have incentive to unilaterally deviate the strategy. In general uniqueness and existence of NE is not guaranteed; neither is convergence to equilibrium when one exists. But it is already proved that for mixed strategy at least one NE exists [18]. The proof for the existence of NE is presented in [18] using *fixed point theorem* and *Kakutani theorem*.

/* Define Mixed strategy space size */

$n \leftarrow:$ Up to which the pure strategy is to be divided

$C \leftarrow:$ Access Cost

/* for corresponding $(AP_i)_{k,l}$, $(U_i)_{k,l}$ is calculated */

for i=0 to 1

for k=1 to n

for l=1 to n

$(U_i)_{k,l} \leftarrow: f\left(ap_i^{prob}, C\right)$

end for

end for

end for

Payoff matrix, U is generated

NE of U is calculated /* NE is Nash Equilibrium */

if no. of NE > 1 /*because there could be multiple NEs */

Choose NE with highest value of $ap_{i=0}$

end if

NE pair is mapped with corresponding access probability

Access probability to each connection of each class is assigned

Table 3-1: Pseudo-code for game theoretic access probability assignment

The random access game considered in this carried research is mixed strategy game. So in this formulated game of access probability based access mechanism, NE is always to exist. Equilibrium payoff is mapped reversely to the probable access probabilities which yielded that equilibrium payoff. That particular probable access probability will be the access probability, ap_i which will be assigned to the connections belonging to the particular QoS classes, G_{rtPS} and G_{nrtPS} connections respectively. The overall procedure of access probability assignment can be summarized with pseudo code as presented in table 3-1.

3. Performance Evaluation of GTAP-BRM

Now we investigate the basic formulae for analyzing the performance of the proposed BW-REQ mechanism. Let us assume that S slots are allocated in the current uplink subframe for bandwidth requests. One connection, which attempts the random access, will randomly select a slot numbered between 1 to S to send the bandwidth request. The probability of that event of selecting any of the available slots is $\frac{1}{S}$. If the numbers of all users who are attempting random access is J, the probability of the given resource that is chosen by that connection and not selected by other remaining connections can be written as

$$P_{no\_collission} = \left(1 - \frac{1}{S}\right)^{J-1} \qquad (3\text{-}10)$$

In the pool of connections attempting access for BW-REQ, probability that only j connections succeed in accessing and other $k = J - j$ connections fail is given by

$$P_s\left(j \mid J\right) = \begin{cases} \dfrac{\left(S-j\right)^{J-j} \coprod_{k=1}^{J-1} \left(S-k\right)^{J-1}}{S^{J-1}} & , for\ j \rangle 1 \\ \left(1 - \dfrac{1}{S}\right)^{J-1} & , for\ \ j = 1 \end{cases} \qquad (3\text{-}11)$$

With this information, we will investigate the average success probability and average access delay for both G_{rtPS} and G_{nrtPS} connections. When J connections of the considered two QoS classes are attempting for the bandwidth request, the number of connections of each class, J_{rtPS} and J_{nrtPS} are given by equation 3-3a and 3-3b respectively. Let us again define (J_0, J_1) as a vector which consists of the number of succeeded attempts of connections, where $0 \le J_0 \le J_{rtPS}$ and $0 \le J_0 \le J_{nrtPS}$. Then the probability that the event (J_0, J_1) occurs is given by

$$P(J_0, J_1) = \left[\binom{J_{rtPS}}{J_0} ap_0^{J_0} (1 - ap_0)^{J_{rtPS} - J_0} \right] \times \left[\binom{J_{nrtPS}}{J_1} ap_1^{J_1} (1 - ap_1)^{J_{nrtPS} - J_1} \right] \tag{3-12}$$

The success probability that G_{rtPS} connections will receive will be

$$P_{success}^{rtPS}(j \mid J_0) = \sum_{J_1=1}^{J_{nrtPS}} P_s^{rtPS}(j \mid J_0) P(J_0, J_1) \tag{3-13}$$

Finally, the average success probability of total accessing connections of rtPS without collision will be

$$P_{avgsuccess}^{rtPS} = \frac{\sum_{j=1}^{J_{rtps}} j \sum_{J_0}^{J_{rtPS}} P_{success}^{rtPS}(j \mid J_0)}{J_{rtPS}} \tag{3-14}$$

Similarly, average success probability of nrtPS can be derived.

For the access delay, the time delay is characterized as the time needed for the successful connection. It is represented as

$$D_{rtPS} = T_{frame} \times \sum_{x=1}^{\infty} x(1 - P_{avgsuccess}^{rtPS})^{x-1} P_{avgsuccess}^{rtPS} \quad \text{(3-15a)}$$

$$D_{rtPS} = \frac{T_{frame}}{P_{success,rtPS}} \quad \text{(3-15b)}$$

Where, T_{frame} is the duration of one MAC frame.

The analysis is carried out in an integrated environment of MATLAB and GAMBIT [19]. The parameters chosen for the analysis are summarized in table 3-2.

Parameters	**Value**
No. of connections	30
ϕ_{rtPS}	30%
ϕ_{nrtPS}	70%
C	0.050
T_{frame}	1ms and 10 ms

Table 3-2: Parameters considered for the numerical analysis of GTAP-BRM

There are 30 numbers of connections. Of all the connections attempting contention based BW-REQ, ϕ_{rtPS} is 30% and ϕ_{nrtPS} is 70%. This is based in the system that can occur in actual systems. This access attempt ratio (30% G_{rtPS} and 70% G_{nrtPS}) may be changed depending on specific environments. n is taken as 4. So probable access probability matrix size is 4×4. The frame length is taken as 1 and 10 ms for two different situations. The cost value, C for every access is taken to be 0.050. It is noteworthy that the payoff decreases as the value of C increases. This can be noted in Fig. 3-2. Hence, it is desirable to reduce C as much as possible. If the cost of access is made high, there will be change in the payoff matrix elements' values because of change in C , the equilibrium resulted from that changed payoff values shift to some other values as shown in Fig. 3-3.

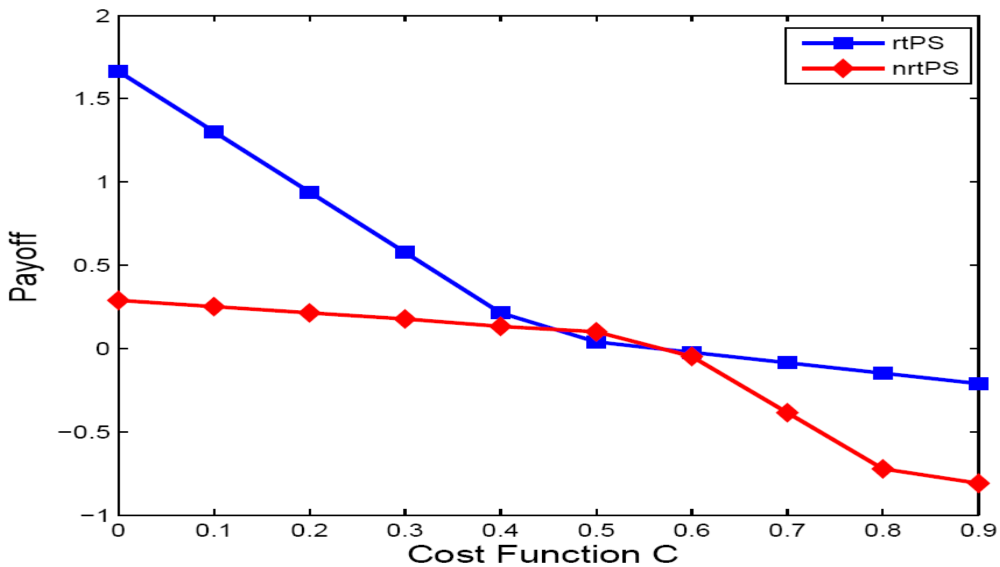


Figure 3-2: Payoff versus cost function, C

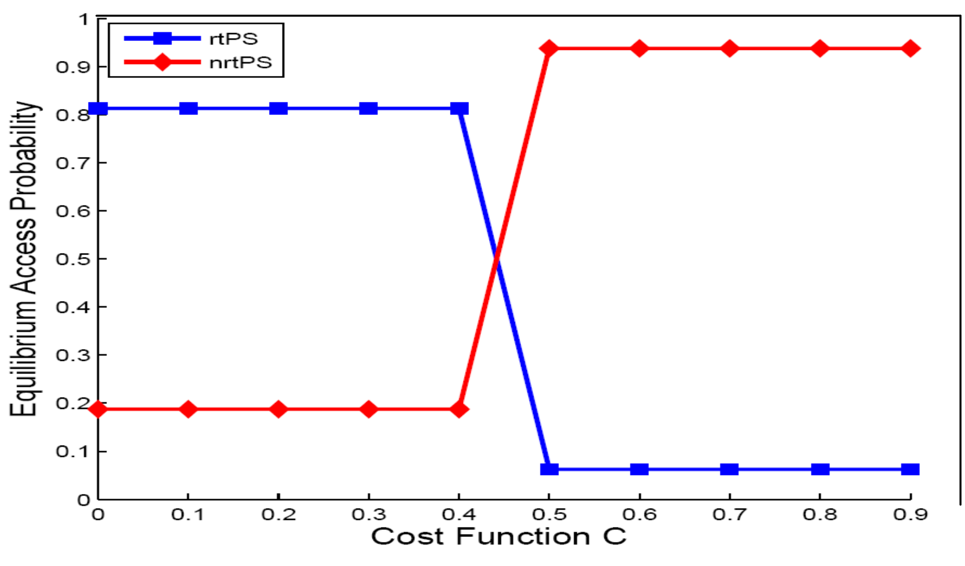


Figure 3-3: Equilibrium access probability versus cost function

The NE for the generated payoff for this particular case when 30 users are present in the system is found to be $\left(U_{i=o}^{eq}, U_{i=1}^{eq}\right)$= (1.4828, 0.273) for G_{rtPS} and G_{nrtPS} connections. For this equilibrium, the corresponding access probability is found to be 0.8125 and 0.1875. This equilibrium access probability is obtained when the value of C is 0.050. If the cost function is maintained well below 0.45, the equilibrium will not shift but if the C exceeds 0.45 the equilibrium alters to some undesirable value which yields low access probability for high priority G_{rtPS} connections as shown in Fig. 3-3. So C should be maintained below a level whose corresponding payoff value is still positive. For e.g., in this particular case should always within the bound, $C \leq 0.45$.

The average success probability and the average access delay for this set of access probabilities are plotted in Fig.3-4 and Fig.3-5 respectively. The provision we made for the time constraint G_{rtPS} connections is to offer them a constant and reliable average success probability incorporating very less average access delay. From Fig. 3-4 and Fig. 3-5 well maintained average success probability and average access delay for G_{rtPS} connections can be observed. Average success probability for all G_{rtPS} connections is almost one. At the same instant, average success probability of less priority class, G_{nrtPS} connections, however decreases with increase in the number of connections and eventually falls to zero yielding even null success probability. Null success probability conveys that their access is denied. So they will re-access with the same procedure and their delay will be elongated even more. This elongation in delay, however, doesn't harm the QoS profile of delay intolerant nrtPS class. Hence the achievement we made for rtPS with the sacrifice of the nrtPS seems logical and acceptable in QoS perceptive. As the success probability of G_{rtPS} connections gradually drops, it is obvious that delay will be elongated as presented in Fig. 3-5. If the ratio of BW-REQ slots to the data slots is high, delay will also be long. The elongation in the delay will be the multiple of ratio of normalized frame duration to success probability. In Fig. 3-5, for the frame length of 10 ms the delay is almost 10 times than the delay when the frame length is 1 ms.

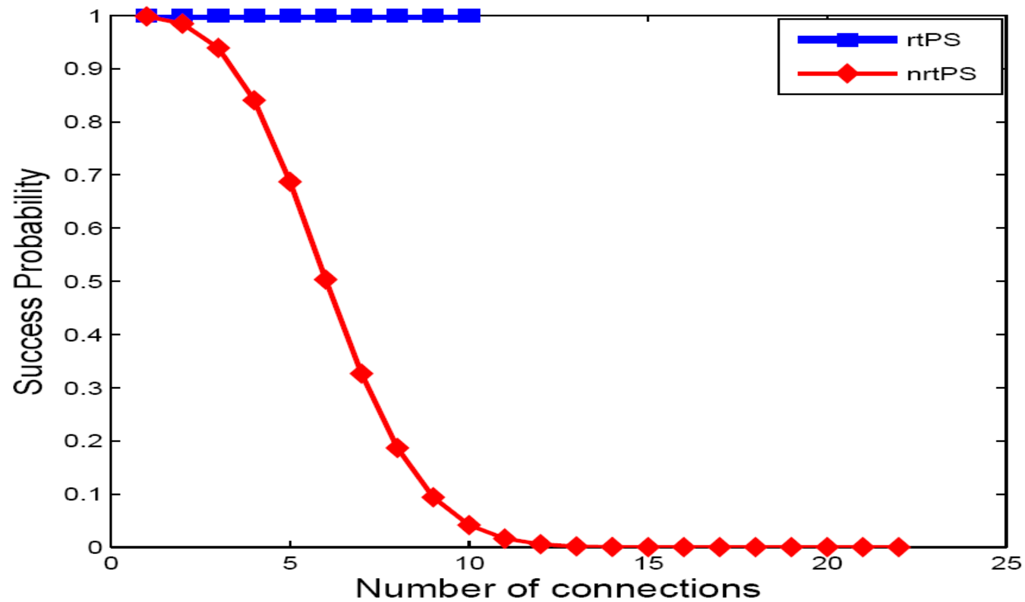


Figure 3-4: Average success probability versus number of connections

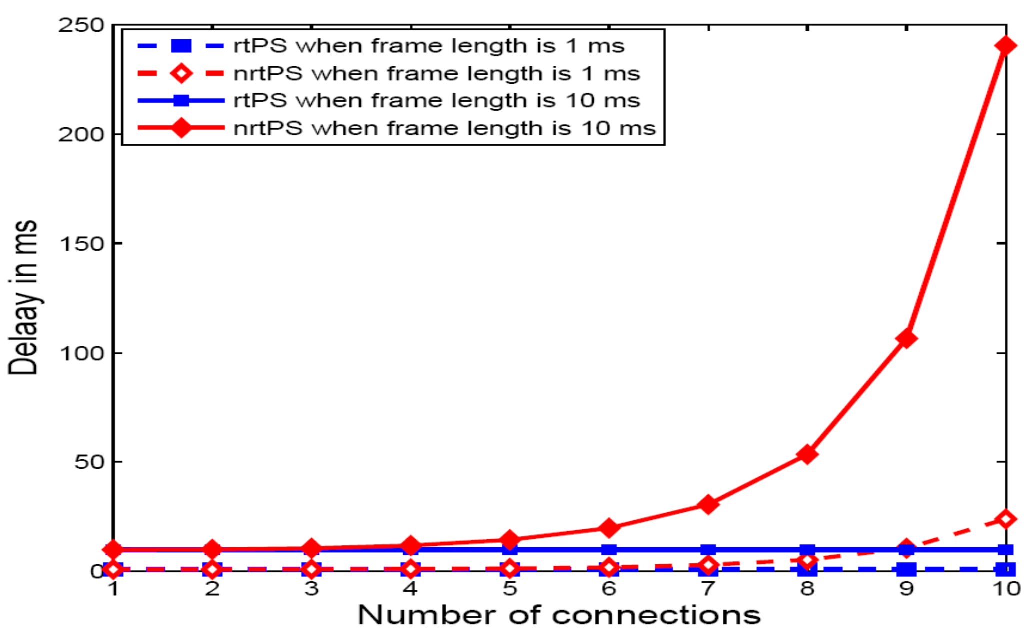


Figure 3-5: Average access delay versus number of connections

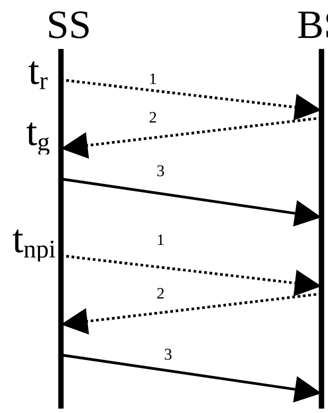
IV. Online Traffic Prediction Mechanism for Adaptive Bandwidth Request

A. Introduction

Real time VBR video traffic generated from diverse multimedia applications such as video conferencing, video streaming, VOD and IP broadcasting is expected to be the major portion of traffic in future integrated services networks. Hence, for study and analysis of network performance under VBR video traffic, modeling of VBR traffic holds the primary concern. VBR applications, however, oblige stringent requirements in terms of QoS, which requires an efficient traffic control and sophisticated network management functions. Some of these functions include CAC (Call Admission Control), traffic shaping, and smoothing and resource management. Design and implementation of such control and management functions can be improvised by making the model which could accurately predict the VBR traffic. How to perceive the traffic dynamics in advance in order to determine the resource needs is the matter of concern of this paper.

IEEE 802.16 MAC protocol is connection oriented protocol. Any application must establish a connection with BS as well as classify itself to one of the provisioned QoS service classes in [20] and follow bandwidth management procedures in conjunction with BS before it can transmit data. For rtPS service type, bandwidth request grant procedure as shown in Fig. 4-1 will be followed for simple resource allocation.

For the VBR video application belonging to rtPS, resource allocation should be done in such a way that their delay obligation should be met with the least possible PLR performance.



1: BW-BR 2: BW-Grant 3: Scheduled Video

Figure 4-1: Bandwidth request grant procedure for rtPS in WiMAX

If the SS is polled to issue a bandwidth request at t_r, grant response will be issued by BS at t_g implying $t_g > t_r$. It indicates that this request grant procedures always adds delay even for the packets that already arrived in the queue. The data packets that will arrive in the queue between the interval (t_r, t_g) will have to wait up to next polling interval, t_{npi} for making request which results longer delay resulting increase in PLR since the packets with expired deadlines will simply be dropped. The long delay suffered, $t_{npi} - t_g$, can be reduced if we can make request for the video packets at t_r not only for the packets that are already suited in queue at t_r but also for packets that might arrive in the queue up to t_g.The proposed solution to aforementioned problem can be viewed in Fig.4-2. At this point, it is understood that forecast of future traffic parameters is essential to solve the aforementioned problem. To make the forecast, it should be kept in mind that traffic model considered is correctly representing the real VBR work load. Little error during characterizing traffic by the model could be multiplied in the predicted values and the bandwidth management using these elongated error values will be inefficient. So traffic model considered should be accurate enough. Not to welcome the error in characterizing the VBR into our analysis, we selected real trace as our traffic model. We converted the trace information into a chaotic time series and made training input output pairs. The trained ANFIS will make the prediction for the future video frames.

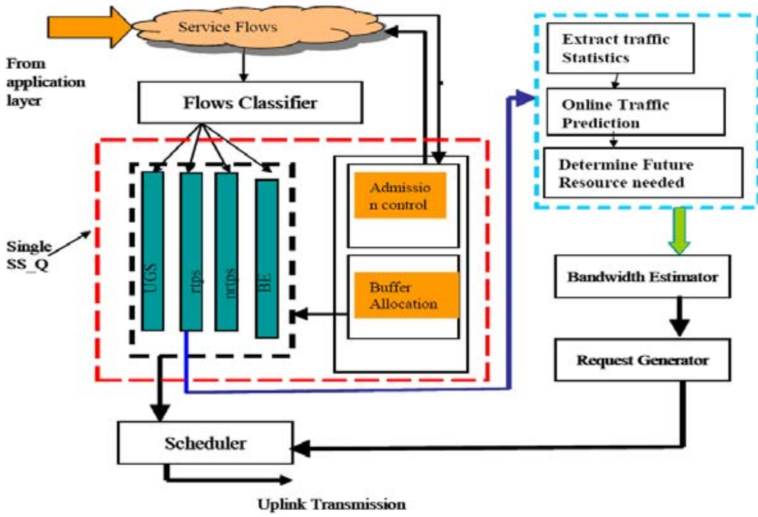


Figure 4-2: Coupled online traffic prediction model in SS architecture

B. Proposed Prediction Model

1. Traffic Model

MPEG provides efficient video coding covering the range from very low bit rates of wireless communication to bit rates and quality labels beyond HDTV (High Definition Television). So we choose MPEG encoded video trace [21] as the video traffic model for our analysis. We took two MPEG encoded movies namely star war IV and silence of the lambs. The frame rate for both encoded video is 25 fps and GOP length of 12 with periodic pattern of IBBPBBPBBPBB. The received trace contains the information like frame index, frame type, time and frame length as tabulated in table 4-1.

Frame Index	Frame Statistic		
	Type	Arrival Time (ms)	Length (Bytes)
1	I	0	919
2	P	120	1161
3	B	40	302
4	B	80	432
.	.	.	.
.	.	.	.

Table 4-1: Trace information of MPEG coded movie

2. ANFIS Prediction Model

ANFIS is a FMLP network which uses neural network learning algorithms and fuzzy reasoning to map an input space to an output space. The prediction with ANFIS starts with extraction of fuzzy rules from training numerical data set and constructing a rule base. In this prediction module we constructed the training input output set with the known values up to the time , say t, to predict the value at some point in the future, say $t + t_g$. In the considered traffic trace frame rate is 25 fps. It corresponds that there is the deterministic arrival of the frames with the constant inter arrival time of 40 ms. If the MAC frame length is 10 ms, that means we are making request for the video frame that will be available in the system after 4 MAC frame time. So applicability of our proposed solution will holds under the criteria:

$$\lceil t_{npi}^{n} \rceil > \frac{1}{[\, fps \times f_l \,]} \tag{4-1}$$

Where, $$t_{npi}^{n} = \frac{t_{npi}}{f_l} \tag{4-2}$$

is the normalized polling interval with respect of frame length, and f_l is the length of one MAC frame.

In this considered ANFIS we applied a hybrid learning algorithm to recognize the parameters of membership function and *Takagi and Sugeno*'s type FIS (Fuzzy Inference System). Membership function considered in this work is bell-shaped. Three bell-shaped membership functions are assigned to each input. A combination of least squares and back propagation gradient descent methods is adapted to train FIS membership function parameters. In our experiment, we make the tri input- one output training set for the frame length. We didn't considered about the frame type information since we simply are interested in the frame size in advance that could appear within the (t_r, t_g). That means with reference to the past 3 frames length we can know what could be the size of the next frame to come.

3. Performance Evaluation

The performance is analyzed using MATLAB. For the evaluation of the frame length prediction, we made two sets, namely training set and testing set. Training and testing sets are constructed as described in the previous section from the trace of Silence of Lambs and Star War IV respectively using the information of 500 frames. The truncated part of the MPEG coded movie used for making training set is shown in Fig. 4-3. We constructed the testing set from different movies simply to ensure that the system can produce equally efficient results for new input values, different from those used during training. The original and the predicted frame values for training is shown in Fig. 4-4 while for testing is shown in Fig.4-5. It can be noted in both figures that original frame size values are almost overlapped by the predicted values. For evaluating the accuracy of the frame length prediction of the considered ANFIS system, we checked the RMSE. RMSE simply gives a quantitative measure on how close the predicted outputs are to the original values. Mathematically, RMSE is simply a ratio between the

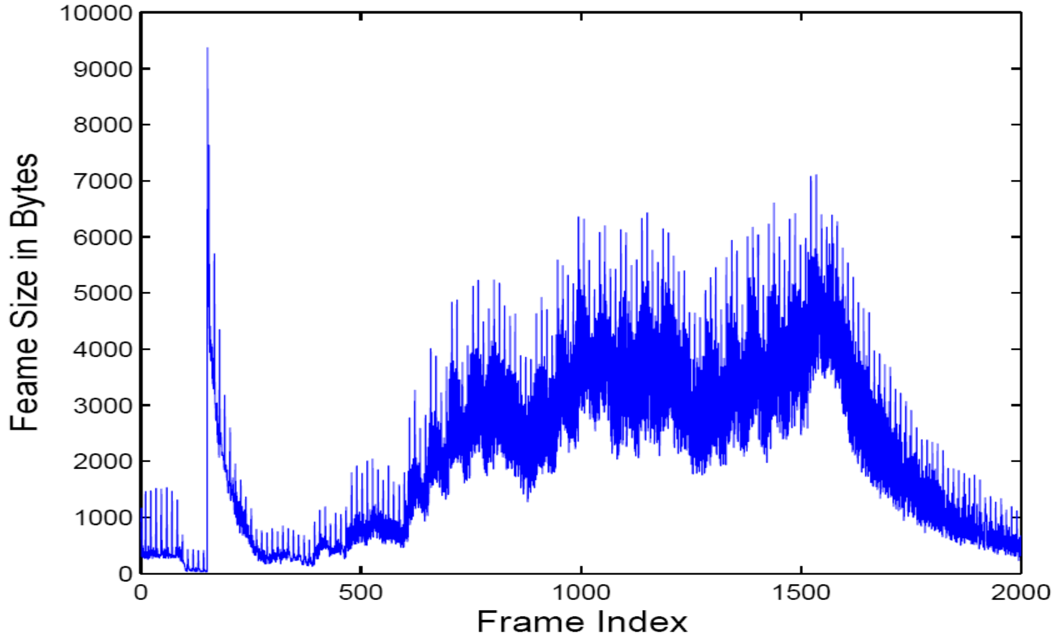


Figure 4-3: Truncated MPEG trace of the movie Silence of Lambs for generating training input output pairs

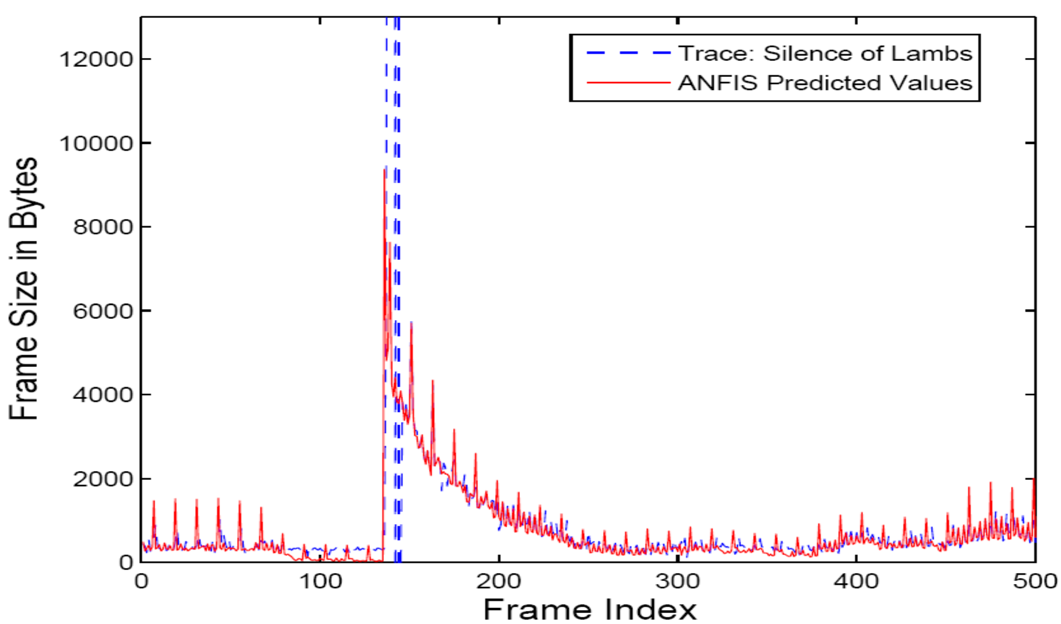


Figure 4-4: Predicted frame length versus frame index for training sets from movie Silence of Lambs

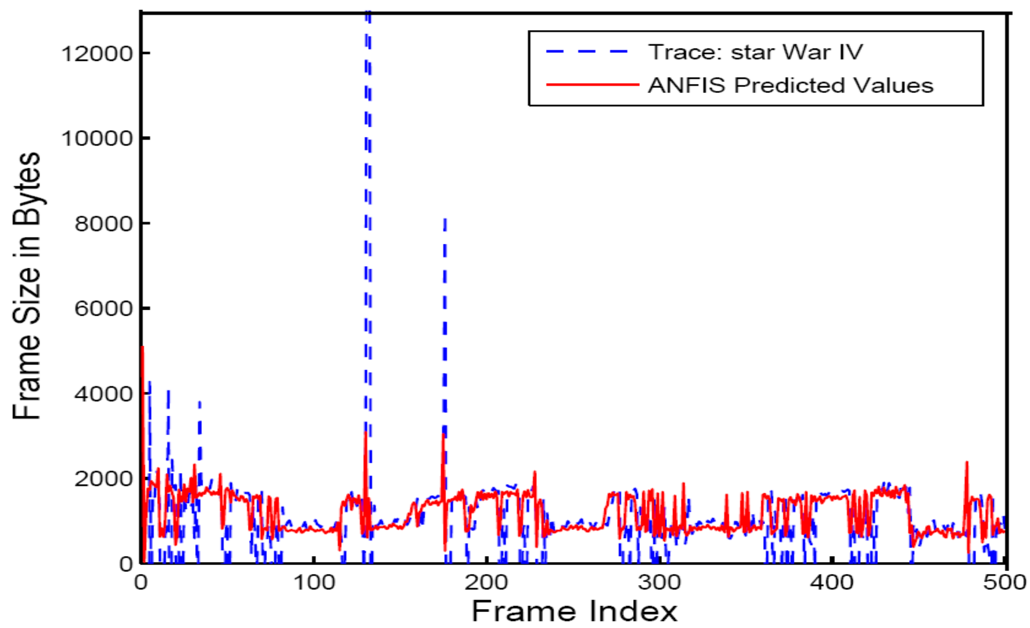


Figure 4-5: Predicted frame length for testing sets generated from movie Star War IV

sum of the square of the prediction errors and the sum of the square of the actual values. It can be expressed as

$$RMSE = \frac{\sum (V_A - V_P)^2}{\sum (V_A)^2} \quad (4\text{-}3)$$

Where V_A is the actual value and the V_P is the predicted value. Table 4-2 summarizes the performances in terms of RMSE.

Trace Source	RMSE in %
Scilence of lambs	2.08
Star Wars	2.14

Table 4-2: RMSE performance of ANFIS predicted frame length for training and testing input output pairs

V. Uplink MPEG Scheduling

A. Introduction

IEEE 802.16 MAC is the connection oriented protocol. For uplink data transmission, all applications should establish a connection with BS as well as should classify itself to one of the provisioned QoS classes in and follow bandwidth management procedures in advance. Bandwidth management in IEEE 802.16 is hierarchical in nature [3]**.** Hierarchical bandwidth management simply implies that the allocation is TLBA, BS-level allocation and SS-level allocation. In GPSS[1] mode of BS-level allocation, even though the bandwidth request is made per connection basis, grant is offered per SS basis. To redistribute the granted bandwidth to multiple connections belonging to a SS, class based schedulers for each QoS classes are implemented locally in each SS. Selection of scheduling algorithm for each QoS classes depends on the QoS requirement of particular type applications.

In this work, we suggested a scheduler at SS for rtPS class in IEEE 802.16 based MAN, assuming all the video applications consists only MPEG video frames. In MPEG encoded video streams, packets of the video stream do not contribute evenly to the video quality at a receiver side and packet is useful to the receiving side only if (a) packet arrived prior to its delivery deadline, and (b) all the previous packets required for the correct decoding were already received. Conventional sequential sending schedulers like EDF (Earliest Deadline First) can not guarantee (b). Some opportunistic approaches are needed in the scheduler to guarantee the constraints put forward by (b). In this work, we employed prioritization based approach. For every frame of GOP, transmission and selective drop order is determined. Determination of scheduling and

[1] Currently, GPSS is the only bandwidth grant mode in IEEE 802.16 since GPC is eliminated from the standard.

selective drop order of frames is in accordance to the importance of these frames to other frames for being correctly decoded. In other words, prioritization value assigned to the frames is proportional to their importance of being present at receiver side during decoding. We also coupled other network resource factor like queue length and network performance factor like PLR to obtain a scheduling index at every scheduling epoch. Scheduling will be performed with reference to that index.

B. Proposed Uplink MPEG Scheduler

1. Prioritization Scheme for MPEG Frames

Encoding is must for the transmission of large data content. So because of encoding the original data content, data content will be interpreted in different way than originally it was meant. There could possibly rearrangement of the content with some fraction containing much information while other with less information. In such condition, in wireless domain it is wise to save part of the encoded video which carries more information if any loss to occur. So to derive a new prioritization scheme for MPEG encoded video, in the next section, briefly traits of it are analyzed.

a. MPEG Traits

In MPEG coded video streams, [$F_1F2_1F_3$……F_N] where F_i is a frame, there are three types of frames, intra coded (I), inter coded (P), and bidirectional coded (B). These frames are arranged in a periodic GOP with parameter (N, M). For e.g. GOP sequence, IBBPBBPBBPBB has GOP parameter of (12, 3) as shown in Fig. 5-1. I frames are the only frames in a GOP which are independently decoded. Their size is large compared to other frame types. P frames are forward predicated from an earlier frame, which could be an I frame or a P frame. P frame's size is roughly half of the I frame. The B frames are bidirectionally interpolated frames from earlier or later I or P frames. B frame’s size is around one fourth of I frame. I frame provides stop point for the error propagation while P and B frames allow good compression.

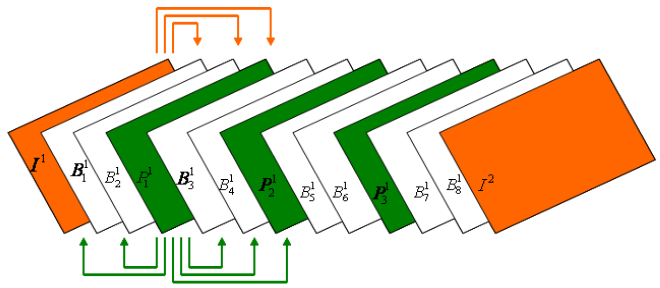


Figure 5-1: Inter frame dependent GOP of MPEG encoded video

b. Effect of Frame loss on MPEG Video Quality

Not all packet losses create the same visual impact, different realizations of video content and packet loss may lead to vastly different visual quality [22]. How the effect of lost frame propagates along the subsequent frames determine the visual quality. Because of the use of intra coding for I, and bidirectional linear interpolation to other frame types, effect of loss of a frame depends on its type. Hence, effect of loss of frame differs with the frame type. To explore the effect, we carried a mathematical analysis with the notations and definitions as summarized in the table 5-1. The whole probabilistic analysis is carried using two simple rules of probability. Firstly, multiplication of probabilities is carried to find the chance that several events occur in sequence or at the same time. Secondly, with the information of the product of the probabilities, expected value is derived.

Let us start with the mathematical characterization of the number of I, P and B frames. For a GOP with parameter (N, M) number of respective frames within a GOP can be represented in equation as follows:

Notation	**Definition**
N	GOP length, I-I frame distance in number of frames
M	I-P (first p frame in GOP) distance or P-P distance
N_I	Number of I frame
N_P	Number of P frames in a GOP
N_B	Number of B frames in GOP
N_{BP}	Number of B frames between successive reference frames
$P_{L,I}$	Probability of loss of I frame
$P_{L,B}$	Probability of loss of B frame
$P_{L,P}$	Probability of loss of P frame
$\overline{L}_I$	Expected number of lost frames in GOP because of the loss of I frame
$\overline{L}_P$	Expected Number of lost frames in GOP because of loss of P frame
$\overline{L}_B$	Expected Number of lost frames in GOP because of loss of B frame

Table 5-1: Definition of the notations used for mathematical analysis

$$N_I = 1 \tag{5-1}$$

$$N_P = \left(\frac{N}{M} - 1\right) \tag{5-2}$$

$$N_B = (1 + N_P) \times N_{BP} \tag{5-3}$$

So with this information we can say that total number of frames per GOP is

$$N = N_I + N_P + N_B \tag{5-4}$$

Effect of loss of I frame propagates to all frames. Hence expected number of loss of frames because of the loss of I frame can be represented as

$$\overline{L_I} = P_{L,I} \times \left(N_I + N_P + N_B\right) \qquad (5\text{-}5a)$$

$$\overline{L_I} = P_{L,I} \times N \qquad (5\text{-}5b)$$

Unlike the single I frame in a GOP, usually there are multiple P frames as suggested by equation 5-2. So among the P frames, the effect of loss of j[th] P frame while other preceding j-1 P frames received well is derived to be

$$\overline{L_P} = \sum_{j=1}^{N_P} N_{BP} P_{L,P} \left(1 - P_{L,I}\right)\left(1 - P_{L,P}\right)^{j-1}$$
$$+ \sum_{j=1}^{N_P} \left[M\left(N_{BP} + 1 - j\right) P_{L,P} \left(1 - P_{L,I}\right)\left(1 - P_{L,P}\right)^{j-1}\right] + P_{L,I} \times N \qquad (5\text{-}6)$$

Substituting equation 5-2 on equation 5-6, it yields,

$$\overline{L_P} = \sum_{j=1}^{\frac{N}{M}-1} \left(M - 1\right) * P_{L,P} \left(1 - P_{L,I}\right)\left(1 - P_{L,P}\right)^{j-1}$$
$$+ \sum_{j=1}^{\frac{N}{M}-1} \left[M\left(\frac{N}{M} - j\right) P_{L,P} \left(1 - P_{L,I}\right)\left(1 - P_{L,P}\right)^{j-1}\right] + P_{L,I} \times N \qquad (5\text{-}7)$$

Further simplification of equation 5-7 results an expression for the expected loss of frames in a GOP because of loss of P frame. This can be expressed as

$$\overline{L_P} = P_{L,P}(1 - P_{L,I}) * \sum_{j=1}^{\frac{N}{M}-1} \left[N - 1 - M\left(j - 1\right)\right](1 - P_{L,P})^{j-1} \qquad (5\text{-}8)$$

The expected loss of frames per GOP because of the loss of I and P frames is presented in Fig 5-2 and Fig 5-3.

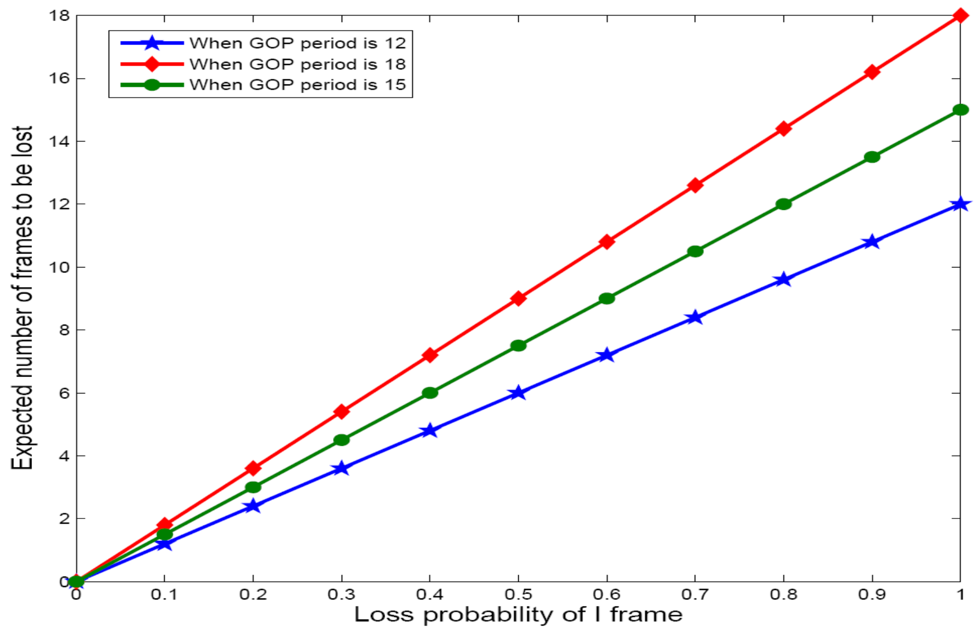


Figure 5-2: Expected number of frame loss in a GOP when I is lost

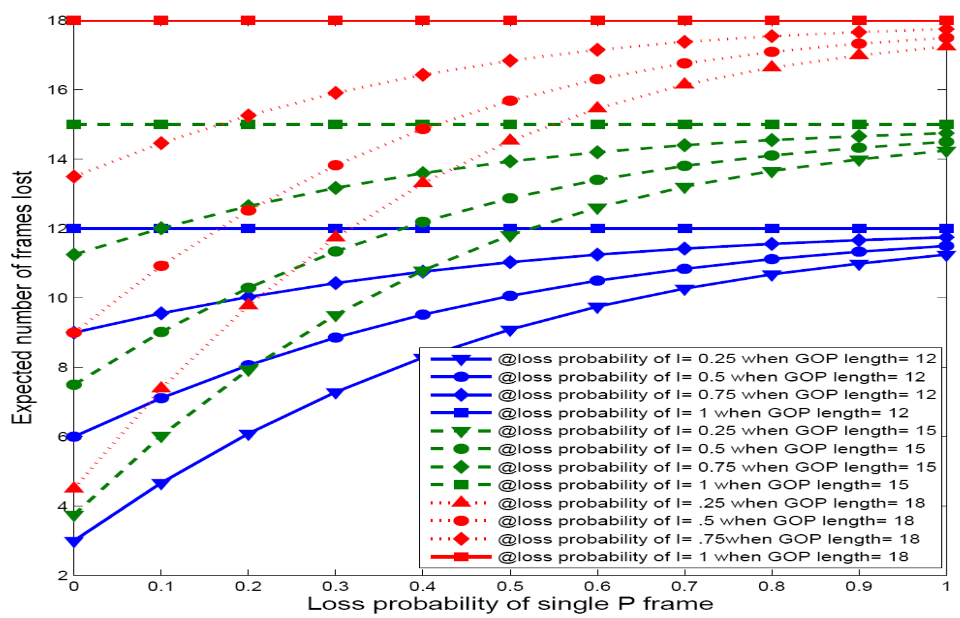


Figure 5-3: Expected number of frame loss in a GOP when P is lost with certain loss probability of I

Expected loss of B frames is not worthy to calculate since the effect of loss of B frames doesn't propagate to other frames. It is well understood that the effect of loss for I, P and B varies sharply. So I leads the importance among all other frames, P follows I and B remains in the last. This is the fact which drives us towards the prioritization of frames within a GOP.

c. Proposed Prioritization Scheme

Inter frame dependent frame loss analysis in the previous section makes us identify the importance of particular type of frame within a GOP. The importance order of frames is I, P, B in descending order. B frame needs both the preceding and the following P or I frame to be decoded. So we modified the frame order in GOP for transmission of MPEG video streams. In this modified transmission order all the frame(s) needed for decoding of another frame is(are) already available at receiver side at decoding time of that frame. This new importance based arrangement of frame will be like IPBBPBBPBBIB for GOP with structure IBBPBBPBBPBB with GOP parameter of (12,3). It could be a new transmission order if the resource available is enough to transport all the frames. However, if there is not enough resource to carry all the frames, then we should drop frame(s) which results least possible degrade in video quality. So to address the transmission order and the selective packet drop order in general case, we splitted MPEG encoded video as shown in Fig. 5-4 and priority weight assignment for special case of GOP of (12,3) is shown in Fig. 5-5. Each frames are arranged in different layers from layer 1, L_1 to layer $\frac{N}{M}+1$, $L_{N/M+1}$. For a video source with m number of GOPs, the bottom layer having highest priority for scheduling will have all I frames $(I_1^1, I_1^2, I_1^3 \ldots\ldots I_1^m)$. The next immediate layer above the bottom layer will have $(P_1^1, P_1^2, P_1^3 \ldots\ldots P_1^m)$. The layer for P is equal to the number of P frames in a GOP. The last layer for P will have the following set of frames $\left(P^1_{\left(\frac{N}{M}-1\right)}, P^2_{\left(\frac{N}{M}-1\right)} P^3_{\left(\frac{N}{M}-1\right)}, \ldots\ldots\ldots\ldots P^m_{\left(\frac{N}{M}-1\right)}\right)$.

The top layer with least transmission priority and maximum drop priority holds all B frames, $\left(B_1^1, B_2^1, \ldots B^1_{((1+N_p)*N_{BP})}, B_1^2, B_2^2, \ldots B^2_{((1+N_P)*N_{BP})}, B_1^m, B_2^m, \ldots B^m_{((1+N_P)*N_{BP})}\right)$.

In this structured layer architecture, transmission priority increases with decreasing layer index and dropping priority increases with the increasing layer index. For simplicity, we assigned equidistant priority weight to each layer. Scheduling and packet dropping equidistant priority weight for each frame inside any layer l is simply generated with the following relations respectively.

$$P_{SI}^l = 1 - \left(\frac{l-1}{L_{N_P} + 2}\right) \tag{5-9}$$

$$P_D^l = \left(\frac{L_{N_P} + 2}{1 - L_{N_P} + 2 - l}\right) \tag{5-10}$$

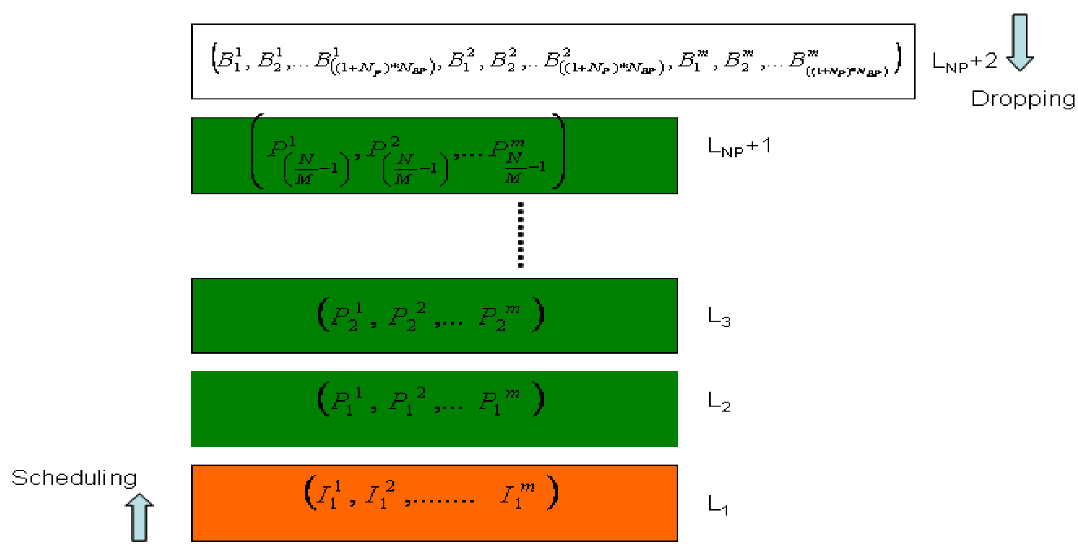


Figure 5-4: Prioritization scheme for MPEG frames

After packetization of the frames, multiple packets of a same frame will have the same priority weight as the frame from which they are fragmented. The priority weight for any packet is denoted as α_{ij} which simply corresponds that priority weight for j^th packet in i^th flow. α_{ij} is assigned a value at each scheduling epoch with reference to Eq.5-9.

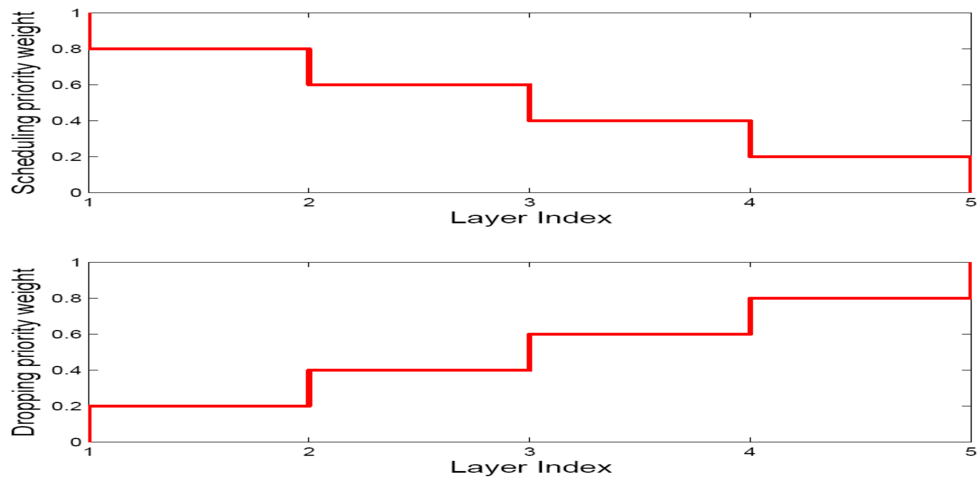


Figure 5-5: Scheduling priority weight and dropping weight versus layer index when GOP= (12, 3)

After Packetization of the frames, (as will be discussed in next section), packets of a frame will have the same prioritization value as the frame from where it is fragmented. The prioritization value for the packet is denoted as α_{ij} which simply means prioritization value for j^{th} packet in i^{th} flow.

2. Proposed Uplink Video Scheduler

In this proposed scheduler we employed network based approach to protect more important video data [2] for being lost. In the employed approach, prioritization based packet scheduling in accordance with the importance of the video packet for correctly decoding other video packets in the receiver side is considered. Moreover, additional buffer management incorporating selective drop of least important video packet is also adopted. In this proposed scheduling algorithm, scheduling is performed with the following sequential actions.

[2] Video data corresponds to MPEG frames through out this thesis.

1. At first scheduling index, SI of every active flow[3] is calculated at scheduling time t.

$$SI(t) = \max_i \left[NPLR_i(t) * Q_i(t) * \sum_{j=1}^{P_{total}^i} \alpha_{ij}(t) \right] \quad (5\text{-}13)$$

Where $Q_i(t)$ is the current instantaneous queue length and $\alpha_{ij}(t)$ is the prioritization weight for the j[th] packet in a queue of i[th] flow . $NPLR_i(t)$ is the normalized packet loss ratio of i[th] flow when the maximum allowed packet loss ratio is PLR_{MAX} . This normalized packet loss ratio can be expressed as

$$NPLR_i(t) = \left(\frac{PLR_i(t)}{PLR_{MAX}} \right) . \quad (5\text{-}14)$$

P_{total}^i is the number of total packets in i[th] flow. Provision of HARQ (Hybrid Automatic Repeat reQuest) in IEEE 802.16 allows us to get the feedback of the transmitted packets about success or failure of transmission. Hence packet loss ratio can be obtained as

$$PLR_i(t) = \left(\frac{N_i^d}{N_i^d + N_i^s} \right), i \in K \quad (5\text{-}15)$$

Where K is the total number of nodes in a SS. N_i^d and N_i^s are the number of dropped packets from the i[th] flow's queue in SS and the number of successfully received packets at BS respectively from i[th] flow.

[3] Flow and connection are used interchangeably in this thesis.

2. All the queues are sorted logically in the descending order of the scheduling index.
3. All the packets inside the queue are sorted as per the priority value as described in previous section (as shown in Fig 5-4). For the packets which can't be sorted only with the priority value, they are sorted with the additional timing information. For the packets with the same priority weight, their scheduling order is determined by the waiting time in the queue. The packet which waited long is chosen first among the packets with identical priority weight.
4. The HOL (Head of line) of the upper queue is transferred.
5. Bandwidth remained is calculated

$$BW_{REM} = BW_{MAX} - HOL_{SIZE} \tag{5-16}$$

   Where, BW_{REM} is the remained size of the data slots, BW_{MAX} is the size of total available data slots per SS and HOL_{SIZE} is the size of the transferred HOL packet.
6. If BW_{REM} is larger than the HOL of next immediate queue, transfer HOL.
7. Repeat step 4 and 5 iteratively unless BW_{REM} is less than HOL size of packet which is in turn for transmission.
8. All the packets which are not scheduled in the current scheduling time are backlogged if their deadline will not expire within next scheduling period.

a. Performance Evaluation

In IEEE 802.16 based MAN, if we consider an uplink topology, there could be multiple SSs with multiple connections per SS belonging to different QoS classes. But in this paper we are simply interested to study the performance of the scheduler for only one QoS class, i.e rtPS. This proposed scheduler can be thought as UDS-rtPS (User Data Scheduler for Real Time Polling Services). This proposed scheduler is to be resided in every SS along with other schedulers for other QoS classes. To study the performance of this scheduler, we simply considered an environment with single SS with multiple video connections and single BS as shown in Fig. 5-6 with dotted ellipse.

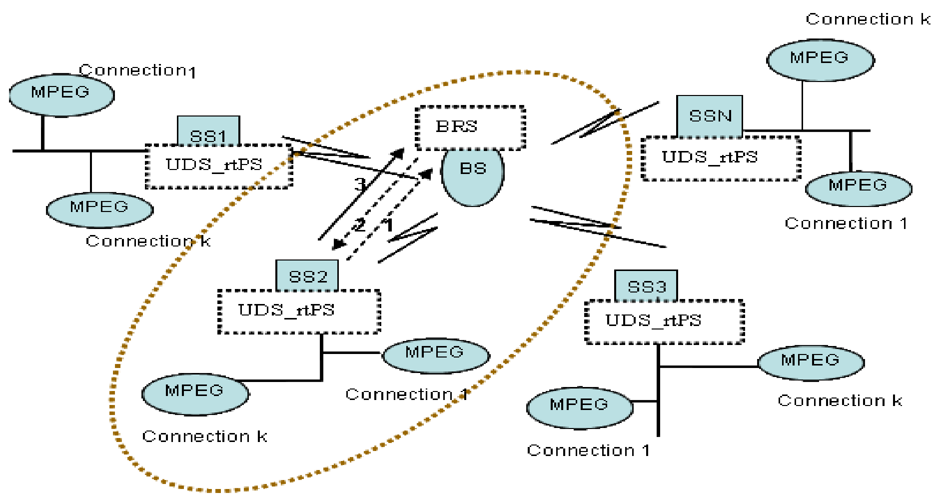


Figure 5-6: Considered topology for WiMAX uplink

With the tabulated simulation parameters and assumption of bandwidth-limited and error free channel, we analyzed the performance in terms of PLR versus concurrent number of connections per SS using Matlab.

Since we used trace as input, input to scheduler are deterministic since we considered the deterministic arrival pattern. So even in the repetition of the simulation of the scheduler we will receive the same output. It is because there is no randomness in both input and scheduler. So in this case the results obtained even with one execution are absolute results and accurate as well [23].

It is assumed that the bandwidth allocation to the connections within a SS is made as per GPSS mode by BRS (Bandwidth Request Scheduler). With the

Parameters	Values
No. of BS	1
No. of SS	1
Connection in SS	1-15
Scheduling period	100 ms
D_{MAX}	120ms and 240 ms
$BW_{MAX\ Size}$	5000 and 10000 Bytes/scheduling interval
MTU	1000 Bytes
Video source Parameters	**Values**
Encoding	MPEG
GOP	(12,3)
Frame rate	25 fps
Mean frame size	1+e03 Bytes
COV of frame Size	0.66
Minimum frame Size	26 Bytes
Maximum frame Size	9370 Bytes
Minimum bit rate	2.8+e05 bps
Peak bit rate	1.9 +e06 bps

Table 5-2: Simulation parameters for MPEG scheduler

allocated bandwidth by BRS, UDS-rtPS schedules MPEG frames from k different connections. The maximum available bandwidth from SS to BS is considered to be BW_{MAX} per scheduling interval. In the SS there are 1-15 nodes which join SS in the interval of 100 ms. Every connections connected with the SS has MPEG video to be transmitted to BS. Source video is truncated trace of Star War IV [21]. In the truncated trace there are altogether 50 frames with frame rate of 25 fps (frames per second). The parameters considered for the simulation are summarized in table 5-2. Generally the frame size of the video frames is larger than the MTU. Hence, the frames in the GOP of MPEG coded videos are packetized into IP packets or ATM (Asynchronous Transfer Mode) cells if they are to be transmitted in WiMAX network since the available CS in WiMAX is either ATM based or IP based. For this performance analysis, we considered the convergence sub layer to be IPCS[4]. So, the frames in the GOP of MPEG coded videos are packetized into IP packets with maximum size of MTU (Maximum Transfer Unit) Bytes. In the MPEG trace of Movie Star Wars IV, I frame is typically 7 times larger than the usual B frame and generally does not fit into a single packet. Therefore I frame should be divided into packets by using either scan order slices (preferred) or the GOB (Group of Blocks) mechanism as in H.26x. The packetizing module considered in this paper as shown in Fig. 5-7, first MPEG frames are changed into RTP packets then into UDP packets and finally to the IP packet as compatible to the MTU for IPCS. So our payload will be as per IPv4 specification. The IPv4 payload can be a maximum value of

$$MTU = 2^{MAC_{bits}} - Header_{size} - CRC_{size} \tag{5-11}$$

Where, MAC_{bits} is the number MAC frame bits, $HEADER_{size}$ is the MAC header size in Bytes and CRC_{size} is the size in bytes that cyclic redundancy check field could occupy. For example if MAC_{bits} is 11 bits, $HEADER_{size}$ is 6 Bytes and CRC_{size} is 4 Bytes then MTU is 2038 Bytes. If we neglect the packing option simply MAC overhead can be calculated as

$$OH_{MAC} = \left(\frac{HEADER_{size} + CRC_{size} + subHEADER_{size}}{Payload} \right) \tag{5-12}$$

As suggested by equation 5-12 we can witness the decrease in overhead for increasing payload in Fig 5-8. In this work we took MTU as 1000 Bytes which only results around 1% overhead.

[4] The IPCS is a term used in IEEE 802.16 to describe a process which allows IP datagrams to be directly carried in the 802.16 PDU.

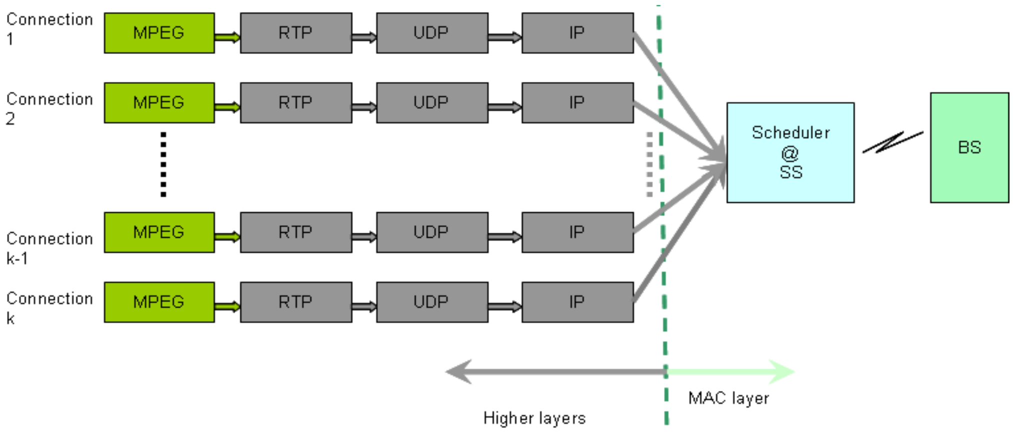


Figure 5-7: Packetization scheme for the video frames

However larger payload most probably suffered with high packet error rate. So as a tradeoff, we took the MTU as 1000 Bytes while default MTU size is 1440 Bytes.

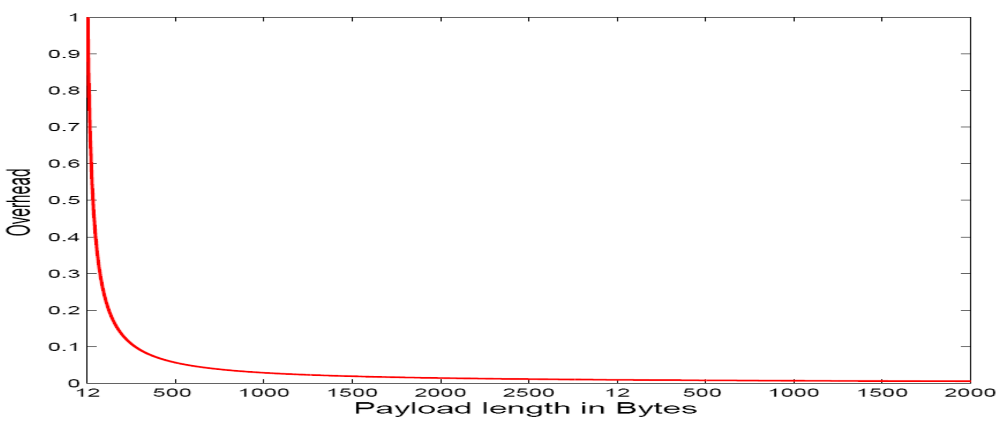


Figure 5-8: Payload versus overhead

We analyzed the performance with variation in parameters in two dimensions. In the first one, available resource is varied and in second, delay requirement is varied. In the current work, we made variation in the value by doubling the original values. Doubled values are referred as the high values while original values are low values. This give rise to the four cases as stated below:

(a) Low BW_{MAX} and Low D_{MAX},(b) Low BW_{MAX} and High D_{MAX}, (c) High BW_{MAX} and Low D_{MAX}, (d) High BW_{MAX} and High D_{MAX}

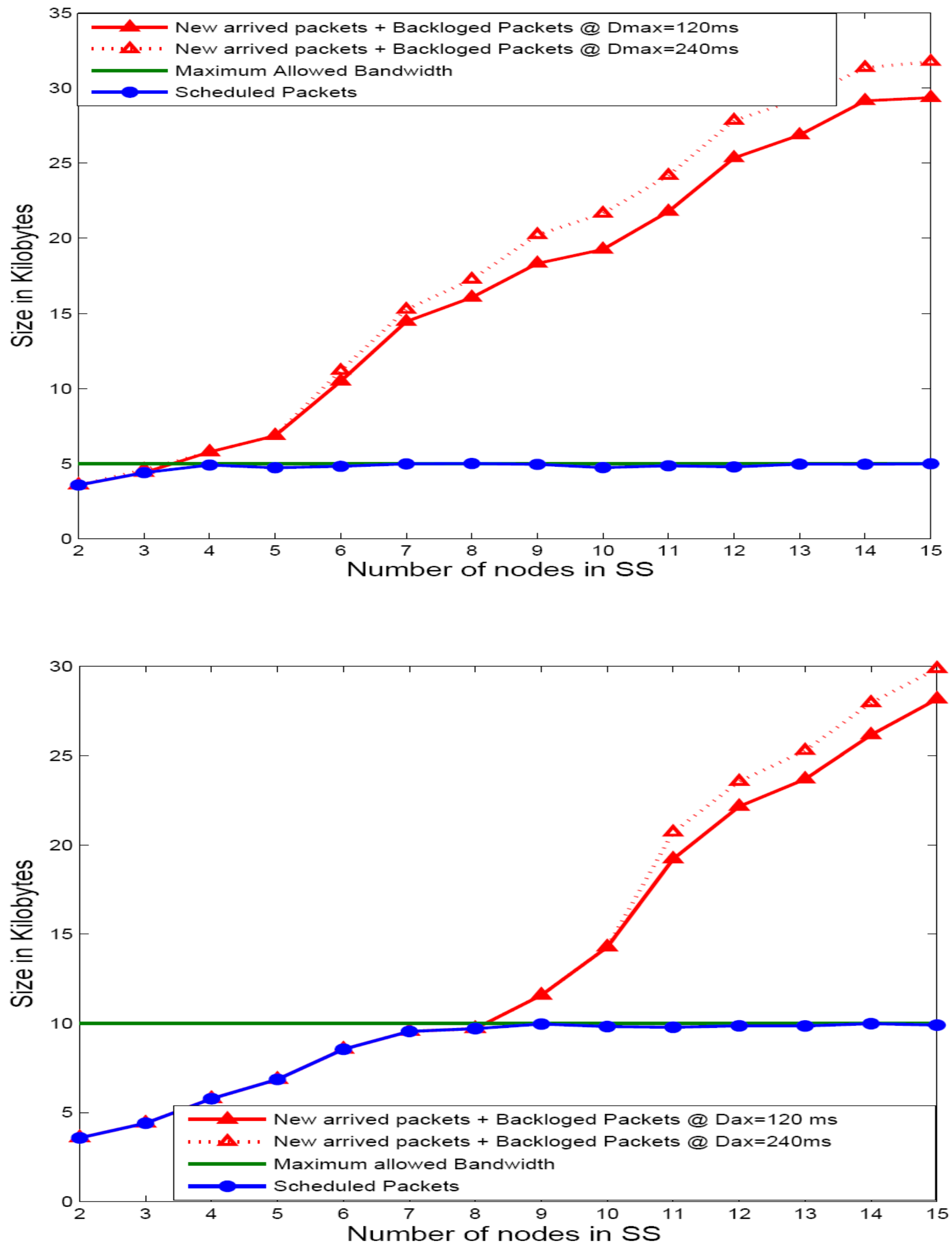


Figure 5-9: Instantaneous offered load and scheduled load when BW_MAX is 5000 (up) and 10000 Bytes/scheduling epoch (down)

In Fig. 5-9, the instantaneous offered load seems different for low and high BW_{MAX} even though all source parameters, characteristics and behavior are kept identical. The difference is observed because of the difference in the amount of backlogged packets.

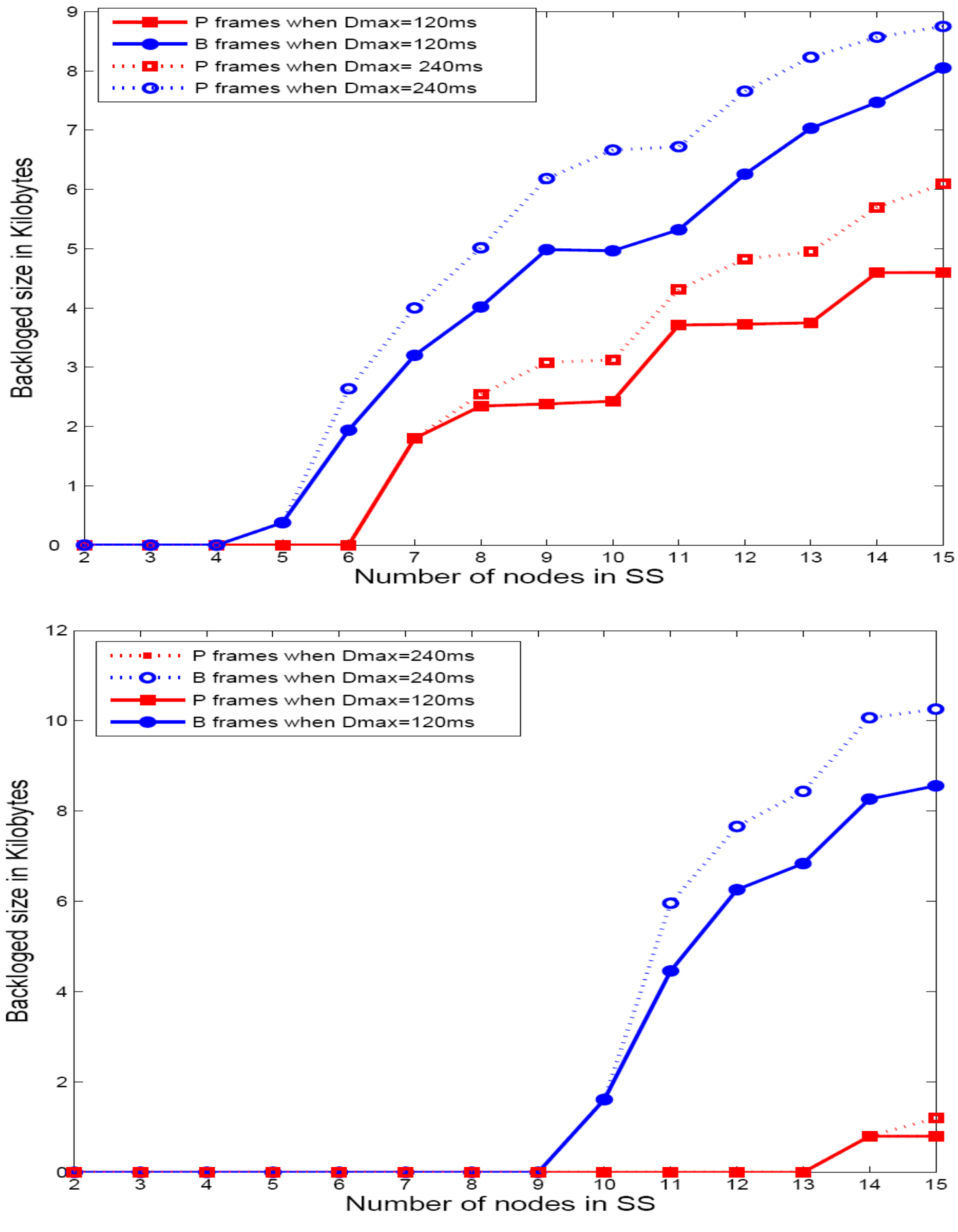
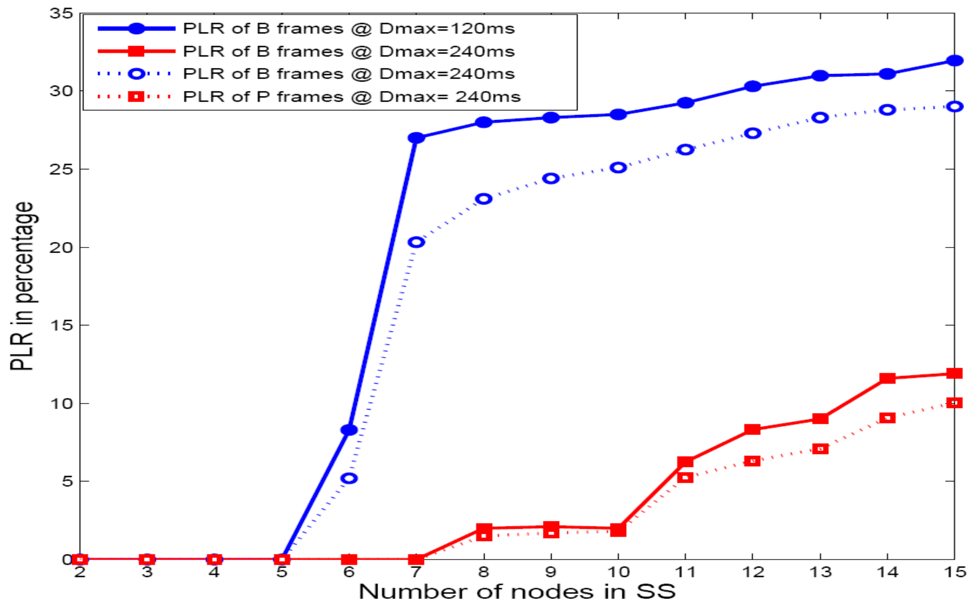


Figure 5-10: Backlogged size when BW_MAX is 5000 (up) and 10000 Bytes/scheduling epoch (down)

Unscheduled packets whose deadlines are not expired and will not be expired up to next scheduling period are backlogged. These backlogged packets are added to the fresh new arrival and hence the instantaneous offered load seems higher in the low BW_{MAX} case. For example, in Fig. 5-9 up, when the number of nodes is 9, instantaneous offered load is about 19 KB in the first experimenting scenario. This 19 KB is constituted of 8 KB of backlogged packets (5 KB of B and 3 KB of P) and rest 11KB of the new arrived packets. In the later two experimenting scenarios, at the same number of nodes, there is only 11 KB of instantaneous offered load with all new arrived packets and nil backlogged packets from P and B frames. The increase in backlogged size, which eventually increases the instantaneous offered load at the scheduling epoch for all experimenting scenarios, is shown in Fig. 5-10. In experiment scenarios c and d, backlog starts with less than half number of nodes from where backlog started for both P and B frames respectively for experimenting scenarios a and b. It is note worthy to say that in all cases there is no backlogged I packets. For the low and high BW_{MAX}, there is presence of backlogged packets of B frames when number of connections in SS reaches 4 and 9 respectively. Similarly when the nodes number is greater than 6 and 13, there is presence of backlogged packets of P packets.

The PLR for each frame types is shown in Fig. 5-11. From the quick observation, it can be verified that the PLR performance is good for the later two experimenting scenarios because the resource available for them is double than the first two scenarios. It is still note worthy to observe that PLR of I frames is nil in all experimenting scenarios even though there is high PLR of B frames and small loss of P frames. If some I frames were lost, PLR of P and B frames would be more because P and B frames of a GOP which are remained to be scheduled should also be dropped after the loss of I. Hence saving I frames from being lost is a remarkable superiority of this proposed scheme. In Fig. 5-11, even after increase in PLR of B frame, P frames' nil PLR is maintained to more number of nodes and nil PLR for I frame is maintained through out. It is also note worthy to



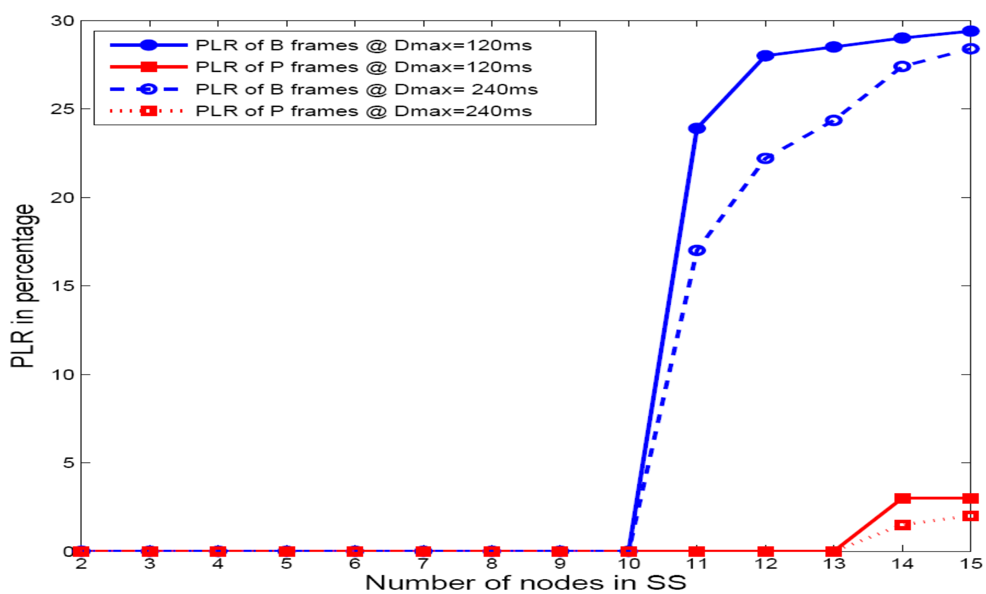


Figure 5-11: PLR in percentage when BW_MAX is 5000 (up) and 10000 Bytes/scheduling epoch (down)

see that increase in the value of the D_{MAX} decreases PLR. In other words, if D_{MAX} is increased more connections can be adopted in the SS for the fixed PLR requirement.

VI. Conclusion and Future Works

The contributions we made in the uplink video transmission in this carried research were presented in separate chapters. So the conclusions of the works are also presented chapter wise.

In chapter three, we developed a model for contention based BW-REQ mechanism for WiMAX based system where there is already a differentiation of connections to different QoS classes. This contention based access is carried with the prioritized access probability assignment for every connections belonging to considered QoS classes. Game theoretic access probability assignment is formulated to assign access probabilities. We analyzed the contention based BW-REQ with connections belonging to two different QoS classes, namely rtPS and nrtPS. In the WiMAX standard, there is no permission for the contention based BW-REQ access for the rtPS service with the assumption that it will make rtPS connections suffer long access delay, which is strongly undesirable for the rtPS connections. Our approach provides the almost perfect success probability with the access delay of nearly one frame unit. This is almost equivalent that they are polled to make bandwidth request. So we can conclude that this GTAP-BRM can be used for all classes in WiMAX where either polling or contention based access are specified to request bandwidth. This GTAP-BRM can be used for more number of QoS classes as well. The inclusion of more QoS classes, however, will add more complexity for finding NE.

In chapter four, we shortly discussed about the importance of online traffic prediction based bandwidth allocation for VBR video application in frame work of WiMAX. We suggested a lower bound value for nominal polling interval for which online prediction based allocation will be effective. The more accurate prediction will be required in such prediction based works. We modeled an ANFIS based system for prediction of frame length and from the performance analysis it is found to be good enough with the RMSE of 2.08% and 2.14% for training and testing sets extracted from two different MPEG coded movies. On the other hand, since this model is dealing with real trace rather than empirical models, its performance is closer to the real world approximation. With this small RMSE, we anticipate that proposed prediction model can be recommended for being used in the prediction based resource allocation which will be the next immediate work we are supposed to undertake.

In chapter five, we proposed a specific scheduler for scheduling MPEG encoded videos in IEEE 802.16 MAN for uplink. The prioritization scheme suggested in this chapter modifies the transmission order of the frames for making them available in the receiver for accurate decoding of other frames. The performance of the proposed scheduling algorithm was observed in terms of PLR with the increasing number of concurrent connections in the SS with certain threshold of maximum allowable bandwidth. The performance study carried for four different scenarios permuted from low and high values for D_{MAX} and B_{MAX} confirm that this scheduler yields better PLR performances for important frame types, I and P, in expense of higher PLR for least important B frames.

The work we carried in this thesis, leads us to the comfortable position to realize two new uplink video transmission schemes for WiMAX, a dominant name of BWA.

1. Integration of extended GTAP-BRM with the bandwidth grants information with the proposed scheduler.
2. Integration of online traffic prediction module with legacy contention-free polling based bandwidth request grant mechanism with the proposed scheduler.

The performance study of above stated integrated schemes in 1 and 2 will be the part of our future work to be carried shortly.

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본인이 저작한 위의 저작물에 대하여 다음과 같은 조건아래 조선대학교가 저작물을 이용할 수 있도록 허락하고 동의합니다.

\- 다 음 -

1. 저작물의 DB구축 및 인터넷을 포함한 정보통신망에의 공개를 위한 저작물의 복제, 기억장치에의 저장, 전송 등을 허락함
2. 위의 목적을 위하여 필요한 범위 내에서의 편집·형식상의 변경을 허락함. 다만, 저작물의 내용변경은 금지함.
3. 배포·전송된 저작물의 영리적 목적을 위한 복제, 저장, 전송 등은 금지함.
4. 저작물에 대한 이용기간은 5년으로 하고, 기간종료 3개월 이내에 별도의 의사 표시가 없을 경우에는 저작물의 이용기간을 계속 연장함.
5. 해당 저작물의 저작권을 타인에게 양도하거나 또는 출판을 허락을 하였을 경우에는 1개월 이내에 대학에 이를 통보함.
6. 조선대학교는 저작물의 이용허락 이후 해당 저작물로 인하여 발생하는 타인에 의한 권리 침해에 대하여 일체의 법적 책임을 지지 않음
7. 소속대학의 협정기관에 저작물의 제공 및 인터넷 등 정보통신망을 이용한 저작물의 전송·출력을 허락함.

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