Perspectives on Quality of Experience for Video Streaming over WiMAX

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The advent of broadband wireless networks, such as WiMAX, is paving the way for the widespread deployment of high-bandwidth video streaming services for mobile users. To provide acceptable end-to-end performance in such a network, it is important to monitor the Quality of Experience (QoE) of the user, since the inherent variability in the wireless channel can undermine the video quality significantly. In this context, this paper undertakes a simulation study to evaluate the user's QoE (using PSNR as the representative metric) when video is streamed from a source to a Mobile Station (MS) via a WiMAX Base Station (BS). The WiMAX Forum's ns-2 simulator is used to carry out all the simulations. In particular, we explore the impact of the following parameters, namely, (1) the reserved rate at the BS for the video stream, (2) the Modulation and Coding Scheme employed, (3) the distance between BS and MS, and (4) the tolerable end-to-end delay, on the QoE. Our results point to various trade-offs that exist among these parameters, which can be effectively used to manage the user's viewing experience under varying channel conditions and resource constraints.

I. Introduction

The active deployment of broadband wireless access networks, such as Worldwide Interoperability for Microwave Access (WiMAX) [3, 6], is enabling telecom service providers to offer high bandwidth services to mobile users. One of the most popular of these services is *video streaming*, in which the user can start playing (i.e., watching) the video without downloading the entire content. As compared to download-and-play schemes, the user's Quality of Experience (QoE) in video streaming is more dependent on the underlying Quality of Service (QoS) characteristics of the communication network, significant ones among them being the reserved bit rate, packet loss rate, packet delay and delay variation (jitter). The effect of these network metrics on QoE also depends on other factors specific to the video sequence (e.g., duration of the video, its mean and peak bit-rates, burstiness, etc.), and the type of the video streaming application itself. For instance, near-live video streaming that is required for event broadcasting (e.g., soccer games) can tolerate only a few seconds of delay, whereas streaming of pre-recorded videos (e.g., movies) can tolerate higher delays [10].

Providing stringent QoS guarantees over wireless channels is in general a challenging task, however, the service-flow-specific QoS support in WiMAX renders it amenable for video streaming. WiMAX is based on the IEEE 802.16 standard for wireless transmission [12,13], which also provides an architecture for the radio access network and the connectivity to the IP core network. The MAC layer in IEEE 802.16 is connection-oriented and provides the QoS support. Traffic over a logical connection between a Base Station (BS) and a Mobile Station (MS), with pre-defined QoS parameters, is called a *service flow*.

The IEEE 802.16 standard supports different scheduling services that are suitable for service flows with different traffic types, such as constant bit rate (CBR) or variable bit rate (VBR) traffic, and real-time traffic. The WiMAX BS schedules packets over the wireless channel while ensuring the QoS guarantees of each service flow. However, the QoS parameters assigned to a service flow by the BS, e.g., the reserved traffic rate, maximum BS to MS latency, and tolerated jitter, may vary over the lifetime of the flow. This change in QoS may occur due to the variation in the load at the BS, such as when the number of users connected to the base station changes, or due to the variation in the received signal strength at the MS, such as when the user moves away from the base station, all of which can potentially alter the QoE of the user.

This paper investigates video streaming over a WiMAX network and studies the impact of the following basic network parameters on the user's QoE using PSNR (Peak Signal-to-Noise Ratio) as the representative metric.

1. The rate reserved for the service flow.
2. The Modulation and Coding Scheme (MCS) used for the wireless transmission.
3. The distance between the BS and the MS.
4. The tolerable end-to-end packet delay.

Our evaluation is based on the WiMAX Forum *ns-2* System Level Simulator available from [3]. We simulate the transmission of video clips from a video source in the wired network (i.e., in the IP core) to a vehicular receiver who is moving at a speed of 60 km/hr. Based on our simulations we make two main observations: (1) there exists a tradeoff between the video quality perceived by the user, the reserved rate and the tolerable end-to-end delay, and (2) the operating distance for a given MCS can be increased by either decreasing the video quality or by increasing the tolerable delay. These observations can be used by service providers to manage the users viewing experience while at the same time using the channel resources more efficiently.

The rest of the paper is organized as follows. In Section II, we briefly recall some background information on WiMAX and QoE metrics for video streaming. Sections III and IV describe our simulation framework and its parameters, respectively. The results of our simulation are presented in Section V. We discuss the related work in Section VI. Section VII concludes the paper with some directions for future work.

II. Background

II.A. WiMAX

WiMAX is a broadband wireless data transmission technology for fixed and mobile users [3]. It is based on the air interface protocol specified by the IEEE 802.16 standard, which details the Physical (PHY) and Media Access Control (MAC) layers for the wireless link between the BS and the MS. The standard for the fixed user scenario was given in IEEE 802.16d [12], which was later enhanced to support user mobility in IEEE 802.16e [13]. As the MAC layer can support multiple PHY specifications, in this paper, we consider the Orthogonal Frequency Division Multiple Access (OFDMA) PHY in which the BS allocates transmission resources to multiple MSs and data transmission is done on a frame-by-frame basis. An OFDMA frame is a fixed-sized contiguous region, in both the time and frequency domains, which is divided into downlink and uplink sub-regions. Within an OFDMA frame, the BS scheduler allocates slots to MSs, where a slot is the smallest transmission resource in a frame. In addition, we use the Point-to-Multipoint (PMP) mode of IEEE 802.16e and the duplexing scheme employed is Time Division Duplex (TDD).

As defined earlier, a service flow constitutes a flow of packets between the BS and MS with predefined QoS parameters. The resources allocated to a flow within an OFDMA frame is based on the QoS requirements of the service flow. The IEEE 802.16e standard supports five scheduling services to accommodate flows with diverse QoS requirements. Of these five services, the real-time Polling Service (rtPS) is designed for flows that transport variable bit rate (VBR) traffic, such as MPEG video streaming, while the Unsolicited Grant Service (UGS) is designed for flows that transport constant bit rate (CBR) traffic. Since our primary goal is to investigate the impact that different reserved rates have on the perceived video quality, we performed our simulations using UGS and consider only its *maximum sustained traffic rate* QoS parameter.

The IEEE 802.16 supports multiple Modulation and forward error correction Coding Schemes (MCS). Based on the channel condition experienced by the MS, the BS can choose to employ different MCSs for different MSs, which can further change dynamically across frames for the same MS. Choosing a more robust MCS allows the transmission to tolerate poorer channel conditions, but results in lower data rate (i.e., transmitting fewer number of bytes per slot in the OFDMA frame). WiMAX supports adaptive modulation and coding that adapts the MCS so as to provide the highest data rate for a given channel condition at the MS. The received signal strength at the MS depends on the path loss, shadowing and multipath effects of the wireless channel, which in turn depend on the location, speed and the physical objects in the vicinity of the MS. In the WiMAX Forum *ns-2* simulator, the wireless channel is modeled using Cost-231 Hata path loss model and ITU VEHIC_A channel model. We refer the reader to [4, 14] for additional information.

II.B. QoE for Video Streaming

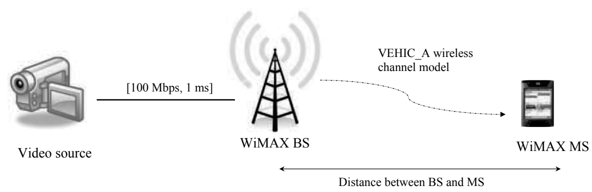
Accurately estimating the end-user QoE is critical for provisioning and managing a video streaming service. The QoE for a streaming service depends on conventional network metrics such as bit rate, packet loss rate, packet delay and jitter, and various video sequence specific factors, such as its encoding scheme and the video streaming application. Two important QoE metrics in a video streaming service are the receiver starvation probability and the received video quality [21]. The starvation probability is the long-run fraction of frames or packets that miss their playout deadline at the receiver. The received video quality can be measured using subjective or objective video quality metrics. Subjective metrics, such as video quality Mean Opinion Score (MOS), is based on the subjective grading by human test subjects [5]. Objective metrics, on the other hand, are based on mathematical models of human perception, and do not require evaluation by human test subjects. Although objective metrics may not accurately capture human perception, they are the preferred metric when it comes to service providers since they need to perform a large number of evaluation experiments and continuously monitor the service that they are offering.

In our simulations, we study the effect of maximum tolerable end-to-end packet delay on video quality. Packets that do not meet this delay bound are deemed to have arrived late, and are consequently discarded by the receiver. This parameter characterizes the impact of receiver starvation due to late packet arrivals on video quality. We note that the tolerable delay simulation parameter is different from the maximum latency QoS parameter for a WiMAX service flow. If maximum latency is set for a downlink service flow, say to γ seconds, then the WiMAX BS scheduler tries to ensure that the BS to MS delay for all the packets in that flow is less than γ seconds. Thus, the maximum latency QoS parameter acts as one of the handles for maintaining the tolerable delay of a video streaming application (we do not set the maximum latency QoS parameter in our simulation as it is not supported by the current version of the simulator).

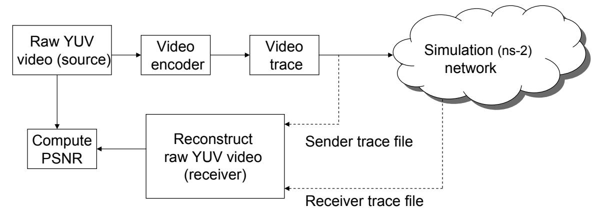
For evaluating the video quality, we use one of the most commonly used objective video quality metric, the Peak Signal-to-Noise Ratio (PSNR). We use the PSNR computation from [16] that compares the maximum possible signal energy of the luminance (Y) component of the frame to the error energy (the mean square error) of the received frame relative to the original frame. All PSNR values reported in this paper are in dB. We also follow the PSNR to MOS mapping in [16]: PSNR of at least 25, 31 and 37, can be roughly mapped to a MOS of 3 (fair), 4 (good) and 5 (excellent), respectively. We note that this mapping is only a rough estimate as PSNR does not capture all the aspects of human perception of a video [11, 18].

III. Simulation Framework

We perform our simulations on the WiMAX Forum *ns-2* System Level Simulator [3] on the simple topology shown in Figure 1(a), which consists of a source node (typically in an IP network) generating video traffic, a WiMAX Base Station (BS) and a Mobile Station (MS). The source node is connected to the BS via a 100 Mbps link and has a propagation delay of 1 ms. In each simulation run, the source node streams a short video clip to a vehicular mobile user. During a run of the simulation, the user moves at 60 km/hr along a circular path maintaining a fixed distance from the BS (we deliberately choose this mobility pattern so that the path loss does not change significantly during a single simulation run). As can be seen from the figure, the streamed video is transmitted over the WiMAX downlink, i.e., from the BS to the MS. The various system parameters used in the simulation are summarized in Figure 4. We set the BS queue length to 1000 packets to ensure that no packets are dropped at the BS due to queue overflow. In line with other QoE studies, we also use short representative video clips (outlined in subsection III.A) and employ the publicly available EvalVid video evaluation toolkit [16] (outlined in subsection III.B) to measure the impact of various network parameters on the QoE.



(a) Simulation Topology



(b) EvalVid Framework

Figure 1: Simulation Framework

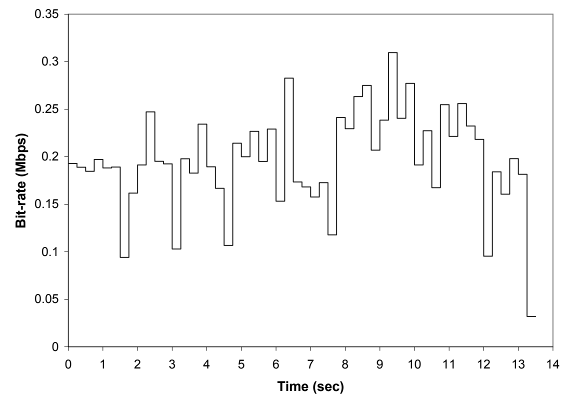
III.A. Streamed videos

We stream short YUV video clips namely, *Foreman*, *Salesman* and *News videos* available from [1, 2] in our simulations. The format used (in terms of the pixel dimensions) is the Quarter Common Intermediate Format (QCIF) as this is one of the formats suitable for mobile devices. QCIF has a resolution of 176×144 pixels per frame. Figures 2(a), 2(b) and 2(c) show the bit rate (averaged over intervals of 0.25s) of the Foreman, Salesman and News videos, respectively. All three videos exhibit variable bit-rate (VBR) characteristics. The mean, peak and standard deviation of the bit rates are given in Figure 3.

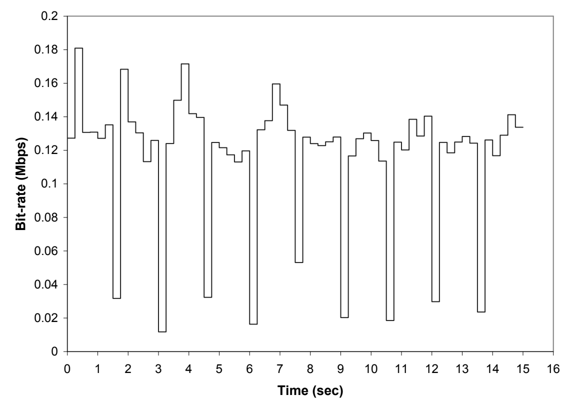
III.B. EvalVid Framework and End-to-End Setup

EvalVid [16] is a toolkit that provides an integrated framework to assess the quality of the video transmitted over a network. It has a modular structure, making it possible to exchange, at user's discretion, both the underlying transmission system as well as the codecs. Figure 1(b) shows the structure of the EvalVid framework and its interfaces with *ns-2*. We use the three agents, *MyTrafficTrace*, *MyUDP*, and *MyUDPSink*, that are part of EvalVid that integrates it with *ns-2*.

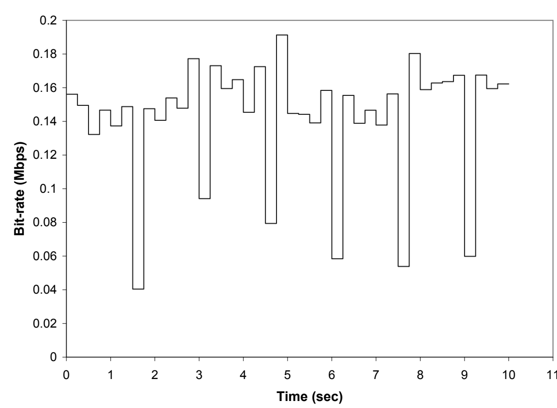
The evaluation process begins with the encoding of the raw YUV video, which then generates the respective MPEG frames (I, P and B) and the corresponding encoding times. The agent MyTrafficTrace is an extension of the *ns-2* agent Application/Traffic/Trace, which extracts the frame type, frame size, and inter-packet times from an input video trace file. It then fragments the frames into segments depending on the MTU (maximum transmission unit) of the underlying network and passes them to the lower UDP transport layer at appropriate times. MyUDP is an extension of the *ns-2* agent Agent/UDP, which receives packets from the upper layer and records, in a sender trace



(a) Foreman



(b) Salesman



(c) News

Figure 2: Bit rates of the videos averaged over 0.25 sec intervals

	Foreman	Salesman	News
number of frames	400	449	300
duration (sec)	13.33	14.96	10
mean bit rate (Kbps)	198	116	142
peak bit rate (Kbps)	325	181	190
sd of bit-rate (Kbps)	50	40	36

Figure 3: Statistics for the three videos

file, the time stamps, packet ID, and payload size of each transmitted packet. Lastly, MyUDPSink receives the transmitted packets over the network and records, in a receiver trace file, the received time stamps, packet IDs, and payload size. After the simulation, based on these trace files and the original encoded video, EvalVid produces the corrupted MPEG-4 video, which is then decoded and error concealed into raw YUV format. Finally, this YUV video is compared with the original YUV video to evaluate the end-to-end delivered video quality (i.e., the PSNR values of the video frames).

Parameter	Value
PHY	OFDMA
Duplex Scheme	TDD
Carrier Frequency	2 Ghz
Channel Bandwidth	10 MHz
Frame duration	5 ms
Transmission power	0.2 W (23 dbm)
downlink/uplink ratio	3:2
Antenna model	Omni-Directional
Base station queue	1000 packets
Fragmentation/Packing	ON
ARQ	OFF
Max MAC PDU size	1028 bytes

Figure 4: System Parameters

IV. Simulation Parameters

We now characterize the impact of the following parameters on the user's viewing experience using PSNR (averaged over all the frames in the original video) as the metric.

Modulation and Coding Scheme (MCS): We use the scenario where the video source transmits clips to the user over the downlink transmission from the BS to MS (see Fig. 1(a)) and consider three MCSs used by the BS: QPSK-$1/2$, QPSK-$3/4$ and 16 QAM-$3/4$ (in decreasing order of their robustness against poor channel condition). The number of bytes that can be transmitted per slot for each of these MCSs are 6, 9 and 18 bytes, respectively. Thus, moving from QPSK-$1/2$ to 16 QAM-$3/4$ results in a tradeoff - a reduction in the number of bytes that can be transmitted (per slot in the frame) in return for a more robust transmission against the adverse channel conditions.

BS to MS Distance: Due to the path loss effect, the received signal strength at the MS decreases as the user moves farther away from the BS. Within a simulation run, we maintain a constant distance between the BS and the MS (i.e., the user moves in a circular arc around the BS) so that there is very low variation in the wireless channel path loss, and captures only the variations due to shadowing and multipath effects. Path loss effects are studied by varying the distance between the BS and MS across the simulation runs.

Reserved Rate: We assume the UGS scheduling policy for the service flow and vary its *maximum sustained traffic rate* (MSTR) QoS parameter. Since there are no other competing flows at the BS, the MSTR requested is always less than the maximum rate that the BS can support, and is the rate allocated to the service flow in every run. We call this rate as the reserved rate for the simulation and consider reserved rates close to the video's mean rate (roughly $\pm 15\%$ the mean rate). Note that the reserved rate is specified in bytes/frame, which can be converted to Kbps by multiplying by 1.6, since 200 frames are sent per second. Also, when we specify rates in bytes/frame, the *frame* refers to the OFDMA WiMAX frame of the IEEE 802.16 standard.

Tolerable Delay: We consider a very simple receiver strategy at the MS. For every run there is a constant tolerable end-to-end packet delay δ (referred to as *delay* in figures). The receiver treats every packet that is not received within δ seconds of its sent time as being late and then discards it. The tolerable delay value can be interpreted in multiple ways: (1) as the end-to-end packet delay that may be required for a video streaming application, or (2) as being proportional to the size of the receiver playout buffer that may be used for a smooth playout.

V. Simulation Results

For a given video, reserved rate, MCS, and distance, we ask the following question: what is the minimum value of tolerable delay for which the PSNR of the received video is greater than or equal to some desired value? To this end, we perform a simulation run with a given video and the above set of parameters to obtain the received packet trace, say R'. Note that R' is recorded before any packets are discarded by the receiver. Next, for each value of the tolerable delay δ (starting from $\delta = 0.05$, and increasing in steps of 0.05 second), we create a *playout* packet trace R from R' by discarding all *late* packets, i.e., packets in R' that are not received within δ seconds of their sending times. Then, using R and the EvalVid setup, we compute the average PSNR of the received video. Finally, we choose the minimum δ for which the average PSNR is greater than or equal to the desired PSNR value.

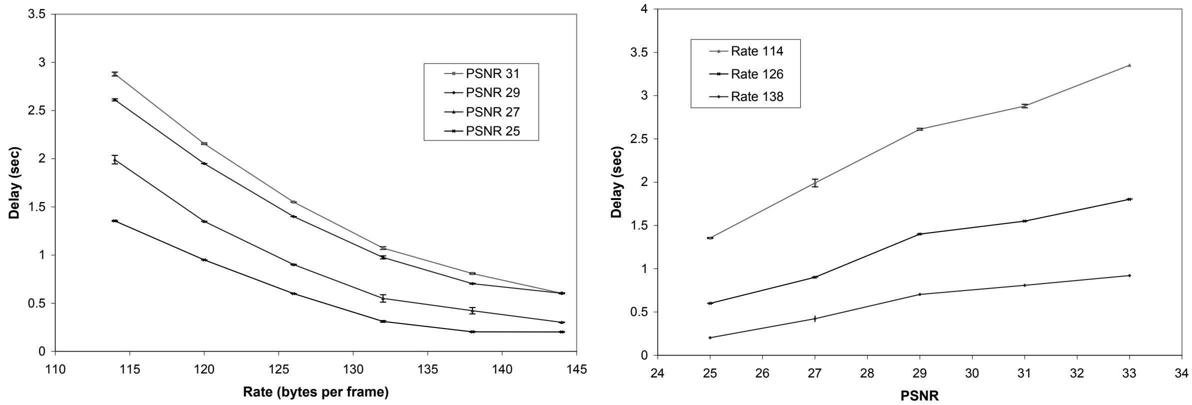
Before presenting our results, we make some observations on the packet loss in our simulation setup. In general, the network loss is composed of loss in the wired network (which is negligible) and in the WiMAX wireless access network. Since we have set the BS queue size to be high (1000 packets) compared to our workload (which are short video clips), there are no packet drops at the BS. Thus, the network loss primarily consists of packets that are lost during the wireless transmission, which is heavily dependent on the MCS used, and the BS to MS downlink wireless channel condition.

Let S be the set of packets sent in a given simulation run and L_N be the set of packets that are lost in the network. Thus, the set of packets received by the user (including late packets) is $S - L_N$. Next, let $L_D(\delta)$ be the set of late packets, i.e., packets that do no adhere to the tolerable delay bound of δ seconds. All packets in $L_D(\delta)$ will be discarded by the receiver. Thus, effectively, the set of playout packets at the receiver is $S - L_N - L_D(\delta) = R$ (recall the definition of R earlier in this section). Note that, the PSNR of the received video is computed based on the frames generated from the playout packets in R. The PSNR for a received frame decreases with the increase in the distortion of the received frame relative to the original frame, and the PSNR metric for a run is the average of the PSNRs of all the frames in the video. Thus, as the number of packets in R increases, the PSNR for the run will either increase or remain the same.

Impact of varying desired PSNR: We now present our first set of results in Figure 5(a) for the Foreman video. We fix the distance of the MS from the BS to 550m and the MCS to QPSK-$1/2$, and consider four different values for desired PSNR: 25, 27, 29 and 31. Next, for a given reserved rate, we find the minimum tolerable delay that provides the desired PSNR for the received video. The reserved rate is varied around the average rate of the Foreman video, which is 198 Kbps ≈ 124 bytes/frame. For each combination of the PSNR and the reserved rate, we repeat the simulation 30 times with different starting times, and plot the mean and the 95% confidence intervals in Figure 5(a). (The width of the error bars are too small to be observed clearly.)

From the figure, we observe that, for a given PSNR, as the reserved rate increases, the tolerable delay decreases, which is due to lower queuing delays experienced by the packets of the service flow at the BS. However, as the rate is increased beyond 124 bytes/frame, corresponding to its mean rate, the network becomes over-provisioned for the video flow, and the additional reserved slots in a frame are primarily useful for the occasional bursts in the video rate. Thus, the decrease in tolerable delay, for every 6 bytes/frame increase in rate, decreases as the rate is increased. Beyond a certain reserved rate, increasing the rate does not change the tolerable delay. We note that, for our system parameter setting, 6 bytes can be transmitted in a slot of the frame when the MCS is QPSK-$1/2$. Since the WiMAX MAC of the simulator allocates integer number of slots to a service flow, we increase the reserved rate in multiples of 6.

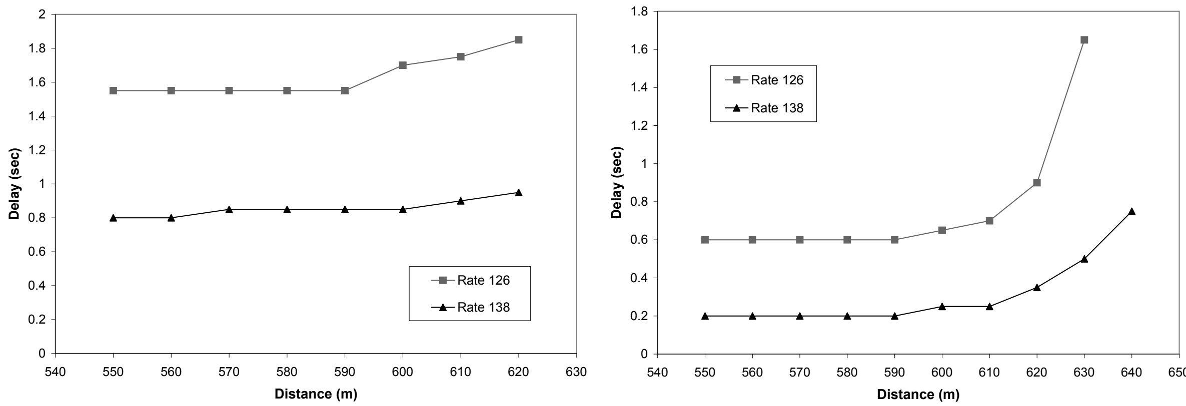
Interestingly, as shown in Figure 5(b), for a given rate, as the desired PSNR is increased, the tolerable delay also increases, which can be explained as follows. In a given simulation run, to achieve a higher PSNR, the number of receiver playout packets should increase. Since the distance, MCS and the rate are fixed, packets lost in the network does not vary significantly. However, increasing the tolerable delay, reduces the number of late packets, and thus increases the number of playout packets. Thus, for a higher desired PSNR, we need a higher tolerable delay. This behavior indicates that, for provisioning a video streaming



(a) Delay vs. Rate for different PSNR

(b) Delay vs. PSNR for different rates

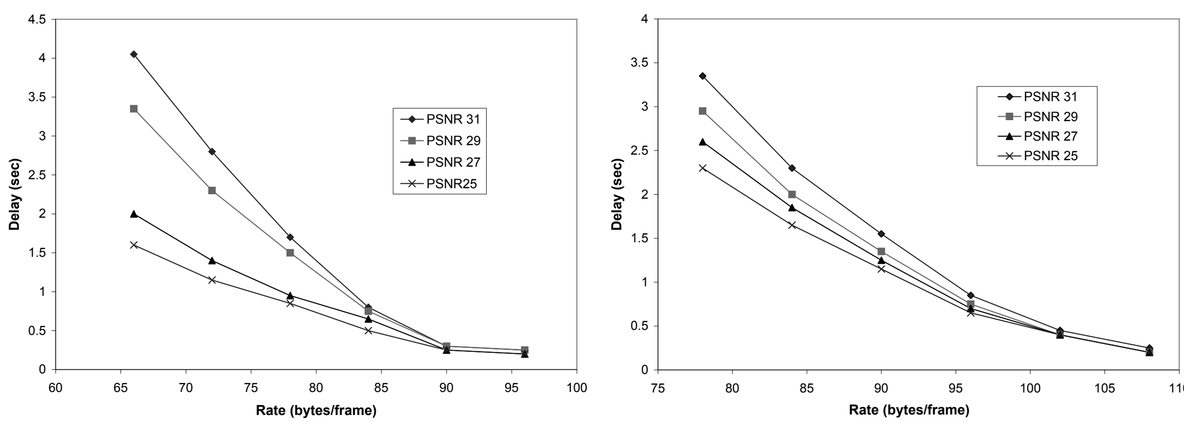
Figure 5: Delay, rate and PSNR for Foreman video (Distance=550m, MCS=QPSK-1/2)



(a) PSNR=31

(b) PSNR=25

Figure 6: Delay vs. distance for different rates and PSNR for Foreman video (MCS=QPSK-1/2)



(a) Salesman

(b) News

Figure 7: Delay vs. rate for different PSNR for Salesman and News videos (Distance=550m, MCS=QPSK-1/2)

service, we can tradeoff user video quality with the end-to-end tolerable delay of the video. Figure 5(b) presents a subset of the results that are shown in Figure 5(a) for three different rates. In Figure 7, we show the tolerable delay versus rate plots for the Salesman and News videos. The general trends in these plots follow the Foreman video mentioned above.

Impact of varying distance: Next, we investigate the variation in tolerable delay as the distance between the BS and the MS varies. We fix the MCS to QPSK-$1/2$. We consider two desired PSNR values 25 and 31, roughly corresponding to 3 (fair) and 4 (good quality) on the MOS scale. We also consider two different rates 126 bytes/frame and 138 bytes/frame. The first rate is close to the average rate of the video (124 bytes/frame), and the second rate represents an over-provisioned case. We then vary the distance between the BS and the MS from 550m to 700m, in steps of 10m, and examine the minimum tolerable delay that can support the desired PSNR. The results are shown in Figure 6. Consider the rate of 126 bytes/frame, the top curve in the figure. For PSNR=31, we note that the tolerable delay does not change significantly with increase in distance. However, at distances higher than 620m, PSNR=31 cannot be achieved for any value of tolerable delay, and therefore, we do not plot the delay values for distances higher than 620m in Figure 6(a). In contrast, for PSNR=25, the increase in tolerable delay is insignificant till 610m, then there is a significant increase until 630m, and finally for distances higher than 630m, PSNR=25 cannot be achieved for any value of the tolerable delay.

The above result can be explained as follows. Recall the set of receiver playout packets is given by $R = S - L_N - L_D(\delta)$. As the distance is increased, due to the path loss effect there is an increase in the number of packets lost L_N in the wireless channel. However, for a given rate, the loss is quite low till some distance (580m), and then starts to increase. This loss of packets can be compensated, and hence the desired PSNR can be maintained, by increasing the tolerable delay δ, which reduces the number of late packets discarded at the receiver $L_D(\delta)$. This explains the trend seen for PSNR=25. For illustration, in Figure 8, we show the number packets lost in the network and the number of packets discarded at

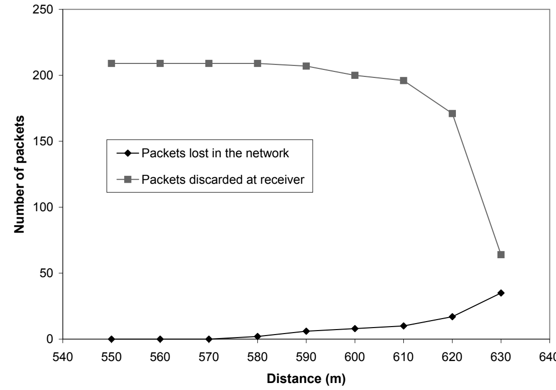


Figure 8: Number of lost and discarded packets vs. distance (PSNR=25, Rate=126 bytes/frame)

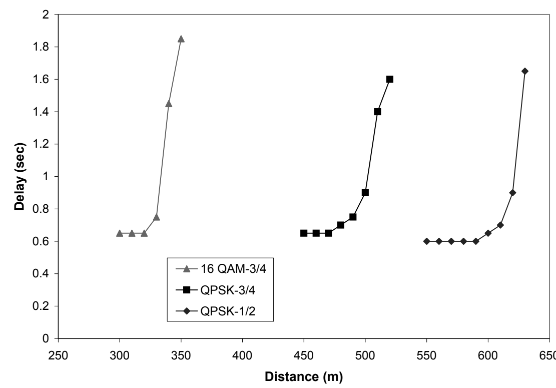


Figure 9: Delay vs. distance for different MCS (PSNR=25, Rate=126 bytes/frame)

the receiver, for desired PSNR of 25 and reserved rate of 126 bytes/frame. We note that, at distances higher than 620m, the packets in $S - L_N$ are insufficient to provide a PSNR=31, even if no packet is discarded at the receiver. Thus at a distance of 630m, no value of tolerable delay can provide the desired PSNR of 31. These trends indicate that, for a given MCS and rate, it is possible to stream a video across larger distances by relaxing the requirements on video quality or the end-to-end tolerable delay.

Impact of varying MCS: We now investigate how tolerable delay changes with distance as we change MCS. We fix the rate to 126 frames/bytes and the desired PSNR to 25. We consider three MCSs: QPSK-$1/2$, QPSK-$3/4$ and 16 QAM-$3/4$, in the order of decreasing robustness to poor channel condition. For our system parameter settings, 6, 9 and 18 bytes can be transmitted in a slot of a frame for QPSK-$1/2$, QPSK-$3/4$ and 16 QAM-$3/4$, respectively. The results are shown in Figure 9. The results for QPSK-$1/2$ is same as in

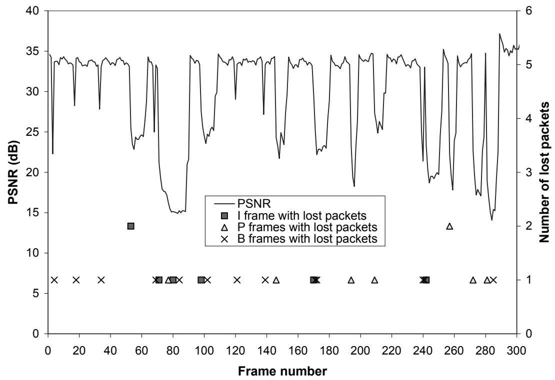


Figure 10: PSNR vs. Frame Number (Rate=126 bytes/frame, Distance=630m, MCS=QPSK-1/2)

Figure 6(b). The trends for each MCS is similar, and their explanation is similar to that of Figure 6. However, highest distance at which the desired PSNR of 25 can be achieved depends on the robustness of the MCS: 630m in the case of QPSK-$1/2$, 530m with QPSK-$3/4$, and 350m for 16 QAM-$3/4$.

Impact of packet loss on PSNR: In all simulation results above, we have considered the values of PSNR averaged over all frames in the video. In Figure 10, we present the PSNR for each frame for a portion of a run with poor channel condition. We use the Foreman video for this run, with reserved rate, MCS and distance being 126 bytes/frame, QPSK-1/2 and 630m, respectively. To focus on the effect of channel condition on PSNR, we set a very high tolerated delay value for this run to ensure that there are no late packets. (Note that, as indicated by Figure 6, the channel condition is poor at 630m with QPSK-1/2.) For each frame with a lost packet, we also show the number of lost packets and the frame type. For this run, we observe that loss of a packet in an I-frame has severe impact on video quality, both in the amount of decrease in PSNR as well as the duration of the impact. Impact on video quality due to lost packets in a P-frame is less than that of an I-frame.

VI. Related Work

Managing QoE for video streaming services has been extensively studied in both wired and wireless networks. In this section we discuss some papers that are related to our simulation work.

Data from streamed video is usually buffered at the receiver before playback to prevent interruptions due to end-to-end packet delay variations. However, increasing the buffered data also increases the end-to-end delay. For Variable Bit Rate (VBR) videos, various smoothing techniques have been proposed to increase the network bandwidth utilization and to reduce receiver buffering requirements, while ensuring continuous playback [7, 17, 22]. Another approach for reducing receiver buffer is to adapt the playout rate [8, 15]. In our simulations, we assumed a simple receiver that discards any packet whose end-to-end delay exceeds a fixed bound, say, δ seconds. Thus, assuming a constant playout rate, the receiver may use a buffer of size δ times the playout rate. We expect that the buffer management techniques mentioned above will improve our simulation results by either increasing the observed PSNR or by decreasing the tolerable delay.

There has been significant research work on gracefully degrading the video quality when the network conditions deteriorate. Two such approaches are the use of scalable video coding for video streaming [19, 20, 25] and the prioritization of packets in a video stream [10]. We do not use these schemes in our simulation, however, we expect that using them may have significant benefit. For example, a simple scheme that prioritizes packets of I-frames over other packets, and thus, reduces their chance of being discarded due to late arrival, may improve PSNR of a run.

Recently, there has been considerable interest in multimedia streaming over WiMAX. The paper that is closest to our work is [24], where the authors study the quality (measured using PSNR and MOS) of video streaming over a fixed WiMAX network. Two 802.16 scheduling services, Best Effort (BE) and real-time Polling Service (rtPS), are evaluated using commercial WiMAX equipments. For each of the two scheduling services, the paper considered two different reserved bit rates: under-provisioned and over-provisioned with respect to the average bit rate of the video. Also for rtPS, the paper considered three different maximum latency QoS parameter values. From the experiments, the paper observed that both scheduling services perform well for the over-provisioned case. However, for the under-provisioned case, the video quality was fair (MOS $\geq$ 3) for rtPS only when the maximum latency was set to a high value, and the video quality was poor (MOS $<$ 3) for BE. These trends are consistent with our simulation experiment, where we use the UGS scheduling service with a simple scheduler that only ensures the reserved rate. In addition, our simulation also considers the impact of other parameters, namely, the video quality (measured using PSNR), the BS to MS distance, and the MCS. In [9], the authors compare video streaming over WiMAX and ADSL (with equal reserved rates), and show that with careful fine-tuning of the configuration parameters, performance of WiMAX can be comparable to ADSL in terms of the traditional network metrics (loss, delay, jitter and throughput). The impact of handover on the performance of video streaming in mobile WiMAX is studied in [23]. We do not consider handovers in this work.

VII. Conclusion

In this paper, we investigated the Quality of Experience (QoE) of video streaming over a WiMAX network using the WiMAX Forum *ns-2* simulator. The video quality was measured using the average PSNR of the received video. We studied the impact of the following parameters on the QoE of the received video: the reserved bit rate for the service flow at a WiMAX BS, the distance between BS and MS, the Modulation and Coding Scheme used, and the tolerable end-to-end packet delay. Our results highlight various trade-offs that exist among these parameters, which can be effectively utilized to manage the user's viewing experience in the presence of varying channel conditions and wireless resource limitations.

There are multiple directions for future work. Firstly, our simulation can be extended to study the impact of specific techniques, such as scalable video coding and packet prioritization, on the user's experience [10, 20]. Another direction for extending our simulation could be to investigate the QoE for multiple video streams over mobile WiMAX networks. Finally, although the observations from our simulation results can be used to manage a video streaming service, the network parameters (e.g., rate and delay) need to be specifically tuned for each video. Further work could explore how to set the network parameter with minimal knowledge of the video streams.

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