A Quality-Driven Cross-Layer Solution for MPEG Video Streaming Over WiMAX Networks

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***Abstract*—Extensive efforts have been focused on deploying broadband wireless networks. Providing mobile users with high speed network connectivity will let them run various multimedia applications on their wireless devices. Satisfying users with different quality-of-service requirements while optimizing resource allocation is a challenging problem. In this paper, we discuss the challenges and possible solutions for transmitting MPEG video streams over WiMAX networks. We will briefly describe the MPEG traffic model suggested by the WiMAX Forum. A cross-layer solution for enhancing the performance of WiMAX networks with respect to MPEG video streaming applications is explained. Our solution uses the characteristics of MPEG traffic to give priority to the more important frames and protect them against dropping. Besides, it is simple and compatible with the IEEE 802.16 standards and thus easily deployable. It is shown that the proposed solutions will improve the video quality over WiMAX networks.**

***Index Terms*—Cross layer, MPEG, quality-of-service, video, wireless, WiMAX.**

I. Introduction

THE excessive demand for ubiquitous broadband wireless access has attracted tremendous investment from the telecommunications industry in the development and deployment of WiMAX networks. The WiMAX technology is promising to provide broadband wireless access to mobile users in the near future. It is expected that video streaming will be a very attractive application for the rapid deployment of WiMAX networks. The stringent quality-of-service (QoS) requirements including high bitrate and low latency are some of the challenges the service providers and network designers are confronting. Furthermore, the popularity of many online video servers such as YouTube will encourage an increasing number of users to watch video clips on their mobile devices.

The scarcity of available bandwidth in wireless networks has called for efficient resource management. WiMAX networks are based on the IEEE 802.16 standards which have defined different QoS classes to support a broad range of applications with varying service requirements. The IEEE 802.16 standards provide true QoS classes for different types of applications. As a result, in WiMAX networks, each traffic flow is mapped into an appropriate service class based on its service requirements and the user's service level agreements (SLAs). Selecting appropriate service classes with proper parameters to support the required QoS while not wasting the scarce resources is the key challenge that we address in this paper. We study the traffic characteristics of video streaming applications and will show that an application driven, traffic aware service classification will provide the WiMAX subscriber stations (SSs) with better video quality.

The importance of efficient resource management has prompted a keen interest in the research community on supporting video streaming applications in wireless networks. Reference [1] has reviewed the challenges of video streaming in wireless networks. It has also proposed a network adaptive rate control and cross-layer design for enhancing the overall received video quality. Many other research works have also considered feedback based video rate control [2]–[6]. In most of the rate adaptive methods, the server receives some information such as the available bandwidth, loss rate, buffer size at the receiver, or the end-to-end delay to adapt to the optimum video coding rate. One of the main drawbacks for the rate adaptive methods is caused by the channel variations in the wireless networks. Owing to the fast variation in the wireless physical channels, the adaptation methods may not be able to track the fast changes in radio channel conditions and adapt to the optimum rate accordingly. Furthermore, selecting the appropriate rate increases the computational complexity at the video server which can result in overloading the video streaming servers. Moreover, sending feedback is not a feasible option in some multicasting applications such as IPTV or MobileTV. In such cases, a video server transmits the same content to multiple receivers with different physical channels. The heterogeneity of receivers in these applications makes it very complicated for the server to attain the flexibility and sustain the efficiency.

In order to reduce the complexity at the server side and support various types of clients, scalable video coding (SVC) has been introduced in [7]. The goal of this method is to encode high quality video streams into some groups of bit streams including one base sublayer and multiple enhancement sublayers. All clients register to receive the base sublayer. The addition of enhancement sublayers improves the video quality. Thus,

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clients select the number of enhancement sublayers to receive based on their network connection and the availability of resources [8].

The challenging and crucial problem of video streaming over WiMAX networks has attracted many researchers. Reference [9] presents an adaptive SVC approach for streaming video on demand to the subscriber stations. Different WiMAX network architectures and cross-layer solutions for supporting the broadcasting and multicasting applications have been introduced [10], [11]. They take advantage of the broadcasting capabilities of WiMAX wireless medium and leverage this feature to deliver video multicasting applications such as IPTV to WiMAX customers. A channel based, rate adaptive solution for video streaming in WiMAX network has been introduced in [12]. Reference [13] illustrates an active frame dropping approach for streaming real-time video over IEEE 802.16 networks. In this active dropping approach, the base station drops a frame if it does not have enough confidence about successfully delivering a video frame within the application delay limit. The concept of active dropping at the video server for video streaming has been employed in different research works. A frame discarding solution based on the packet lifetime is introduced in [14]. In this method, frames that cannot meet the deadline are dropped by the video server or the intermediate routers. Reference [15] explains a priority based frame dropping algorithm. In response to a temporary bandwidth reduction, the video server selectively drops least effective frames. In this work, the authors have considered an MPEG video streaming system in which the video server determines the priority of each frame based on the frame type.

In this paper, we propose a cross-layer design to enhance the quality of MPEG video streaming for the end users in WiMAX networks. Our solution uses the characteristics of MPEG traffic to give priority to the more important frames and protect them against dropping. Unlike the approach proposed in [15], we do not increase the complexity of video servers since the frames are dropped at the BS. Moreover, the complexity of the BS is not increased as well. In fact, our proposed scheme is capable of being used by any WiMAX certified BS. In our method, we do not send real-time feedback to the video server. Thus, a video server will be able to support multiple clients simultaneously, and this makes our solution flexible for multicasting applications as well.

The outline of the rest of the paper is as follows. Section II explains the video streaming traffic model used in our research. In Section III, we provide an overview of the quality-of-service support in WiMAX networks. Section IV explains our proposed solution for video streaming in WiMAX. We will present our simulation results in Section V. Concluding remarks are given in Section VI.

II. Video Streaming Traffic Model

The transmission of digital video over broadband communication networks is an important service. However, the existence of different video generating applications along with the availability of numerous networking technologies with different service features have made this an extremely challenging problem. Providing the required quality-of-service to the end users is a difficult problem requiring in-depth understanding of the video traffic characteristics.

The variety of video applications and networking technologies has called for implementation of a variety of video coder and decoder (codec) standards. These standards may be deployed in a broad set of applications ranging from low bitrate video conferencing to high bitrate movie streaming. However, the efficiency of these codecs is different for different applications. Valid traffic models accounting for the key video characteristics are required to investigate how best to transport video from different applications and codecs over different networks.

Moving picture expert group (MPEG) has a series of advanced video compressing standards, of which MPEG-2 and MPEG-4 are the most pervasive ones, and the latter is the latest and the most advanced one. All these standards rely on removing the redundant information of each frame by predicting the changes between subsequent frames. The idea of prediction is based on the fact that consecutive scenes have few differences and the information in their pictures is highly correlated. By coding the small differences between the scenes, much less data needs to be transported and thus achieving data compression.

MPEG-4 encodes the input video into a sequence of frames called group of picture (GOP). The number of frames in each GOP is typically constant. It is possible that an MPEG frame is fragmented into multiple IP packets when transmitted over an IP network. The MPEG-4 encoder divides each scene of the video into a number of consecutive GOPs. The number of GOPs generating a scene is a function of the scene complexity and compression ratio. There has been extensive research work on modeling the MPEG video traffic. Reference [16] has separated the video traffic into I, P, and B frames. The authors have modeled each type of frame separately and have also provided a model for the combined traffic. Reference [17] has also suggested a traffic model by separating the MPEG frame into three different types. The WiMAX Forum has adopted the traffic model proposed in these papers and has given the parameters determined from empirical video traces for different video applications such as video conferencing and video streaming [18]. A rather comprehensive work on modeling MPEG video traffic is given in [19].

As mentioned above, the MPEG coded videos are composed of three different frame types, i.e., I, P, and B. I (intra coded) frames are single still images used as the reference frame in each GOP. I frames are used for synchronization of all frames in a GOP. If a GOP is lost or corrupted, the next GOP will be built based on its I frame which is not coded using any other frames. P (predicted) frames are built by predicting the changes from the closest match in the preceding I or P frames of their GOP. However, B (bi-predictive) frames use previous I or P frame and the next P frame to predict the changes in the picture. Thus, the B frames are used to predict both the backward and forward changes in the motion. Based on these definitions, it is understood that these frames are interrelated, and some P and B frames are derived from each I frame in a GOP. Similarly, some B frames are also derived from each P frame. Therefore, loss of I or P frames will affect some other frames in their corresponding GOP and this will degrade the perceived quality. We will elaborate on this problem in Section V.

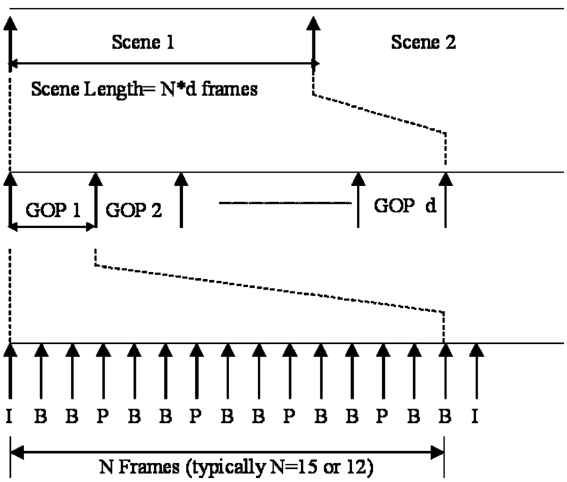


Fig. 1. Source model.

Although not required by the standard, MPEG encoders usually use a fixed pattern of frames in GOPs. The GOP pattern indicates the number of frames in each GOP, and their permutation order. In a regular GOP pattern, a GOP begins with an I frame and the number of B frames between I and P frames or between two P frames is constant. Such regular GOPs can be defined by two parameters: the I-to-I distance "N", and the I-to-P or P-to-P distance "M". A schematic illustrating the decomposition of video scenes into GOPs and formation of a GOP with $N = 15$ and $M = 3$ is depicted in Fig. 1.

The need for simulating the network performance has introduced different traffic models for different video applications such as video conferencing and video streaming. In this research, we use a traffic model for video streaming adopted by the WiMAX Forum [18]. In this model, I, P, and B frames are modeled separately, and a fixed pattern, similar to what is shown in Fig. 1, is used in building GOPs. As shown in Fig. 1, the number of GOPs in each scene is denoted by d, and it is modeled as a geometrically distributed random variable. We will explain the MPEG-4 traffic model next.

The I frames are modeled as variable bitrate (VBR) traffic. Based on the real MPEG traffic traces, the I frames have exhibited different behavior at different time scales. At the shorter time scales of a few seconds, the bitrate varies a little around a mean level. However, the mean level varies tremendously at larger time scales. The change of the mean levels in the large time scales is called the *scene* variation [20]. A scene is a short part of the movie that does not contain sudden changes in the view while it can possibly include some zooming or object movement. In our adopted traffic model, the concept of scene has been incorporated in the model, which results in more accurate performance prediction.

As explained above, the variations of the size of I frames have two scales: 1) the small variations within a scene period; 2) the large variations among different scenes. Thus, the model considers two independent components for defining the size of the nth I frame of the video stream, $X_I(n)$, located at the kth scene:

$$X_I(n) = \bar{X}_I(k) + \Delta_I(n). \qquad (1)$$

TABLE I
MPEG MODEL PARAMETERS [18]

Model Parameter	Value
Display size	320x240
GOP pattern	$N = 15$, $M = 3$
Frame rate	30 frames per second
Scene size parameter (d)	Exponential with p=0.1
I frame size (bytes)	Log-normal ($\mu = 17068$, $\sigma = 7965$)
P frame size (bytes)	Log-normal ($\mu = 9190$, $\sigma = 7005$)
B frame size (bytes)	Log-normal ($\mu = 6839$, $\sigma = 5323$)
AR coefficients	$a_1 = 0.39$, $a_2 = 0.15$, $\sigma_\varepsilon = 4.36$

$\bar{X}_I(k)$ is the mean activity of scene k and represents the large variations; thus, it may vary greatly from scene to scene. $\bar{X}_I(k)$ is constant for all I frames in scene k while it will be different for other scenes. The $\bar{X}_I(k)$ is modeled by a log-normally distributed random variable [16]. $\Delta_I(n)$ represents the small variations of the I frames around the mean level of each scene. The $\Delta_I(n)$ is modeled by an order two autoregressive process, AR(2):

$$\Delta_I(n) = a_1\Delta_I(n-1) + a_2\Delta_I(n-2) + \varepsilon(n). \qquad (2)$$

The a_1 and a_2 are assumed to be constant for each video stream and $\varepsilon(n)$ is a normal random variable with zero mean and constant variance for each stream [16]. The parameters defining the random variables depend on the content of the video; however, we will use a constant set of parameters in our simulations similar to what is adopted by the WiMAX Forum.

The sizes of P and B frames are modeled by log-normal distributions with parameters (μ_P, σ_P) and (μ_B, σ_B). The correlation between P frames (and similarly B frames) is negligible as compared to that of I frames, and thus the model considers them as independent random variables [16]. The MPEG model parameters used in this research are presented in Table I.

By considering the parameters displayed in Table I, we see that there are two GOPs per second. Thus, the model generates two I frames per second, eight P frames per second, and 20 B frames per second. Hence, the average bitrate for each frame type is as follows: $\bar{R}_I = 273$ Kbps, $\bar{R}_P = 588$ Kbps, and $\bar{R}_B = 1094$ Kbps. Thus the overall average bitrate for each video stream is $\bar{R}_{tot} = 1955$ Kbps. We observe that the average bitrate of B frames is higher than those of I and P frames. Although the average size of a B frame is less than that of other types, we have a higher bitrate associated with the B frames as there are more B frames in each GOP, as discussed earlier.

III. Quality-of-Service in WiMAX

WiMAX networks are planned to support traffic from many different applications. While the WiMAX technology can be designed to support the backhaul connectivity for broadband communications, it can also be tailored to provide wireless access to mobile users. Supporting different types of traffic requires flexibility in design and functionality. Owing to this requirement, there are many available options in the IEEE 802.16 standards that are to be chosen by vendors and service providers in their product design or configuration. IEEE 802.16 has defined different types of QoS classes. In this section, we elaborate on the QoS classes suitable for video streaming applications.

Video streaming applications generate variable bitrate traffic which is real-time and delay sensitive. Such applications require the network to allocate network resources to handle the corresponding traffic within a limited period of time. By considering these service requirements, we observe that the real-time polling service (rtPS) class is suitable for supporting video streaming traffic. In the rtPS class, each traffic flow is characterized by a few parameters such as the minimum reserved traffic rate and the maximum sustained traffic rate. An rtPS traffic flow will not get admitted into the network if the BS cannot guarantee it the requested minimum reserved bitrate. The BS periodically polls the rtPS queues and assigns resources based on the bandwidth requests and connection parameters.

IV. Video Traffic Classification Over WiMAX

In this section, we introduce our novel solution for enhancing the quality of the video streams at the WiMAX subscriber stations. As mentioned in Section III, the video streaming traffic is mapped into the rtPS class. In order to classify the video traffic as a rtPS class, it is necessary to determine the minimum reserved bandwidth. As described in Section II, the average bitrate of a video stream that we use in this research is about 2 Mbps, and as shown in Table I, the average size of each frame depends on the frame type, and its average value varies from 6.8 Kbytes for B frames to 17 Kbytes for I frames. We note that each video frame may be fragmented into some IP packets and MAC layer data units (MDU). Thus, the loss of an MDU can result in the loss of a frame. In regular WiMAX networks, a video stream is mapped into an rtPS class, and in the case of MAC layer congestion or poor physical (PHY) layer conditions, the MDUs are dropped either at the BS or lost at the air interface. Considering the fact that each frame is fragmented into multiple MDUs, we observe that the probability of a frame dropping increases when the network becomes more congested or the wireless link between the SS and BS becomes less reliable. The main characteristics of the MPEG video traffic discussed earlier has inspired us to propose a cross-layer, content-aware traffic classification method called *multilevel service classification*.

A. Multilevel Service Classification

As mentioned in Section II, MPEG video streaming traffic is composed of I, P, and B frames. Although B frames generate the most amount of traffic, they have the least impact on the video quality. In our method, the video server indicates the type of each frame in the type-of-service (ToS) field of the IP header. Therefore, the BS can distinguish the frame type of each packet. At the BS, MPEG frames are mapped into three different rtPS classes with different minimum reserved bandwidth parameters. The parameters of each traffic flow are determined by the SS, or the video server and sent to the BS during the call admission control process. Hence, each video stream is projected into three traffic flows. A schematic explaining the concept of multilevel service classification is shown in Fig. 2. We note that the proposed traffic classification is fully compatible with the current WiMAX certified products and does not require any changes at the BS or SS.

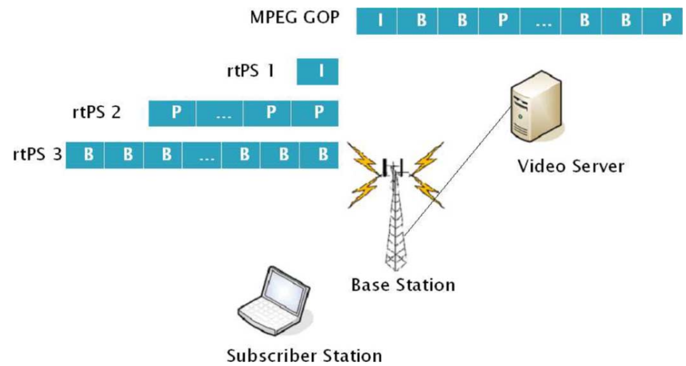


Fig. 2. Multilevel service classification.

In order to sustain the video quality, it is crucial to send as many frames as possible to the end user. Furthermore, it is important to protect the more valuable frames, i.e., the I frames, against dropping. As expected, the loss of frames will affect the video quality. Since all frames of GOPs are built over the base frame which is the I frame, the loss of an I frame will propagate throughout the GOP and all other frames will be corrupted. Similarly, the loss of a P frame also affects all proceeding P and B frames plus some preceding B frames in the corresponding GOP. However, the loss of B frames will not propagate and will result in smaller quality degradation. Therefore, it is more important to protect the I and then P frames from dropping.

When a traffic flow of a streaming application is admitted in the WiMAX network, it has the corresponding downlink (DL) queue in the BS. Thus, the DL traffic from the video server to a WiMAX SS is enqueued at its corresponding DL queue in the BS. If the traffic flow is admitted as an rtPS flow, the BS has to guarantee the requested minimum reserved bitrate for that. The DL queue will overflow if the input traffic rate exceeds the guaranteed reserved bitrate and the BS cannot allocate more resources to that flow due to either congestion or lack of bandwidth availability. In any case, the waiting time in the queues will impose some delay to the traffic flow. Full queues will drop the incoming traffic and this will degrade the video quality. On the other hand, requesting higher bitrate for flows will decrease the chance of admittance in the network. Hence, increasing the minimum reserved bitrate will make the BS admit less number of flows in the network, and this will decrease the overall network utility. Thus, there is a trade-off between the video quality and network utility; this is optimized by choosing the optimum minimum reserved bitrate value for each video stream.

As explained earlier, the I, P, and B frames require different bitrates and the B frames require the largest bandwidth while they have less effect on the video quality. Inspired by these observations, we allocate more minimum reserved bitrate to the rtPS flow corresponding to the I frames in the multilevel classification method. In order to minimize the probability of dropping I frames, we request conservatively a bitrate that is higher than the average bitrate of I frames. We note that if the I frame traffic flow cannot consume all the allocated bandwidth, the BS will spare the remaining part to other traffic flows. In order to

TABLE II
SIMULATION PARAMETERS

WiMAX Parameter	Value
MAC Frame Length	5 ms
Symbol Duration	102.86 μs
Frequency band	5 GHz
Bandwidth	20 MHz
Buffer size per SS	6 MB
Duplexity	TDD
Modulation and Coding	Adaptive
Number of Subcarriers	2048
DL Usage Mode	PUSC

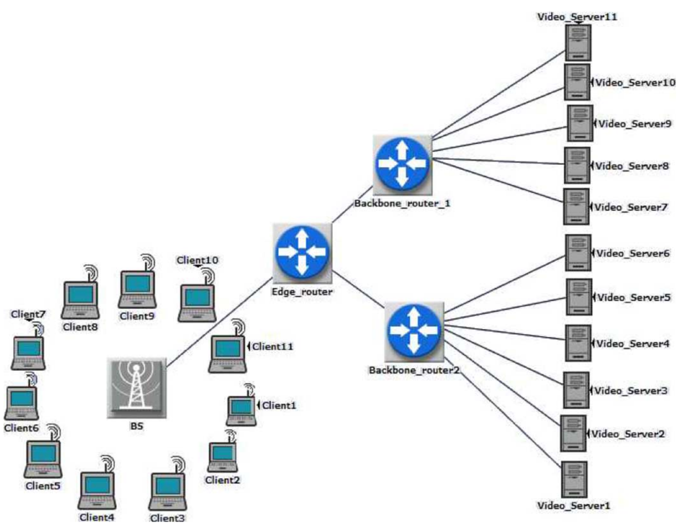


Fig. 3. Network architecture.

sustain a minimum video quality, we also request minimum reserved bitrates for the P and B frames. However, the requested bitrates are less than the average bitrates of these flows. We request higher bitrates for P frames as compared to B frames, and therefore increasing the probability of dropping B frames. We will elaborate on these threshold values in Section V where we will discuss the simulation results.

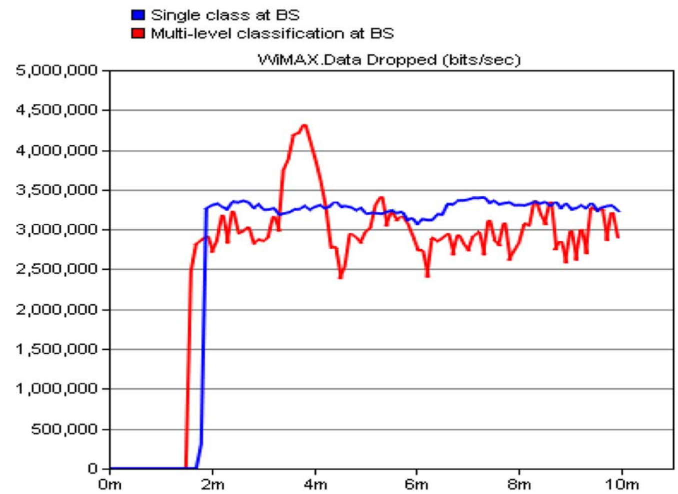
V. SIMULATION RESULTS

In this section, we elaborate on and compare the performance of the rtPS flow classification methods discussed in Section IV. We study the number of video frames received by the clients and the number of video frames dropped by the BS through some simulations. We generate video streaming traffic based on the model parameters mentioned in Table I. We use the OPNET simulator and its WiMAX package to perform our simulations. In order to achieve more accurate results, we fully simulate the PHY and MAC layers of WiMAX in our simulations. The parameters used in our simulations are described in Table II.

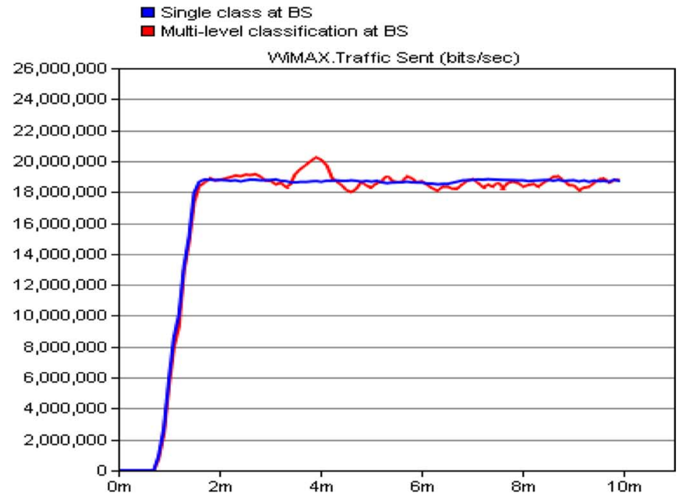
In our simulations, we considered a WiMAX network comprising of a BS and 11 SSs. Each of the SSs receives an MPEG-4 video stream from one video server. An overview of the network architecture is shown in Fig. 3. In the conventional, scheme each SS asks for one rtPS traffic flow with the minimum reserved bitrate equal to 884 Kbps. While in the multilevel traffic classification scheme, each SS asks for three rtPS flows corresponding to three different MPEG frame types. The SS will request 384 Kbps for I frames, 300 Kbps for P frames, and 200 Kbps for B frames. The simulation is run under highly traffic loaded conditions of the network. Each subscriber station asks for 884 Kbps reserved bandwidth while each video stream requires almost 2 Mbps. Hence, the BS may drop some MDUs due to lack of resources. As mentioned in Table II, the BS allocates 6MB of its memory space as buffer for each SS. This space is divided equally among different traffic flows of the SS in the multilevel service classification scheme. It is obvious that any arriving packet will be dropped if its corresponding queue is full at the BS. Table II also shows that the adaptive modulation and coding scheme (MCS) is used in simulations, and the BS chooses different MCSs based on the PHY channel status.

We conducted simulations for both of the traffic classification schemes, and compared their performance with respect to the number of MPEG frames received correctly at the end users. We ran the simulations under similar PHY characteristics and video traffic for each SS for both the schemes to understand the impact of the multilevel service classification scheme on the video quality. Fig. 4(a) and (b) shows that in both the schemes, the BS has almost dropped/sent the same amount of traffic. Thus, any difference at the video quality at the SSs is due to the different classification schemes. Fig. 5(a) and (b) represents the number of frames received by the users with the worst physical channels and the best physical channels, respectively. It is shown that although the SSs with the best PHY channels have received almost the same number of frames, the SSs with the worst PHY channels in the single class scheme have received fewer number of frames as compared to that for the SSs with the worst PHY channels in the multilevel service classification scheme. Owing to the adaptive modulation and coding deployed in our simulations, the queues are emptied much faster for the users with the best PHY channel and thus fewer frames drop for these users unlike the users with the users with the worst PHY channel.

Fig. 6(a) and (b) shows the number of frames received for each type of MPEG frames by the user with the worst and the best PHY channels, respectively, when the multilevel service classification is deployed. We note that mainly B frames are dropped in this scheme, while the I frames and most of the P frames are received at the SSs. It is also worth noting that in the single service class scheme, all types of frames are subject to dropping since no preference among the frame types is made. We thus understand that the multilevel classification scheme has better performance for the users with the worst PHY channel by delivering more frames and protecting more important frames as depicted in Figs. 5(a) and 6(s). Nevertheless, Fig. 5(b) shows that both of classification schemes deliver around 27 frames per second to the users with the best PHY channel. While the multilevel classification scheme protects most of the I frames and P frames from dropping for all users, the single classification scheme is also expected to deliver most of these frames to the users with the best PHY channel since only one or two frames are on average dropped from each GOP for those users as shown in Fig. 5(b). We thus conclude that the performance of these two schemes is comparable for users with the best PHY channel.

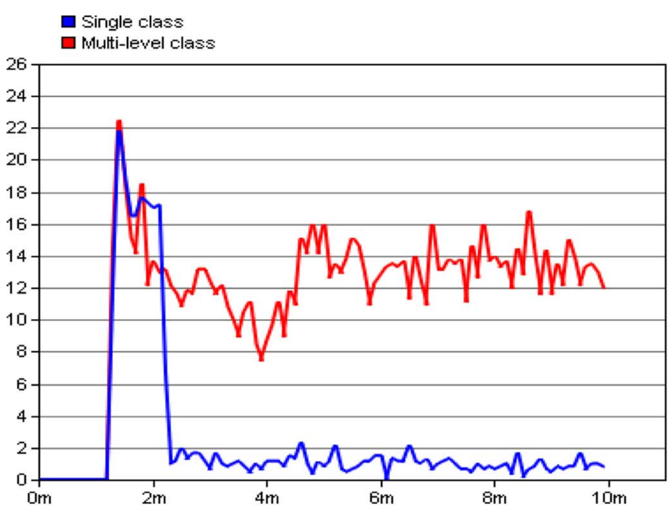


(a) Frames dropped at the BS.

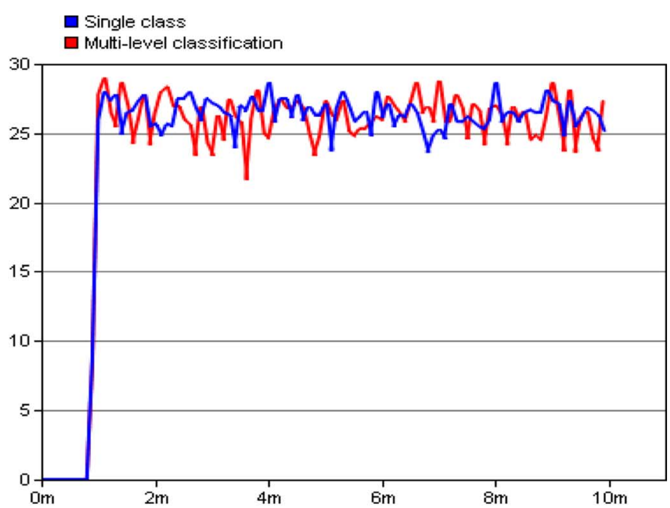


(b) Traffic sent by BS.

Fig. 4. BS performance.

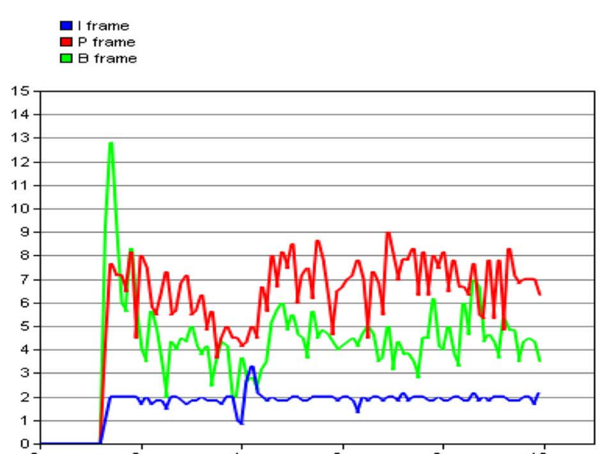


(a) Frames received by users with worst physical channel.

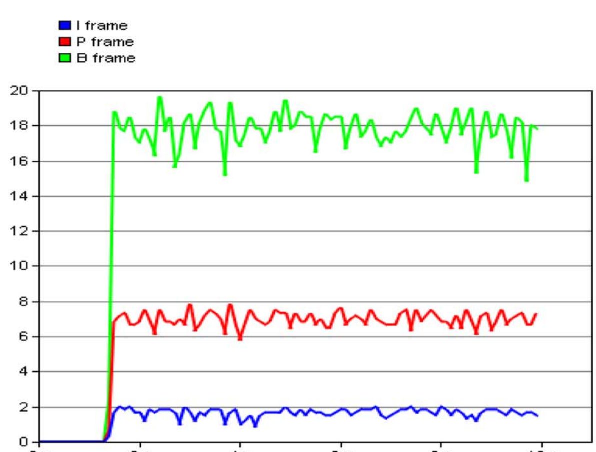


(b) Frames received by users with best physical channel.

Fig. 5. Client performance.



(a) Frames received by users with worst physical channel in the multi-level service classification.



(b) Frames received by users with best physical channel in the multi-level service classification.

Fig. 6. Multilevel classification performance.

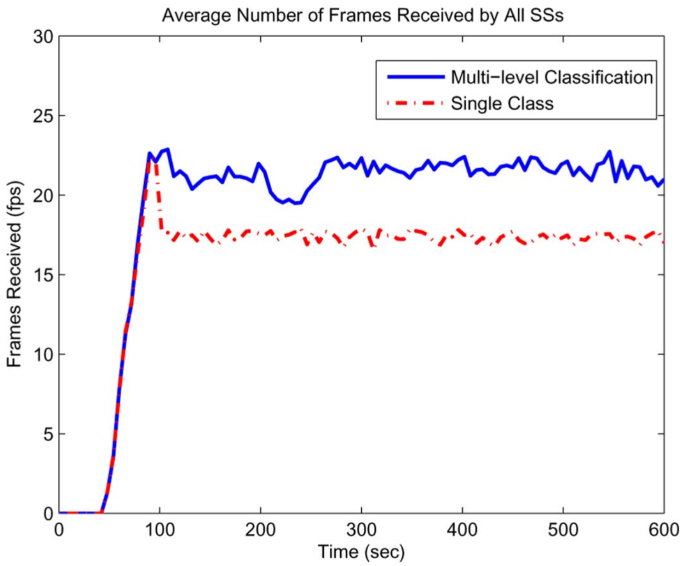


Fig. 7. Average number of frames received by all SSs.

In Fig. 7, we have calculated the average number of frames received by all users in both schemes. It is shown that the multi-level classification scheme outperforms the single class scheme in terms of the average number of frames received by all users in the network.

VI. Conclusion

In this paper, we have proposed some novel solutions to increase the performance of MPEG video transmission over WiMAX networks. We have introduced a cross-layer approach which relies on the characteristics of the MPEG frames and the elaborate QoS classification features at the WiMAX MAC layer. We have explained the challenges of transmitting video traffic over wireless networks, and discussed some of the WiMAX networks constraints and design tradeoffs, which can dramatically impact the quality of video. We have illustrated the main characteristics of the MPEG traffic, and described the MPEG model which categorizes the traffic frames into three types: I, P, and B frames. We have shown that by providing the BS with information about the type of video frame, it can map I, P, and B frames into three different rtPS service classes with different service requirements. It is shown that by incorporating our proposed traffic classification scheme at the BS, the overall number of frames delivered to each SS increases, which translates in enhanced quality of video at the end users. It is a simple and reliable scheme which can be readily deployed in WiMAX networks.

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