Options for WiMAX Uplink Media Streaming

Salah Saleh and Martin Fleury*
University of Essex, United Kingdom

ABSTRACT

IEEE 802.16e (mobile WiMAX) uplink behavior is a relatively neglected area of investigation, but emerging interactive media services have highlighted the need for closer investigation of uplink issues. In this simulation study of uplink queue management, media streaming by TCP-Friendly Rate Control (TFRC) was found to offer advantages. However, temporal behavior of the queuing disciplines exhibits oscillations in buffer occupancy with build-up of delay during video streaming. The paper suggests a possible remedy lies in the choice of the more complex H.264/AVC Main profile, though this will impact upon mobile devices. It was also found that WiMAX video delivery is sensitive to choice of transmission frame size.

Keywords: Active Queue Management, H.264 codec, media streaming, mobile WiMAX, TFRC

INTRODUCTION

IEEE 802.16 (known as WiMAX) (IEEE, 2005) allows rapid deployment of media services in areas in the world unlikely to benefit from extensions to both 3G such as High Speed Downlink Packet Access (HSDPA) and Universal Mobile Telecommunications System (UMTS) such as Long-Term Evolution (Ekstrom et al., 2006). WiMAX's uplink capacity should exceed that of HSPDA's 384 kbps, though not LTE's. In Brazil, mobile WiMAX is the basis of a digital TV service but there is also interest in exploitation of uplink interactive services (Meloni, 2008), which could involve media streaming. WiBRO in Korea is also now harmonized with WiMAX and likewise (Kim, Lee & Lee, 2008) interactive media services are proposed. Uplink media streaming is a value-added addition to a broadcast IPTV service. As such it is attractive to commercial operators as it could be used for mobile video conferencing and video telephony, as well as the exchange of personal video clips.

For these services, selection of appropriate video transport and uplink (UL) queue management will be important to reduce packet loss and latency. Therefore, the contribution of this paper is to make recommendations for video streaming sourced from a WiMAX Subscriber Station (SS). The paper particularly examines active queue management (AQM) schemes over IEEE 802.16 networks. We compare RED, REM, BLUE, and question whether they offer any advantage over drop tail queue management schemes. We also compare simple UDP and congestion-controlled TCP-Friendly Rate Control (TFRC) (Handley et al., 2003) video stream transport. A number of schemes (e.g. Fu et al. (2006)) have been developed for TFRC-based streaming over a wireless network, particularly if a further network path exists over a core wired network (Görkemli, 2008). While a basic UDP stream represents an AQM non-compliant source (see the next paragraph), TFRC represents an AQM compliant source. As TFRC responds to packet loss, as well as round-trip time and packet size, it will change its rate if packets are actively dropped from an SS queue. TFRC is intended to provide a less aggressive congestion control than TCP's 'sawtooth' response to network congestion, resulting in a smoother sending rate more suitable for video streams.

Within the wired Internet, AQM techniques (Koo, Ahn, & Chung, 2004) are intended to achieve high link utilisation without introducing delays. The most well-known AQM technique is Random Early Detection (RED) (Floyd and Jacobsen, 1993) in which packets in the queue are intentionally dropped or explicitly marked as a signal to the source to reduce its rate. For a compliant source a dropped packet prompts the congestion control mechanism to reduce its sending rate. For a non-compliant source, one that does not reduce its rate, RED also keeps the queue size low but does not discriminate against 'bursty' traffic sources. Dropping occurs in a probabilistic fashion according to the time-wise, exponentially-averaged queue size and, to avoid unreasonable overhead, no record is kept of the state of each connection. By random dropping, rate synchronization amongst compliant sources is avoided.

Our simulation results show that, though a particular queuing discipline may be more suspect than others, in the scenario investigated all the examined AQM schemes can lead to oscillatory queue occupancy during periods of congestion, which makes them unsuitable for video streaming over IEEE 802.16 networks. In fact, one cannot rely on prior studies advocating one AQM technique or another if these studies do not investigate temporal behavior.

A possible solution in a dedicated WiMAX DTV service is priority treatment of video reference frames. For general media streaming, our work suggests that strict access control will be needed for WiMAX uplink services to reduce the risk of congestion adversely affect video transfer. When congestion occurs, a packet dropping policy should again reflect the relative importance of video frame types and their contribution to the data load. However, this is not straightforward because the frame types in use can vary according to the codec profile.

RELATED RESEARCH

Ali, Dhrona, & Hassanein, H. (2009) try to survey all types of UL bandwidth allocation algorithm and evaluate these algorithms with respect to their abilities to support multiple classes of services and to optimise bandwidth utilization. Their conclusion is that no algorithm dominates in all criteria such as average throughput or delay, frame utilisation, request honouring, or packet loss minimisation. However, the work of Ali, Dhrona, & Hassanein, H. (2009)) is not application specific and given the distinct possibility that WiMAX may dedicated to media services, further study is merited.

Kim & Yeom, 2007 propose an uplink bandwidth allocation scheme for best effort TCP streams. The scheme does not involve a bandwidth request process at the SS. Instead, the Base Station (BS) estimates the bandwidth required for a TCP flow based on its current sending rate and subsequently applies a max-min fairness allocation. However, it is unlikely that TCP will be employed to transport media services, because of its aggressive congestion control mechanism that results in rapid fluctuations in sending rate. Because TCP provides a reliable service high latency can result from repeated attempts to deliver a packet, causing interruptions to the media stream display.

Niyato & Hossein (2006) propose a bandwidth allocation scheme for 802.16 real-time and non-real-time traffic (treated as a single queue) in an SS. A rate-control mechanism such as AQM, though this is not precisely specified in the paper, allows a target queue size to be met. Consequently, delay and the packet dropping probability can be controlled. However, this is a bandwidth allocation study and as such is rather remote from possible applications. RED is mentioned as a possible way of implementing the rate control mechanism but no detailed work on the impact upon a media service was carried out.

Turning to AQM in general, the research of Lakkakorpi et al. (2006) considers congestion at the BS for downlink TCP connections and the effect on buffer overflow and delay. The authors apply different AQM techniques on an SS queue basis, as, unlike for Internet routers, queue aggregation is not possible in WiMAX. It was found that the AQM algorithms reduced delay most for long-lived TCP connections with large advertised windows. However, correct configuration of AQM was found to be important to avoid service deterioration. UDP packets were not managed by the AQM. However, Rahmani, Hjelm, & Aklund (2008) in a general context studied the behavior of several AQM algorithms when UDP flows under the control of TFRC competed with TCP flows in a heterogeneous network. It was found that the tested AQM schemes were sensitive to traffic burstiness, load and round-trip-time (RTT) and were differentiated by their ability to stabilise the queue length. Again correct configuration was important.

Cuny & Lakkakorpi (2006) applied variants of RED at the BS of a cellular General Packet Radio Service (GPRS) system, considerably improving the goodput and delay in this situation, provided the AQM was correctly configured. Scheduling (Raghu, Bose, & Ma,2008) is also queue length aware across the SSs in a broadband system but the scheduler favors the real-time over non-real-time service in bandwidth allocation by the BS.

An interesting analysis of video packet priority-based AQM occurred in Orlov & Necker (2007). In that work bi-predictive (B-) frame packets were dropped from the queue upon impending congestion. The scheme is intended for HSDPA with the MPEG-4 codec. Unfortunately, using H.264's Baseline profile this option would not be available as B-frames are not supported. Data-partitioning, another means of prioritization, is also not supported, again to reduce potential computation on a mobile device. Therefore, if AQM is to be applied to the low-bit rates arising from H.264 another form of frame type or content prioritisation may need to be applied.

Turning to TFRC congestion control, in Fu et al. (2006), a reassembly failure at the Radio Link Control (RLC) layer signals such losses to a 3G User Equipment (UE) (the 3G equivalent of an SS). Feedback packets contain an estimate of the wireless channel packet loss rate. However, this approach (Fu et al., 2006) assumes an absence of congestion at the base station and requires cross-layer interaction. It is also reliant on the safe arrival of the feedback packets.

In Tappayuthpijam et al. (2009), TFRC was applied over LTE 4G broadband wireless technology. The rationale was that, in this type of mobile network, retransmissions at the data-link layer effectively remove packet loss at the expense of end-to-end delay and throughput. TFRC was combined with the Scalable Video Coding (SVC) extension to the H.264 codec, in such a way that the layering was adapted to the bitrate. The results are encouraging compared to not using TFRC in terms of reduced packet losses, reduction of streaming interruptions and end-to-end delay, and buffering levels. However, that approach has the potential to introduce unbounded delay across the wireless link, apart from the drop in throughput that results. This reintroduces the problems that led to the search for an alternative to TCP transport for media streaming.

WiMAX SCHEDULING

Background

WiMAX networks support multiple service classes to accommodate heterogeneous traffic with varying requirements. In particular, these are Unsolicited Grant Service (UGS), extended real-time Polling Service (ertPS), real-time Polling Service (rtPS), non-real-time Polling Service (nrtPS), and Best effort (BE). UGS is designed for synchronous services such as legacy voice without silence suppression. ertPS (Zhang et al. 2006) is intended for voice with silence suppression, when a SS may not have data available when polled by the BS. rtPS is most suitable for real-time video services, particularly for Variable Bitrate Video (VBR), which is employed to maintain delivered video quality but may lead to 'bursty' arrival rates. The rtPS class is the focus of our study.

The IEEE 802.16 standard in all its versions (parts a–e) does not actively specify queue management algorithms at the SS (or BS) (though there is a default round-robin queue service), allowing these to be designed by vendors as a means of commercial differentiation. Choice of queuing discipline is sensitive to application type and service class, whether it is implemented on the BS or the SS. Considerable attention continues to be given to WiMAX DL scheduling (Thaliath et al., 2008), which involves bandwidth allocation by the BS between different SS and between different service classes. However, less attention has been applied to efficiently utilizing uplink capacity. In the recent past, this may have been because WiMAX link traffic was expected to be asymmetrical but emerging media services may change that situation. Therefore, uplink media traffic now becomes of interest.

Several AQM schemes are considered in this paper. RED, as established in the Introduction, has a number of helpful features but it is known (Chen and Zakhor, 2006) to underperform when there are few coexisting flows in a queue, because buffer occupancy fluctuates rapidly. In an SS, the number of coexisting flows could well be limited. RED maintains two thresholds: a low threshold in which all arriving packets are accepted and a high threshold when all arriving packets are rejected. When buffer fullness is between the thresholds, RED is in the congested state in which packets are randomly dropped. However, RED may not be able to react quickly enough to 'bursty' traffic operating within this region. Unfortunately, if buffer size is increased to compensate, apart from the drain on energy at any mobile device arising from the increased buffer size, interactive media applications may suffer from a large buffer as a result of increased waiting times.

Random Exponential Marking (REM)'s packet marking probability (Athuraliya et al, 2001) grows considerably more aggressively than RED's. Its congestion measure is based on mismatch between input rate and link capacity and between queue length and target length, though in practice rate change is measured by the rate of change of the queue length. The aim of REM is to stabilize the queue length to a target which is independent of traffic arrival intensity and of RED's queue thresholds. Like REM, the BLUE algorithm (Feng et al., 2002) also decouples queue length from congestion management. Rather than queue occupancy, BLUE employs a history of the current buffer packet overflow rate and link utilization to form the packet dropping (marking) probability. By updating its estimates over time, BLUE is able to learn the correct form of queue regulation. BLUE can also randomize times between marking probability updates to avoid source synchronization.

Though not an AQM, drop-tail (FIFO) queue management (Clark, Shenker, & Zhang, 1992) has the advantage that it scales well and shares delay between different connections. However, drop-tail may lock-out some connections to the advantage of others occupying the buffer. It also may maintain buffer fullness for longer periods of time, as it only signals congestion when the buffer is full. Drop-tail queue management is widely deployed across the Internet's routers.

Subscriber station operation

At the SS, for reasons of send/receive separation, uplink capacity will always be available. As SSs may be mobile devices, to reduce energy consumption their buffer capacity may not be large, leading to a need for efficient management. Cicconetti et al. (2006) showed that delay arising at the SS is likely to be more than that at the BS queues, in part because of the bandwidth request mechanism.

In Figure 1, once a BS has allocated bandwidth to each SS (in WiBRO for simplicity the capacity is equally divided), each SS must manage its queue according to the data arrival rate from user applications. In Point-to-Multipoint (PMP) mode, there is no SS-to-SS communication unless it is via the BS and, therefore, uplink queue management and scheduling of bandwidth is also necessary when communicating between SSs via the BS.

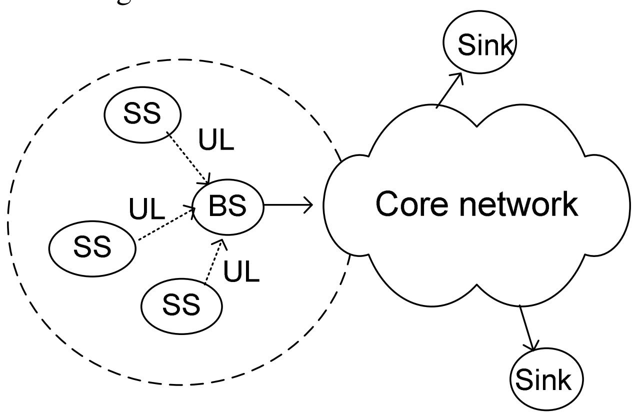

Figure 1. IEEE 802.16 uplink service architecture

Figure 2 shows how an SS communicates with a BS and how the uplink packet scheduling works across different service classes. An initial request by the SS results in a bandwidth allocation by the BS to that SS. The SS packet scheduler is able dynamically to request a modification to its bandwidth allocation in a piggybacked requested to the BS. As an SS may not have data available, it is polled by the BS. Polling may be on a group basis or individually. As group polling may result in access delay at the SS, this paper assumes unicast polling. The BS passes video and other data to the SS over the DL sub-frame whereas the UL sub-frame accommodates its traffic according to service class and burst slot availability.

The UGS is allocated bandwidth as of priority, while by default SS servicing of polled queues takes place in round-robin fashion. In Figure 2 the queues are serviced by the packet scheduler in the order indicated with BE traffic considered last. We assume an exhaustive service with the round-robin queue service by the SS packet scheduler.

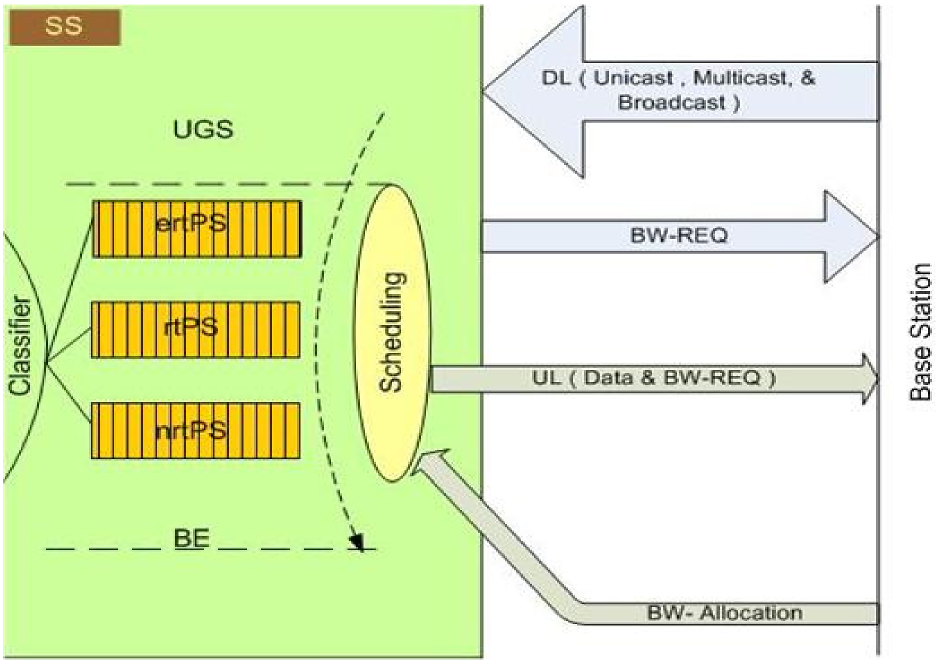

Figure 2. IEEE 802.16 uplink scheduling system.

METHODOLOGY

The WiMAX system operating in PMP mode was simulated by well-known ns-2 simulator (v. 2.29 used) augmented by a WiMAX module (Tsai et al. 2006) that has proved an effective way of modeling WiMAX's behavior. The PHY settings selected for WiMAX simulation are given in Table 1. The antenna was modeled for comparison purposes as a half-wavelength dipole. In WiMAX, the ratio of UL to DL is adaptive and in Table 1 it is assumed that the ratio has been temporarily adjusted for uplink communication. In fact, the WiMAX waveform can be changed from simplex to duplex, though simplex operation was not modeled. Current implementations have apparently mostly opted for a fixed 5 ms Time-Division Duplex (TDD) frame size, though this is difficult to verify, because of the commercial nature of the systems. Allowable frame lengths are specified in the Standard (IEEE, 2005), ranging from 2.5 to 20 ms. In the majority of our results a 5 ms TDD frame length was assumed. However, the longer 20 ms frame length was also tested, as this potentially allows more packets to be removed from queues at each SS polling event. With the frame length fixed at 20 ms, the UL portion of the frame length, that is 15 ms, is more than the common 5 ms frame length.

Parameter	Value
PHY	OFDMA
Frequency band	5 GHz
Duplexing mode	TDD
Frame length	5 , 20 ms
Max. packet length	1024 B
Raw data rate	10.67 Mbps
IFFT size	1024
Modulation	16-QAM 1/2
Guard band ratio	1/8
DL/UL ratio	1:3
DL length	1.25 to 5 ms
UL length	3.75 to 15 ms
MS transmit power	250 mW
BS transmit power	20 W
Approx. range to SS	0.7 km
Antenna type	Omni-directional
Antenna gains	0 dBD
MS antenna height	1.5 m
BS antenna height	32 m

Table 1. Simulated WiMAX settings.

The simulated configuration is shown in Figure 3. There were three SS communicating to the BS, with one of the SS sending VBR video encoded with the H.264/Advanced Video Codec (AVC) (Wiegand et al., 2003). Again this size of network was not intended to be indicative of the likely size of networks. Instead, just enough SSs (three) were provided to cause congestion to the video stream. Therefore, the likely size and capacity of likely WiMAX access networks is outside the scope of this paper, as the network setting is simply to test the video performance.

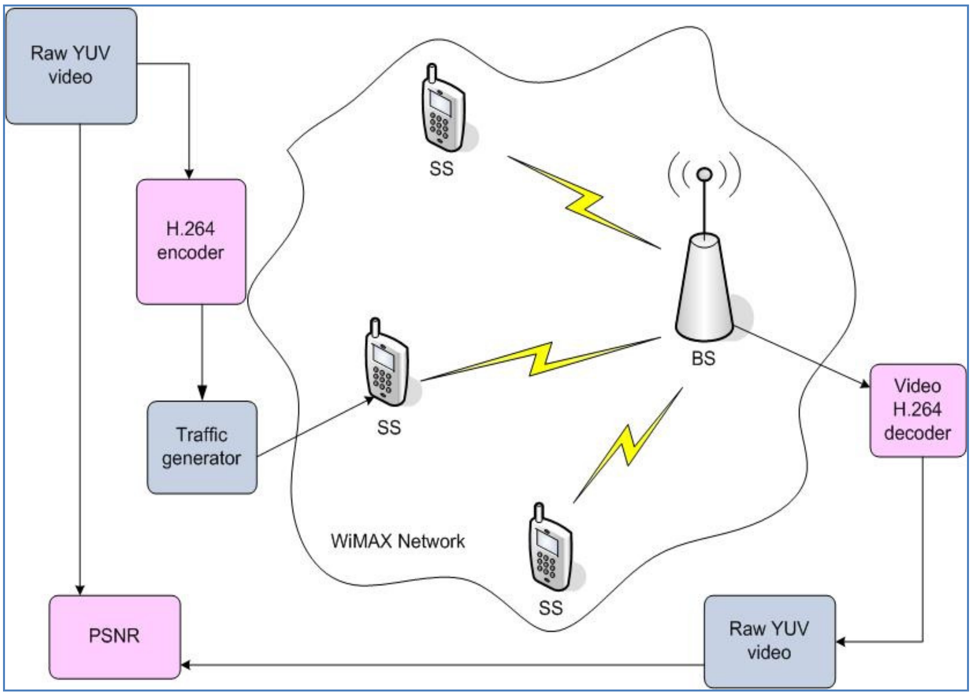

Figure 3. IEEE 802.16 video performance simulation.

A trace file was input to ns-2 and packet losses recorded in the output. The output was employed to form the Peak Signal-to-Noise Ratio (PSNR) as a measure of video quality. PSNR has the virtue that it is a logarithmic measure that accords with the eye's response to images. It is sufficient as a relative measure of quality across the same video clip. Video quality comparisons were made under the EvalVid environment (Klaue, Rathke, & Wolisz, 2003).

The reference 'Paris' clip H.264 VBR-encoded at 30 Hz (frame/s) at Common Intermediate Format (CIF) (352×288 pixel/frame). 1063 frames were transmitted (though graphs record 1000 of these frames). Paris consists of two figures seated around a table in TV studio format, with high spatial coding complexity. The Intra-refresh rate was every 15 frames with IPPP…I Group of Pictures (GoP) structure. The H.264 Baseline profile with reduced computational complexity is suitable for mobile devices. Within a Baseline profile GoP, B-frames are not permitted, though this increases the compressed bitrate.

Table 2 records the simulated traffic characteristics.As mentioned in the Introduction, the video source was transported in two different ways: with simple UDP packetisation and using TFRC. Network Adaptation Layer Units (NALUs) from H.264 were encapsulated with Real Time Protocol (RTP) headers on a single slice per frame basis. RTP includes a frame creation time-stamp allowing end-to-end delay to be estimated. After the addition of IP headers, these in turn formed a single WiMAX MAC Packet Data Unit (MPDU), which are variable-sized WiMAX packets. For simplicity, a WiMAX MPDU is now referred to as a packet. Packets were restricted to 1024 B, as occurs through WiMAX fragmentation. This implies that there is some risk of decoder desynchronisation if a slice is split into several packets, causing slice desynchronisation markers to be missing from some packets. Later tests imposed the 1024 B limit at the encoder.

This has the effect of placing one NALU in each packet, removing the risk of decoder desynchronization from fragmentation. Under UDP transport, packets were transmitted with an inter-packet gap of 0.03 s. For TFRC, the H.264 packet sizes were the same as for UDP but the inter-packet gap was varied according to the TFRC equation.

SS-UL	Service type	Traffic type	Protocol	Packet size (B)
1	rtPS	VBR (video)	UDP/ TFRC	Variable
		CBR	UDP	1000
	nrtPS	FTP	TCP	
2	rtPS	CBR	UDP	1000
	nrtPS	FTP	TCP	
3	rtPS	CBR	UDP	1000
	nrtPS	FTP	TCP	
SS-DL				
1,2, 3	rtPS	CBR	UDP	1000
	nrtPS	FTP	TCP	

Table 2. Simulated traffic characteristics.

As described in Handley et al. (2003), TFRC is a receiver-based system in which the packet loss rate is found at the receiver and fed-back to the sender in acknowledgment messages. The underlying TFRC transport protocol is UDP. The sender calculates the round-trip delay from the acknowledgment messages and updates the packet sending rate. An equation that models TCP New Reno is employed in TFRC to find the sending rate. In a variant to standard TFRC, the packet size in the TFRC equation was dynamically altered in our tests according to the EvalVid-created trace file sizes.

Coexisting rtPS queue CBR sources were all sent at 1500 kbps, i.e. the same target rate as the video source. The inter-packet gap was also 0.03 s for the CBR traffic. The FTP applications were set up out of convenience as a way of occupying the nrtPS queues; otherwise a BE queue might be more appropriate. Likewise, the DL traffic is simply selected to fully occupy the DL link capacity.

The buffer size was set to fifty packets. This is not so large as to cause long queuing delays and compatible with the value in other work on video over wireless (Li, & van der Schaar, 2004). Packets were dropped rather than explicitly marked at the queues, which will affect non-compliant applications, i.e. UDP, as well as compliant ones, i.e. TFRC. For RED, the dropping thresholds were set to 20% and 80% of the buffer size.

SIMULATION RESULTS

Video quality results

Tables 3 and 4 present mean video quality and end-to-end delay for UDP and TFRC transport. PSNR and delay were recorded on a per frame basis, with WiMAX range set to 0.7 km to emphasize the queuing component of delay rather than propagation time. From these Tables, it is apparent that TFRC definitely responds to packet drops at the rtPS queue and consequently improves the overall video quality. The mean delay for TFRC is also a little reduced compared to UDP transport, though the maximum delay is higher, due no doubt to TFRC managing its rate. REM's mean delay is less than the other queuing disciples suggesting a more aggressive regime and more packet drops, which is confirmed by the lower video quality whether under UDP or TFRC transport. For TFRC, drop-tail queuing is actually preferable to AQM queuing, though there is some advantage from AQM if UDP transport were to be used. From the Tables it is apparent that there is considerable variation in video quality on a frame-by-frame basis. The standard deviation (s.d.) is highest for TFRC under REM. The minimum per-frame PSNR values are unacceptable for viewing but the mean quality can be described as good, especially if received on a mobile station. However, our results do not take account of channel error, which would certainly reduce quality or increase delay (if some form of error control were to be applied).

Queue discipline	PSNR (dB)			End-to-end delay (s)	
	Mean	s.d.	Min.	Mean	Max.
Drop Tail	26.7	8.2	9.8	0.035	0.17
RED	29.8	9.1	10.7	0.035	0.17
REM	24.2	8.9	101	0.032	0.16
BLUE	29.8	9.1	10.7	0.036	0.16

Table 3. Overall video quality and latency for UDP.

Queue discipline	PSNR (dB)			End-to-end delay (s)	
	Mean	s.d.	Min.	Mean	Max.
Drop Tail	31.6	8.6	12.1	0.032	0.23
RED	31.4	8.8	12.2	0.033	0.24
REM	29.7	10.6	11.5	0.023	0.19
BLUE	31.4	8.8	12.2	0.033	0.24

Table 4. Overall video quality and latency for TFRC.

The packet drop figures were also compared, refer to Figure 4. As the total number of video packets sent was 4530, the dropped video packet numbers are around 10% of the whole. In video streaming, 10% packet loss is normally taken as recoverable from (Agboma, Smy, & Liotta, 2006). The number of packets lost by the TFRC stream was found to be greater than UDP for all queuing disciplines except REM. REM also balances its dropping rate between CBR sharing the rtPS queue and the video sources. However, REM's packet losses are relatively high. Other queuing disciplines have similar behaviors to each other, whether AQM or drop-tail.

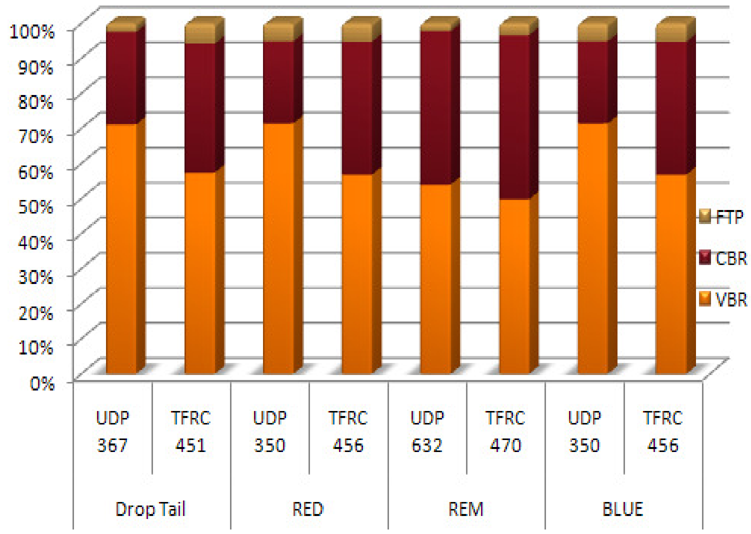

Figure 4. Percentage packet drops according to traffic type and transport for differing queuing disciplines.

Temporal analysis

We inspected the temporal behavior of the queuing disciplines. Figures 5 and 6 represent a comparison between drop-tail and the other queuing disciplines tested. The number of packets dropped over and above the buffer fullness limit of 50 packets is shown. The video was only turned on for part of the time shown, corresponding to 1063 frames. For the most part, the buffer fullness patterns are similar but there are times when packet losses for one or other of the queuing disciplines peaks. An obvious feature of the Figures is the oscillatory nature of buffer occupancy and packet drops, which to some extent is the result of the limited number of sources present at the SSs.

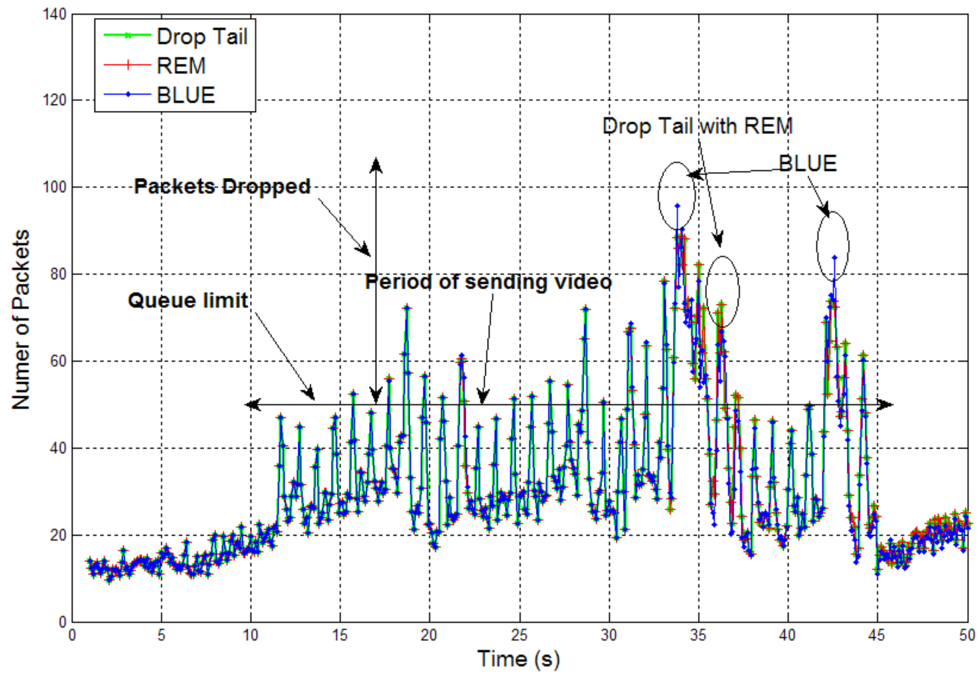

Figure 5. Buffer occupancy and packet drops for UDP transport, with drop-tail, REM and BLUE queuing disciplines.

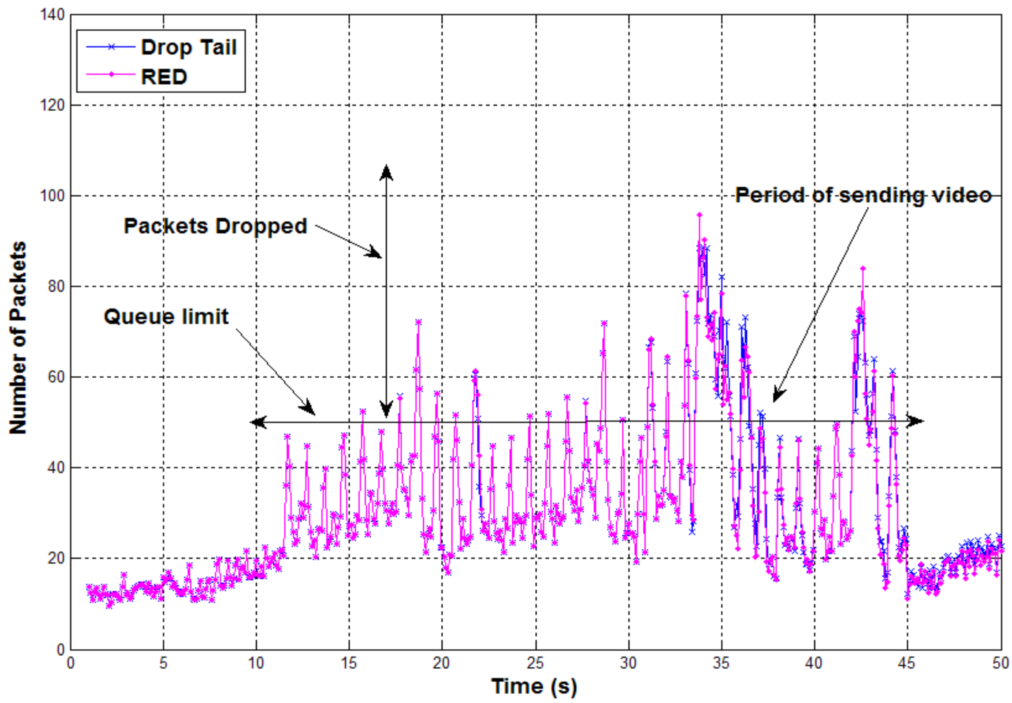

Figure 6. Buffer occupancy and packet drops for UDP transport, with drop-tail and RED queuing disciplines.

Figures 7 and 8 show that each queuing disciplines results in different onset times for drops in video quality. There is also an oscillatory nature to the quality which can partly be attributed to the intra-refresh rate. Another feature is that there is a gradual recovery from packet loss despite ‘bursty’ arrival rates.

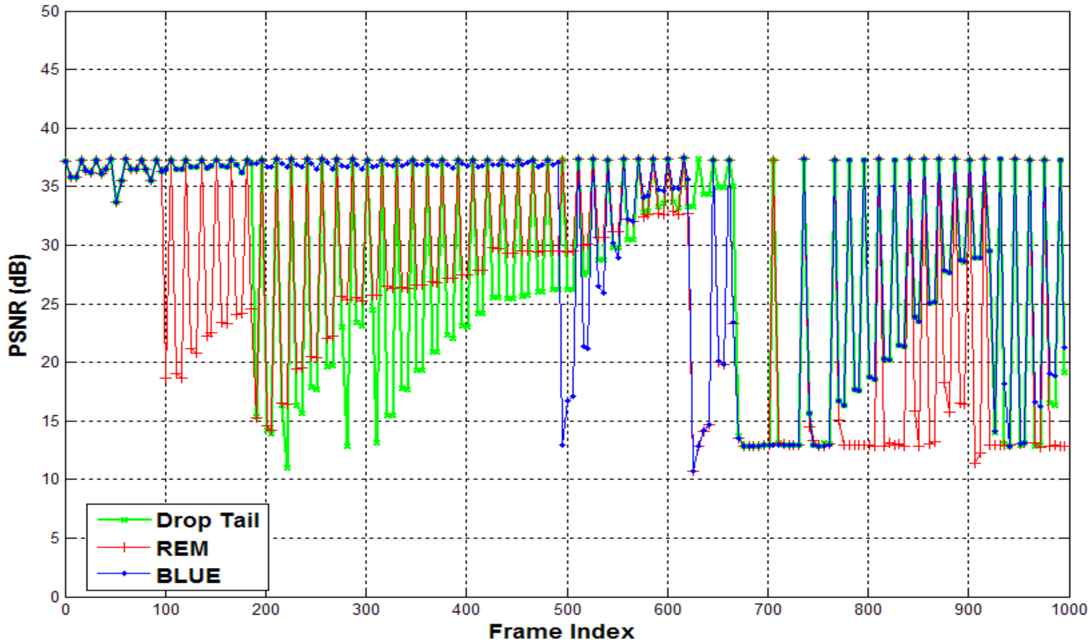

Figure 7. PSNR over time (frames) for UDP transport with drop-tail, REM and BLUE queuing disciplines.

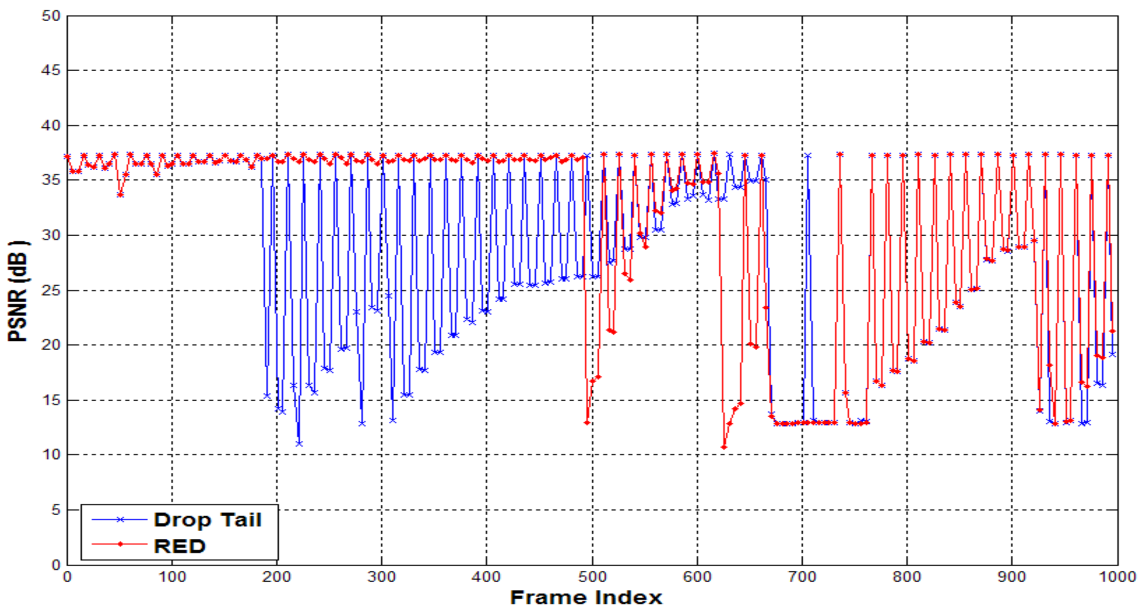

Figure 8. PSNR over time (frames) for UDP transport with drop-tail and RED queuing disciplines.

In Figures 9 and 10, the pattern of TFRC buffer occupancy is similar to UDP, but TFRC transport results in less excursions beyond the buffer limit, though the packet drop peaks are more severe than for UDP.

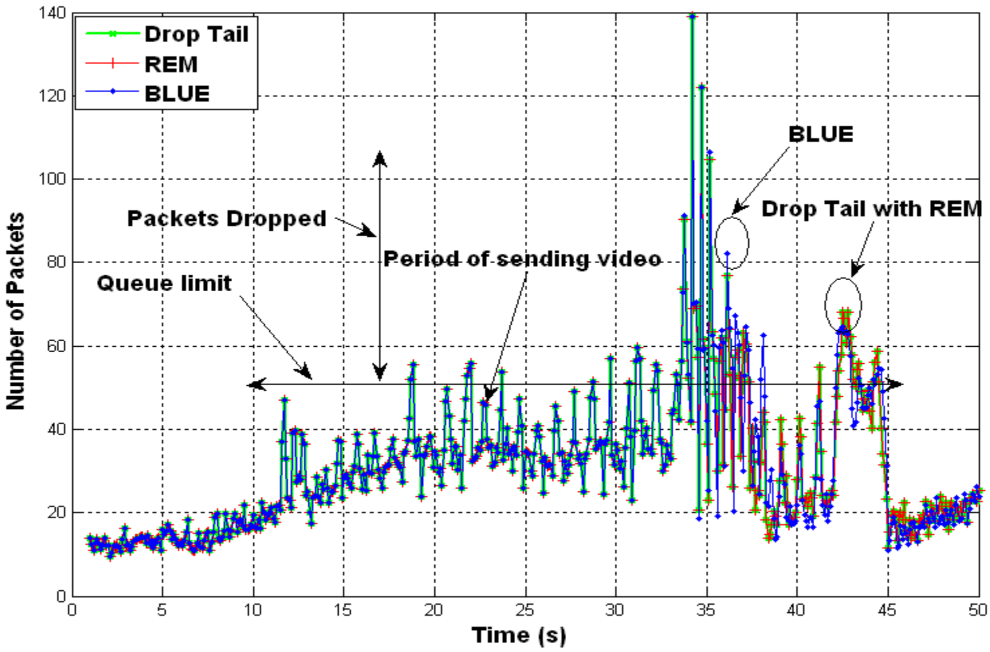

Figure 9. Buffer occupancy and packet drops for TFRC transport, with drop-tail, REM and BLUE queuing disciplines.

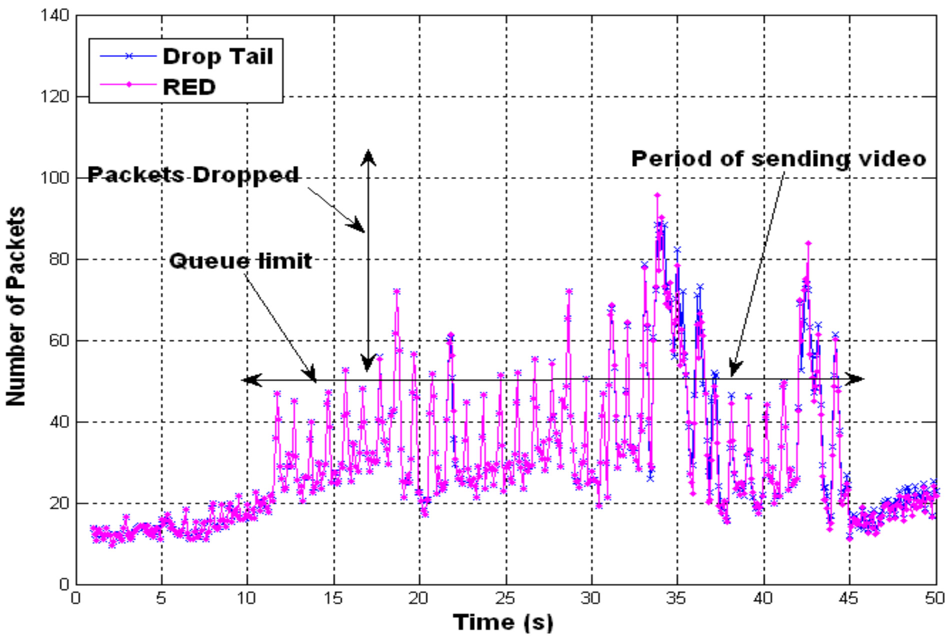

Figure 10. Buffer occupancy and packet drops for TFRC transport, with drop-tail and RED queuing disciplines.

Figures 11 and 12 show that delay builds up during the video sequence but that throughout end-to-end delay is oscillatory in nature. Because encoded I-frames tend to be larger than encoded P-frames, after packetisation to restrict the maximum packet size to 1000 B, I-frames tend to form longer bursts of packets.

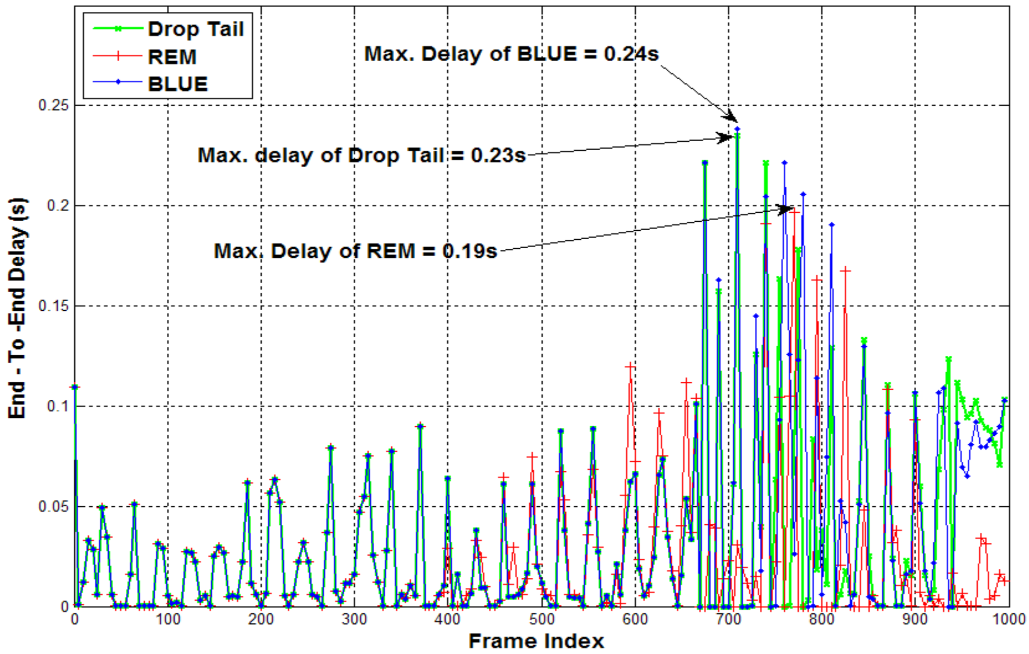

Figure 11. Delay over time (frames) for TFRC transport with drop-tail, REM and BLUE queuing disciplines.

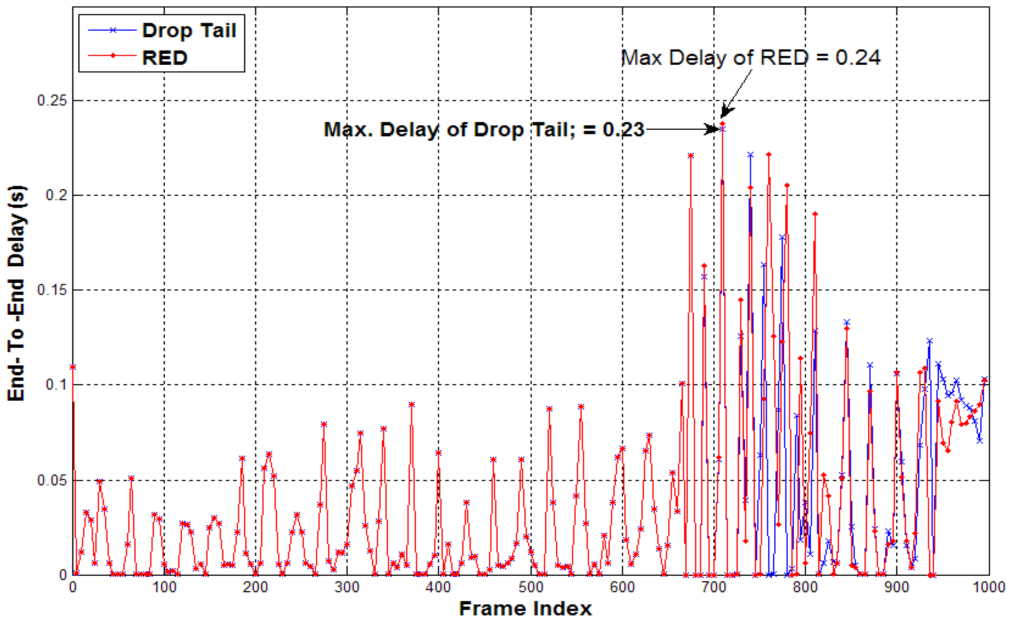

Figure 12. Delay over time (frames) for TFRC transport with drop-tail and RED queuing disciplines.

From Figure 13, the percentage of I-frame packets dropped is larger than the percentage of P-frame packets (though the total of these is larger because there are more P-frames). Clearly this is a problem for REM and employing TFRC tends to result in a greater percentage of I-frame packet drops. This suggests that a drop policy based on preserving I-frame packets would benefit TFRC transport and could be applied using either RED or BLUE.

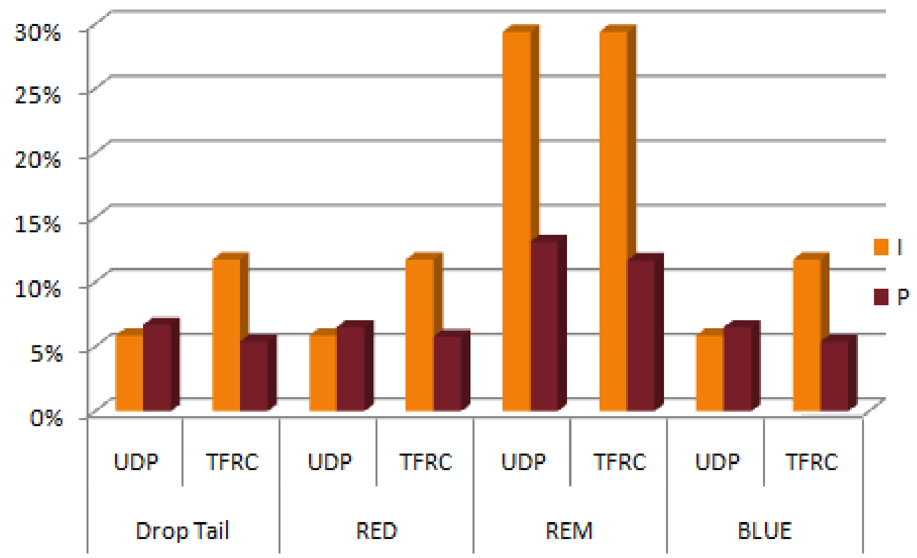

Figure 13. Percentage of I- and P-frame packets dropped relative to numbers of I- and P-frame packets respectively.

Possible improvements

The 'Paris' sequence was also coded with an IPBBP…I GoP structure. As before the refresh period is 15 frames and other coding conditions remained the same as before. As remarked earlier, including B-frames does not permit the use of the H.264 Baseline profile (the Main profile is suitable instead). Including, B-frames (strictly B-slices) allows a lower quality encoding, if the B-frames are not used as reference. Consequently, the bitrate is reduced if this option is taken. As a result, Table 5, for a 5 ms TDD frame, the s.d. is considerably reduced compared to the equivalent TFRC results in Table 4. This leads to an improvement in the per-frame minimum PSNR. The delivery was also improved by changing the method of packetization though not the packet size. However, this improvement is balanced out by the impact of the B-frames which tend to reduce the video quality. Video quality across the AQM methods is much the same. The main difference is that REM now benefits from the reduced bitrates and its relative video quality is now almost 2 dB better compared to Table 4. End-to-end delay for the results in Table 5 was much the same as in Table 4.

However, the main improvement in Table 5 can be seen when 20 ms TDD frames are deployed (with improved packetisation and changed GoP structure). At least as far as video traffic in this scenario is concerned, the ability to place more data onto a frame at each visit to the SS queue (as mentioned earlier) improves the quality by around 1 dB. The minimum per frame quality is also improved. However, a decision on which TDD frame length to use must reflect the traffic mix, not just the media traffic, and the number of SSs present.

Queue discipline	PSNR (dB) Frame length 5 ms			PSNR (dB) Frame length 20 ms		
	Mean	s. d.	Min.	Mean	s. d.	Min.
Drop Tail	31.2	4.4	21.4	31.9	3.8	22.8
RED	30.8	4.3	20.7	32.7	3.7	22.8
REM	31.9	4.2	22.3	33.3	3.3	25.3
BLUE	30.8	4.7	20.7	32.7	3.7	22.8

Table 4. Overall video quality for TFRC transport with change of packetisation and change of frame length.

CONCLUSIONS

This study has found that apart from REM, there is a similarity between drop-tail queuing and AQM for TFRC transport in terms of the resulting mean video quality, if the H.264 Baseline profile is employed. Despite the detractions of using congestion controllers originally designed for the wired Internet, some form of congestion control is needed for WiMAX queues. REM can be too aggressive, resulting in too many packet drops for bursty traffic during periods of congestion. The differences in the AQM mechanisms manifest themselves in different temporal behavior, with reductions in video quality or increases in delay occurring at different times. Unfortunately, precise analysis of temporal differences seems difficult, as it must take in a multitude of possible scenarios. In tests, all queuing disciplines caused oscillatory queue occupancy during periods of congestion, requiring jitter buffers to smooth out delays. The former may be a problem for media streaming sourced from WiMAX SSs, unless there is tighter application access policing to reduce the risk of congestion. Because I-frame packets tend to arrive in bursts, there is a benefit from a dropping policy that favors I-frame packets. However, dropping predictive P-frame packets also has an enduring impact and given that B-frame packets are not available for this purpose in the low-complexity Baseline profile, some other content dependent queue management is required.

However, it has been confirmed that if B-frames are used then the resulting drop in coding efficiency is compensated for by a reduction in video quality fluctuations. In other words, if it is possible to support the more complex Main profile then the viewing experience is likely to be more stable. In fact, in these circumstances REM's performance improved considerably. Another important observation is that the WiMAX system appears particularly sensitive to choice of TDD frame size, with advantage to be gained, at least as far as media streaming is concerned, by the use of a larger frame size. Future work will consider improvements to TFRC for WiMAX media streaming. If a dedicated WiMAX media service is supported it is also possible to tune configuration parameters such as the TDD frame size.

REFERENCES

Agboma, F., Smy, M., & Liotta, A. (2008). QoE analysis of a peer-to-peer television system. *IADIS Int. Conf. on Telecommunications, Networks and Systems*.

Ali, N.A., Dhrona, P. & Hassanein, H. (2009). A performance study of uplink scheduling algorithms in point-to-multipoint WiMAX networks. *Computer Communications*, 32(3), 511-521.

Athuraliya, S., Li, V.H., Low, S.H. & Yin, Q. (2001). REM: Active queue management. *IEEE Network*. 15(2), 48-53.

Chen, M. and Zakhor, A. (2006). Multiple TFRC connections based rate control for wireless networks. *IEEE Transactions on Multimedia*, 8(5), 1045-1061.

Cicconetti, C., Lenzini, L., Mingozzi, E., & Eklund, C. (2006). Quality of Service support in IEEE 802.16 networks, *IEEE Network*, 20(2), 50-55.

Cobb, J. A., & Agrawal, P. (1995). Congestion or corruption? A strategy for efficient wireless TCP sessions. *IEEE Symposium on Computers and Communications* (pp. 262-268).

Clark, D.D., Shenker, S., & Zhang, L. (1992). Supporting real-time applications in an Integrated Services packet network: Architecture and mechanism. *SIGCOMM '92* (pp. 14-26).

Cuny, R. & Lakkakorpi, P. (2006). Active Queue Management in EGPRS. *Vehicular Technology Conference* (pp. 373-377).

Ekstrom, K., Furuskar, A. Karlsson, J. Meyer, M. Parkvall, S. Torsner, J. & Wahlqvist, M. (2006). Technical solutions for the 3G Long-Term Evolution. *IEEE Communications Magazine*, 44(3), 38-45.

Feng, W.-C., Shin, K. G., Dilip, D. K, & Saha, D. (2002). Active queue management algorithms. *IEEE/ACM Transactions on Networking*, 10(4), 513-528.

Floyd, S. & Jacobson, V. (1993). Random Early Detection gateways for congestion avoidance. IEEE/ACM Transactions on Networking. 1(4), 397-413.

Fu, Y., Hu, R., Tian, G., & Wang, Z. (2006). TCP-Friendly Rate Control for streaming service over 3G Network. *International Conference on Wireless Communications, Networking and Mobile Computing* (pp. 1-4).

Görkemli, B., Sunay, M.O., & Tekalp, A.M. (2008). Video streaming over wireless DCCP. *IEEE International Conference on Image Processing* (pp. 2028—2031).

Handley, M., Pahdye, J.,Floyd, S. & Widmer, J. (2003). TCP-Friendly Rate Control (TFRC): Protocol specification. *Internet Engineering Task Force*, RFC 3448.

IEEE, 802.16e-2005. (2005). IEEE Standard for Local and Metropolitan Area Networks. Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems.

Kim, H., Lee, J. & Lee, B. G. (2008). WiBro – A 2.3 GHz mobile WiMAX: System design, network deployment, and services. In K.-C. Chen, and J. R. B. de Marca (Eds.), *Mobile WiMAX* (pp. 257-290). Wiley & Sons, Chichester, UK.

Kim, S. & Yeom, I, (2007). TCP-Aware uplink scheduling for IEEE 802.16. IEEE Communications Letters, 11(2), 146-148.

Klaue, J., Rathke, B. & Wolisz, A. (2003). EvalVid - A Framework for Video Transmission and Quality Evaluation*. International Conference on Modeling Techniques and Tools for Computer Performance* (pp. 255-272).

Koo, J., Ahn, S. & Chung, J. (2004). Performance analysis of Active Queue Management schemes for IP network. *International Conference on Computational Science*, (pp. 349-356).

Lakkakorpi, J., Sayenko, A., Karhula, J., Alanen, O. & Moilanen, J. (2007). Active queue management for reducing downlink delays in WiMAX. *Video Technology Conference* (pp. 326-330).

Li, Q. and van der Schaar, M. (2004). Providing QoS to layered video over wireless local area networks through real-time retry limit adaptation. *IEEE Transactions on Multimedia*, 6(2), 278-290.

Orlov, Z. & Necker, M. C. (2007). Enhancement of video streaming QoS with active buffer management in wireless environments. *13th European Wireless Conference*

Meloni, L. G. P. (2008). A new WiMAX profile for DTV return channel and wireless access. In K.-C. Chen, and J. R. B. de Marca (Eds.), *Mobile WiMAX* (pp. 291-392). Wiley & Sons, Chichester, UK.

Niyato, D., & Hossein, E. (2006). Queue-aware uplink bandwidth allocation and rate control for polling service in IEEE 802.16 broadband wireless networks. *IEEE Transactions on Mobile Computing*, 5(6), 668-679.

Raghu, K.R., Bose, S. K. & Ma, M. (2008). A queue based scheduling approach for WMAN with adaptive augmentation. *Wireless Communications and Networking Conference* (pp. 1374-1379).

Rahmani, R., Hjelm, M. & Aklund, C. (2008). Active Queue Management for TCP Friendly Rate Control traffic in heterogeneous networks. *International Conference on Telecommunications* (pp. 1-7).

Tappayuthpijam, K., Liebl, G., Stockhammer, T., & Steinbach, T. (2009). Adaptive video streaming over a mobile network with TCP-Friendly Rate Control. *International Conference on Wireless Communications and Mobile Computing* (pp. 1325-1329).

Thaliath, J., Joy, M. M., John, E. P. & Das, D. (2008). Service class downlink scheduling in WiMAX. *3rd International Conference on Communications Systems S/W and Middleware* (pp. 196-199).

Tsai, F. C.-D.., et al., (2006). The design and implementation of WiMAX module for ns-2 simulator. *Workshop on ns2*, article no. 5.

Wiegand, T., Sullivan, G. J., Bjøntegaard, G. and Luthra, A.: Overview of the H.264/AVC video coding standard, *IEEE Trans. on Circuits and Syst. for Video Technol,* 13(7), 560-576.

Zhang, H., Li, Y., Feng, S. & Wu, W. (2006). A new extended rtPS scheduling mechanism based on multi-polling for VoIP service in IEEE 802.16e system. *International Conference on Communication Technology* (pp. 1-4).