Improved Video Streaming Scheme for Internet Television with Broadband Wireless Access

Abstract: Internet Television video-streaming schemes do not respond well when access is through a broadband wireless link. A simple but effective negative acknowledgment-based scheme has been designed to meet the needs of Internet Television, one form of Internet Protocol Television (IPTV). When used with broadband Worldwide Interoperability for Microwave Access (WiMAX), simulations show that the scheme has the better features of both raw User Datagram Protocol (UDP) transport and the standardized Datagram Congestion Control Protocol (DCCP) but does not have those protocols disadvantages. The delivered video quality is also found to be superior to multiple-connection DCCP. One way that Internet Television will spread even more rapidly is if it is available to wireless devices as well as through existing conventional wired access. This is more likely to occur if an appropriate wireless video-streaming scheme is used, as video quality is a key determinant of customer choice.

Keywords: *DCCP, IEEE 802.16, Internet Television, IPTV, multi-connections, TFRC, video streaming, WiMAX*

1 Introduction

Internet Television (TV) is one form of Internet Protocol TV (IPTV) (Jain, 2005) that is delivered over a packet-switched network infrastructure by means of a combination of the wired Internet and broadband access networks. Internet TV with video streaming to a personal computer, TV screen, or mobile device across unmanaged networks seems to be gaining ground due to its flexibility, compared to standards-based delivery over a managed IP network (Maisonneuve et al., 2009) to a set-top box, an alternative form of IPTV. In Europe, the phasing out of analog TV from 2012, together with the increased capacity and penetration to households of broadband links is likely to lead to greater take-up of Internet TV. For example, in the UK, the BBC's iPlayer supporting live TV and time-shifted TV has been a success, whereas managed IPTV delivery is not so widespread. Acceptance of Internet TV service amongst customers as a hedonic or enjoyment-bringing technology (Weniger, 2010) is strongly influenced by perceived quality, which includes the perceived system quality, the subject of this paper, as well as other factors such as content quality, security of access and level of interactivity. A related analysis for mobile digital broadcast television was conducted in Shin (2008). One can also look to the experience of mobile video phone services, which according to Lee et al. (2010), is again influenced by the quality of phone calls, apart from additional functionality.

This paper considers broadband wireless access over IEEE 802.16e (mobile Worldwide Interoperability for Microwave Access (WiMAX)) (IEEE, 2005) rather than wired access, typically through Asymmetric Digital Subscriber Line (ADSL). Broadcasting over WiMAX to mobile devices (Kumar, 2008) is receiving active consideration from commercial interests. Mobile WiMAX provides broadband wireless access independently of a pre-existing cellular system, is not reliant on hardware authentication, and can deliver data in a cost-effective way at 3-4 times the rate of third generation (3G) cellular systems. It is particularly suitable for rural areas or where 3G coverage is sparse. A further extension of WiMAX, standardized as IEEE 802.16m (Ahmadi, S., 2011) — a 4th generation (4G) mobile system, will increase bandwidth capacity considerably, though it is not directly backwards compatible with earlier forms of WiMAX. The capacity of the IEEE 802.16e and the IEEE 802.16m variants of mobile WiMAX for transmission of unicast and multicast video has been examined in Oyman et al. (2010). The socio-technical reasons for the evolution of 4G systems are analyzed in Shin (2010).

However, it is important to solve a number of related problems concerned with the delivery of Internet TV over broadband wireless in order to ensure its perceived quality and it is the (partial) solution of these problems that is the theme of this paper. Internet TV unlike its managed counterpart does not guarantee video quality, due to the presence of congestion within an unmanaged network infrastructure. If congestion-induced packet drops are not repaired through retransmission or if excessive latency occurs through retransmission then video quality suffers. Compressed video streams are particularly vulnerable due to the data dependencies that exist between successive packets. There are also likely to be further problems if Internet TV is extended to mobile devices, because existing video-streaming schemes do not respond well when access is through a wireless broadband access network. In general this is because congestion control schemes are specialized for the wired Internet, in which packet drops are mostly through buffer overflow. However, packet loss on a wireless network can occur for a variety of other reasons such as interference, multipath, and radio frequency noise.

A further difficulty is that existing commercial video-streaming schemes, including the BBC iPlayer and YouTube's service for home video clips, depend on Adobe Flash Player technology, which employs block-based progressive download that is pseudo-streaming. Compressed video files, split into a sequence of blocks are delivered by Transmission Control Protocol (TCP) transport, as this protocol underlies Hypertext Transport Protocol (HTTP). TCP reacts to wireless packet loss as if it is buffer overflow due to congestion. As TCP is a reliable transport protocol, it responds to all packet loss by retransmissions, causing delays that appear as 'freeze frame' periods to the viewer. Consequently, there have been many attempts (Balkrishnan et al., 2007) to remedy TCP for delivery over heterogeneous networks, though some of these involve cross-layer intervention at a base station. In Gil et al. (2007), it was reported that 10% of viewers of YouTube on the wired Internet already may have interrupted downloads, due to the jerky nature of the video download, rather than due to poor content. YouTube clips are limited to ten minutes but iPlayer TV programs can be much longer, which suggests that dissatisfaction would grow if the service was transferred to mobile devices without modifying the means of delivery.

Assuming that TCP will not be used for delivery to mobile devices, for the reasons outlined previously and at the beginning of Section 2, it is legitimate to ask what alternatives to TCP exist and whether an improved Internet TV video-streaming control scheme is possible. The main objective of this paper is to show that our proposal of a simple but successful single retransmission scheme will be sufficient for Internet TV when mobile device receivers are targeted. The hope is that excessive delay does not occur but that the various weaknesses of existing transport protocols are avoided. The proposed scheme, based on negative acknowledgments (NACKs), has been simulated for a WiMAX channel with the objective of comparing its performance across a variety of performance metrics to: 1) User Datagram Protocol (UDP)-based streaming across a WiMAX link (Issa et al., 2010) and 2) to industry-standard Datagram Congestion Control Protocol (DCCP) (Kohler et al. 2006). DCCP adds connection set-up negotiation to the TCP-Friendly Rate Control (TFRC) (Handley et al., 2003) (Widmer et al., 2001). TFRC itself has even been adopted by Google's VideoTalk as a means of congestion control. To further increase wireless channel utilization in a heterogeneous environment, multi-connection TFRC is a third alternative means of video transport. However, this paper also aims to demonstrate that multi-connection TFRC is weaker in comparison to the proposed simple NACK-based scheme.

The remainder of the paper is organized as follows. Section 2 considers some related work in streamed video transport over IP networks. Section 3 describes the proposed scheme. The following Section introduces the simulation model we employed to demonstrate the proposed NACK scheme before turning to analysis of experimental results. Finally, the last two Sections present conclusions and consider future research.

2 Related work

Video streaming normally takes place either directly by UDP transport, see Issa et al. (2010) for a WiMAX example, or what is more likely through an application congestion controller (Widmer et al., 2001) that mimics the favorable features of TCP congestion control but avoids those features that are damaging to video streaming. When TCP transport of video streaming is retained, as it is in some commercial implementations (Wang et al., 2008), it may be because of the need to avoid UDP blocking at firewalls. TCP provides a reliable service but this very reliability may introduce delays which prevent video rate display unless large start-up delay is tolerated. However, for mobile devices, the need to smooth-out delays by buffering is also an impediment to using TCP, as large buffers are a significant source of energy consumption (Segars, 2001), and, thus, are likely to lead to reduction in battery life. Consequently, the proposed simple NACK-based scheme and the delivery options to which it is compared are application layer control programs with underlying UDP transport.

In Degrande et al. (2008), ways to improve IPTV quality were discussed with the assumption that intelligent content management would bring popular video content nearer to the end viewer. In Degrande et al. (2008), the advantage of intelligent content management is that it supports rapid TV channel swapping. However, the quality of the video at the receiver (Weniger, 2010) results in greater take-up of IPTV services, though notice that Weniger (2010) was concerned with managed IPTV rather than Internet TV. If the round-trip-time is reduced as a consequence of intelligent content management then the proposed NACK-based scheme becomes attractive as a way of mitigating channel errors and improving quality.

Alternatively, Forward Error Correction (FEC) imposes an overhead, especially when channel conditions are actually favorable. Wireless networks are well-known to be prone to burst errors (Yin et al., 2006). Depending on the length of bursts it is possible that even if application-layer FEC supplements existing Physical (PHY) layer FEC and packet or byte interleaving is used, this protection will still be insufficient. For example, in Liu et al. (2006) burst errors were modeled by the Gilbert-Elliott channel model for packet loss. It was shown that if the average burst length was two, then 12.5% of the packet losses occur in burst lengths longer than three, and if the average burst length is equal to four then 17.9% of packet losses occur in burst lengths longer than six. Therefore, when the delay from retransmitting a packet is tolerable, as it is with intelligent content management, error control through NACKs may well be preferable to the use of FEC.

Other responses to wireless conditions include the adaptation of existing transport protocols such TFRC (Handley et al., 2003), which was originally intended for congestion control of real-time streams on the wired Internet. In Tappayuthpijam et al. (2009), single-connection TFRC was advocated for the emerging Long Term Evolution (LTE) cellular broadband system. It was assumed that the data-link layer transparently retransmits packets until successful receipt occurs. The potential problem that this approach presents is that the video application loses control of packet latency, which implies that the mobile device will require large buffers to compensate, with resulting lengthy start-up times, beyond the maximum of 10 s recommended by the International Telecommunication Union (ITU) in recommendation G.1010.

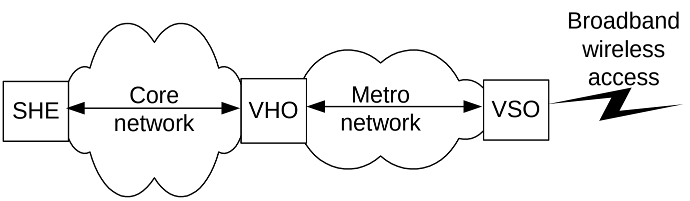
Several other attempts (Görkemli et al., 2008; Fu et al., 2006) have tried to improve TFRC's utilization of the wireless channel, which reduces sharply when channel packet loss occurs. The result can be that considerable interruption to the stream may occur if packet losses cause TFRC to reduce its streaming rate without also reducing the video quality. In Görkemli et al. (2008) and Fu et al. (2006), cross-layer intervention occurs in one way or another to mask channel packet loss from TFRC. Cross-layer intervention is effective if video streaming is the only application supported by the network but it is less attractive if other types of application traffic (data, voice, voice with silence suppression, on-line games …) share the network, as each application might need to be handled differently.

3 Proposed NACK-based scheme

A typical IPTV content-distribution architecture (Degrande et al., 2008), Figure 1, assumes a super head-end (SHE) distributor of content across a core network to regional video hub offices (VHOs). VHOs are connected to video serving offices (VSOs) over a regional metro network. It is a VSO that interacts with users over an access network. In this paper, IEEE 802.16e (mobile WiMAX) (IEEE, 2005) represents the broadband wireless access stage, while streaming experiments with the proposed NACK scheme either take place assuming the VSO is the source or occur sourced from the VHO over the metro network, after distribution of content from the SHE. The metro network is modelled as an IP network, either with best-effort IP-framed delivery of packets or over a managed network. In the former case, one has Internet TV streaming, as introduced in Section 1 as the most demanding of IPTV systems.

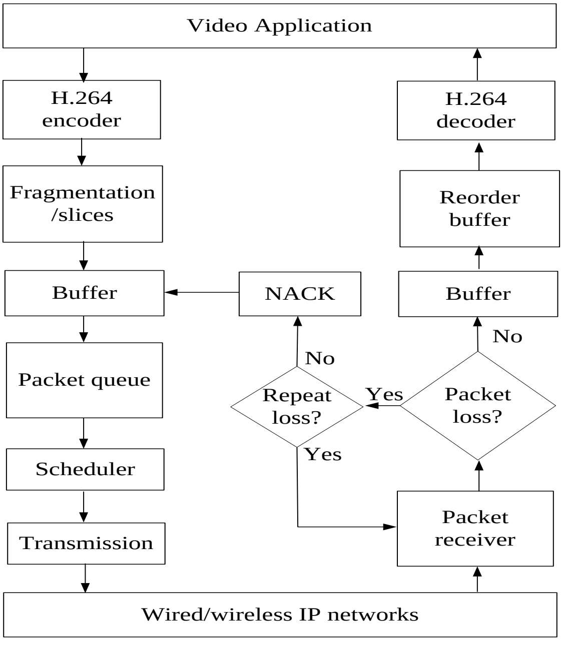
Considering the two examples of video streaming mentioned in Section 1, the BBC iPlayer originally employed a peer-to-peer (P2P) client embedded in each destination machine but was forced to change after Internet Service Providers (ISPs) complained about the increase in traffic and after some destination machines were slowed down by the need to act as P2P hosts. YouTube is known (Gil et al., 2007) to employ a content-distribution network but clearly video streams pass over an unmanaged network, because of the congestion-related problems reported in Section 1.

Figure 1 Schematic IPTV distribution network



Poor wireless channel utilization from conventional rate controllers is an important concern (Chen and Zakhor, 2006) when streaming video. After observing that UDP at least succeeds in good wireless channel utilization, without any protection from channel loss, we introduced the simple NACK-based scheme with its single negative acknowledgment as an addition to UDP. Figure 2 is a representation of the processing involved, showing the NACK response of the receiver. At a mobile station (MS), a record is kept of packet sequence numbers available through the Real Time Protocol (RTP) header and if an out-of-sequence packet arrives a NACK is transmitted to the base station (BS) in the next sub-frame. The BS prevents transmission from its input buffer until a single retransmission of the missing packet in the sequence has taken place. If the missing packet fails to arrive at the BS in the next WiMAX Time Division Duplex (TDD) frame (Andrews et al., 2007) then the BS continues its transmissions with the next packet in sequence. Further retransmissions do not take place, as waiting packets could be delayed and because the failure of one retransmission may indicate continuing poor channel conditions.

Figure 2 Operation of the NACK-based scheme



4 Experimental results

This Section now evaluates the NACK-based scheme against direct UDP transport, congestion-controlled transport, and multi-connection transport (refere to Section 1). We have chosen to compare with TFRC-based multi-connection streaming, as the single connection congestion controller chosen is also TFRC (Handley et al., 2003). However, it is worth noting that there are also TCP-based approaches to multi-connection streaming such as multTCP (Tullimas et al., 2008).

4.1. Simulation model

The WiMAX system operating in point-to-multipoint mode was simulated by the well-known ns-2 simulator (v. 2.29 employed) augmented by a WiMAX module (Tsai et al., 2006). Mean data points reported in the evaluation section are the arithmetic mean of twenty-five runs. These points were found with 95% confidence to be statistically independent of equivalent points. The simulator was allowed to reach steady-state over 20 s before commencing simulated video streaming.

The PHY settings selected for WiMAX simulation are given in Table 1, with additional Medium Access Control (MAC) settings defaulted from Tsai et al. (2006). The antenna is modeled for comparison purposes as a half-wavelength dipole. The frame length is significant, as a longer frame reduces delay at the mobile SS by permitting more data to be removed from any queues at each polling time. The value of 20 ms selected is at the high end of the available TDD frame durations in the Standard (IEEE, 2005). The buffer sizes at the base station and mobile station were set to the ns-2 default of fifty packets, because as mentioned in Section 1, larger buffers are a potential source of excessive energy consumption in mobile devices. In WiMAX and for the purposes of the simulation, a packet corresponds to a single MAC Service Data Unit (MSDU) within a MAC Protocol Data Unit (MPDU).

A trace file of the video packet lengths and transmission times was input to ns-2 and packet losses recorded in the output. The output serves to calculate the objective video quality (Peak Signal-to-Noise Ratio (PSNR)). Video quality comparisons were made under the EvalVid environment (Klaue et al., 2003). As a test, we used the *Paris* sequence, H.264/AVC (Advanced Video Coding) (Wiegand et al., 2003) Variable Bit-Rate (VBR)-encoded at 30 frame/s, with Common Intermediate Format (CIF) (352 × 288 pixel/frame) and with quantization parameter (QP) set to 26 (from a range 0 to 51). The PSNR for this sequence without packet loss but after decoding is 38 dB. The slice size was fixed at the encoder at 900 B. In this way, network segmentation of the packet was prevented. This avoids the loss of synchronization information at the decoder, which otherwise might arise from splitting a slice and losing its header.

Paris consists of two figures seated round a table in a TV studio setting, with high spatial coding complexity and moderate motion. Quality-of-Experience tests show (Agboma and Liotta, 2007) that this type of content is favored by users of mobile devices. This content does not stretch the capabilities of the screen display (as for instance sport sequences would do because of small rapidly moving objects) and it can be appreciated so long as the sound quality is sufficient. The Intra-refresh rate was every 15 frames with a standard IBBP…I coding structure. 1065 frames were transmitted resulting in a video duration of 35.5 s. Previous frame replacement was set for error concealment at the decoder, as is normal and as a point of comparison with others' work.

Table 1 Simulated WiMAX settings

Parameter	*Value*
PHY	OFDMA
Frequency band	5 GHz
Duplexing mode	TDD
Frame length	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	3:1
Path loss model	Two-ray ground
Channel propagation model	Gilbert-Elliott
SS transmit power	250 mW
BS transmit power	20 W
Approx. range to MS	0.7 km
Antenna type	Omni-directional
Antenna gains	0 dBD
SS antenna height	1.5 m
BS antenna height	32 m

OFDMA = Orthogonal Frequency Division Multiple Access, QAM = Quadrature Amplitude Modulation, TDD = Time Division Duplex, IFFT = Inverse Fast Fourier Transform, DL = Downlink, UL = Uplink, MS = Mobile Station, BS = Base Station.

A Gilbert-Elliott two-state, discrete-time, ergodic Markov chain (Haßlinger, and Hohlfeld, 2008) modeled the wireless channel error characteristics at the ns-2 physical layer. A two-state model reproduces conditions experienced during fast fading implying the model is valid for nearby mobile nodes. The relatively short duration of the video sequence means that slow fades are unlikely to occur during the test period. The probability of remaining in the Gilbert-Elliott good state was set to 0.95 and of remaining in the bad state was 0.94, with both states modeled by a Uniform distribution. The packet loss probability in the good state was fixed at 0.01 and the bad state default was 0.05. However, the bad state packet loss probability, P_B, was also varied as [0.01, 0.05, 0.1 , …, 0.2]. In this way, we were able to judge the effect of worsening burst error channel conditions.

4.2 Performance evaluation

In this Section, the NACK scheme is evaluated by bracketing its performance between that of UDP transport (Issa et al., 2010) and that of DCCP (Kohler et al., 2006). It is interesting to see how close to UDP's performance the scheme can approach without displaying the weaknesses of UDP. For naming convenience in the graphs, the simple NACK-scheme is referred to as broadband video streaming (BVS).

Figure 3 shows summary statistics of the packet loss percentages. Because the NACK scheme sends a NACK after each packet loss, packets are re-sent and marked as received without loss if they survive the second sending. Naturally, UDP transport simply results in the packets being transmitted according to the encoder schedule without congestion control. Therefore, its packet loss percentage is simply a reaction to the increasing error rate. However, what may be surprising is that DCCP delivery, when exposed to a wireless environment, barely reduces the packet error rate and can even result in an increase in total packet loss. This is because DCCP, in reaction to packet loss, increases the inter-packet gap at the sender. However, this increase has the effect of extending the duration that the video stream is exposed to channel error, though individual packets are sent more sparsely during this period. The resulting video quality (PSNR), Figure 4, for the NACK scheme (BVS) is over 5 dB higher than that resulting from UDP and DCCP transport.

From Figure 5, mean throughput (packets sent) under UDP approximates to the VBR encoding rate, which is about 820 kbps. As packet errors increase, the NACK scheme's throughput diverges from that of UDP, as a consequence of retransmissions. In contrast, DCCP's throughput dramatically decreases as the errors increase. The DCCP result at a P_B of 0.2

seems anomalous (see Figures 3 and 4 also). However, the result is more likely to be a result of the DCCP packet sending pattern than lack of convergence, as the other plots do not behave in that way.

Figure 3 Packet loss rate for the *Paris* sequence, according to WiMAX channel conditions

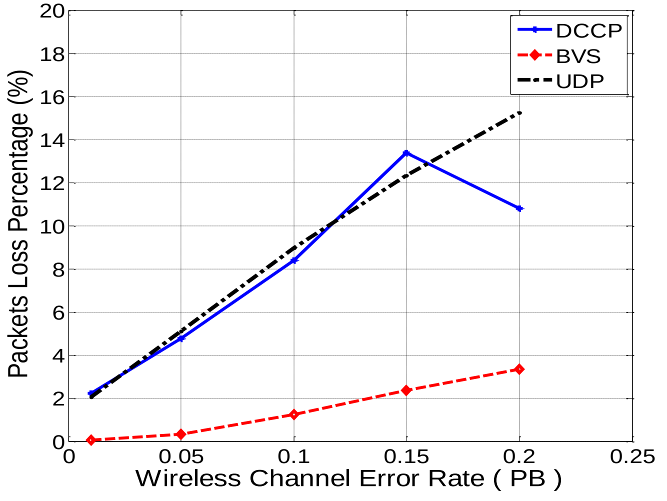


Figure 4 Objective video quality for the *Paris* sequence, according to WiMAX channel conditions

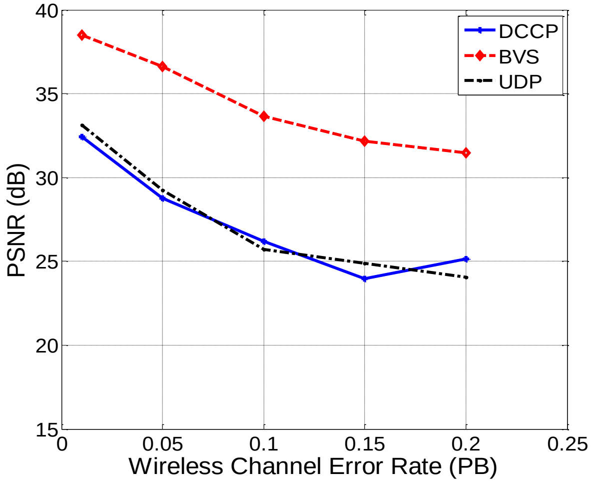
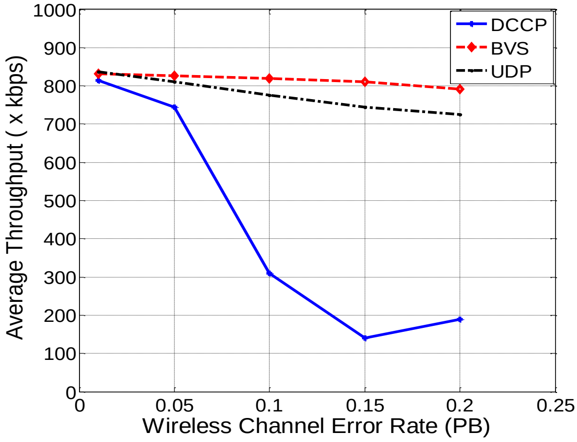


Figure 5 Throughput for the *Paris* sequence, according to WiMAX channel conditions



In the following comparisons, P_B in the Gilbert-Elliott model was set to 0.15. In Figure 6, the simple NACK-based scheme (labelled as BVS) receiving rate over time lags the sending rate so that peaks in receiving rate due to retransmissions occur after those of the sending rate. It is important to notice that the sending rate does not far exceed the receiving rate. Therefore, the overhead from the scheme in these error conditions is limited. Turning to the behavior over time of the three transport methods, one sees from Figure 7 that DCCP throttles its sending rate to such an extent that the sending time is considerably greater than the viewing time of the video, inevitably leading to underflow at the receiving buffer unless it is very large. Notice that in Figure 7, the sending time commences at 24 s (not at 0 s) and thus the receiving time for UDP and the simple NACK scheme is close to the sequence duration, whereas DCCP's sending period is several times the sequence duration. Clearly this is unsatisfactory and implies that unmodified DCCP should only be used in a wireless environment at low error rates, as otherwise the time to send the video sequence will exceed the display time of the sequence.

Figure 6 Simple NACK-based scheme sending and receiving rates over time, with P_B= 0.15

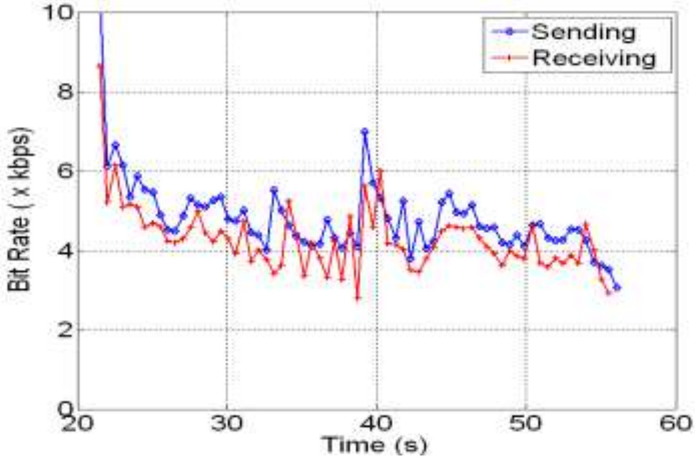
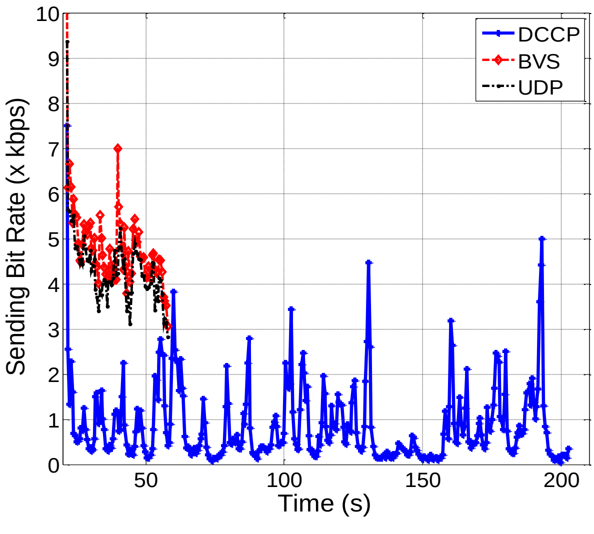


Figure 7 Sending bit-rate over time with P_B = 0.15 for the transport methods



From Table 2's summary, the main penalty of the NACK-based scheme is the increased end-to-end packet delay, mostly caused by queuing delay through self-congestion. However, for video streaming, an end-to-end delay of about 50 ms is not significant. In fact, DCCP's mean end-to-end delay is low, because there is little or no queuing delay included, as the result of DCCP increasing inter-packet gaps is that queues do not build-up in the sending buffer. Maximum delay (interruption) for the NACK-based scheme is about 200 ms but this is three times less than that of DCCP. In fact, the interruption for TFRC reported in Tappayuthpijam et al. (2009) for streaming over LTE was 1.85 s and 0.99 s respectively for the two scenarios tested. However, sending time was not reported, which is unfortunate, as from Table 2 it is the frequency of long interruptions, rather than their maximum duration, that builds up to form TFRC's excessive streaming durations.

In a randomly selected test, the NACK-based scheme's peaks in jitter and end-to-end delay were plotted over time, as plotted respectively in Figures 8 and 9. From both plots it is apparent that there are few very high spikes in the plots. However, for jitter this is not the case for DCCP, as Figure 10 illustrates. This contributes to the build-up in sending time. Notice that in Figure 8 there is negative going jitter. This is explained by the arrival of out-of-order retransmitted packets, rather than packets that arrive in order but are separated from the previous arrival by a positive amount. The reduced number of packets in Figure 10 compared to Figure 8 is explained by the increased number of packet drops when streaming with DCCP.

Table 2. Comparison summary at $P_B = 0.15$

	DCCP	*BVS*	*UDP*
Throughput (kbps)	140	811	745
Loss (%)	13.3	2.36	12.32
End-to-end delay (s)	0.009	0.051	0.041
Interruption (s)	0.789	0.201	0.071
PSNR (dB)	23.95	32.18	24.87
Jitter (s)	0.044	0.007	0.008
Sending period (s)	182.0	35.5	35.5

4.3 Comparison with multi-connection streaming

One way that the performance of DCCP might be improved without the need for cross-layer intervention is through multi-connection TFRC (Chen and Zakhor, 2006). In this way, wireless channel utilization of TFRC single connection may be improved, because if one connection's throughput is reduced due to channel losses then another connection can compensate. In Al-Majeed and Fleury (2010), it was found that, for a small numbers of connections, this seemed to be the case and, in the next set of comparisons, four-connection TFRC according to the scheme of Al-Majeed and Fleury (2010) was employed. Of course, apart from connection negotiation, multiple TFRC is equivalent to multiple DCCP and, in fact, connection negotiation time is excluded from the experimental results.

Figure 8 Jitter over time for a sample run streaming *Paris* under the NACK scheme (BVS)

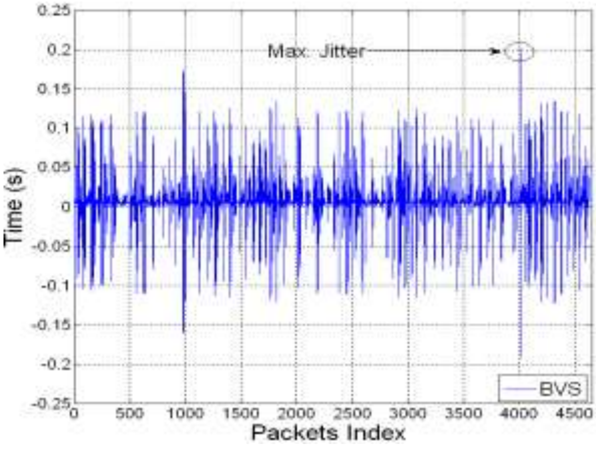


Figure 9 End-to-end packet delays over time for a sample run streaming *Paris* under the NACK scheme (BVS)

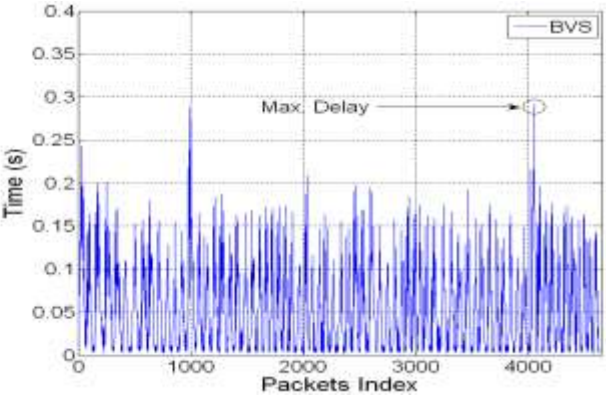
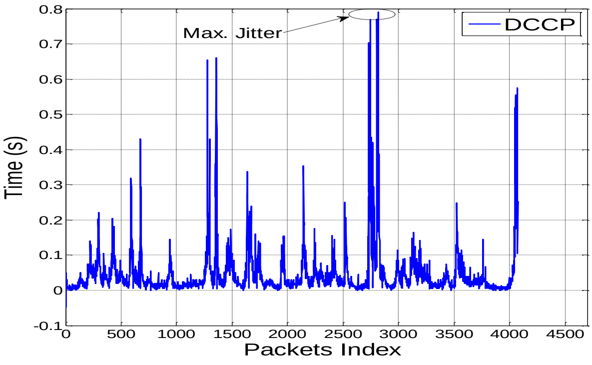


Figure 10 Jitter over time for a sample run streaming *Paris* under DCCP



In Figure 11, it can be seen that multiple TFRC/DCCP does improve wireless channel utilization but it does this by streaming the bulk of the data at the start, when all connections are operating. However, because some connections are slowed down by the congestion controller as packet losses occur, the later part of the streaming session results in similar throughput to single-connection streaming. In Figure 12, multiple TFRC/DCCP experiences a similar percentage of errors as single-connection DCCP encounters. Intuitively, this is not surprising, as the error conditions are experienced by all connections at the same time and the error conditions are, in the mean, statistically similar over any period of time. Consequently, video quality is similarly below that achievable with the simple NACK scheme, as Figure 13 confirms.

Figure 11 Throughput with multiple connection TFRC for the *Paris* sequence compared to the simple NACK scheme (BVS) according to WiMAX channel conditions

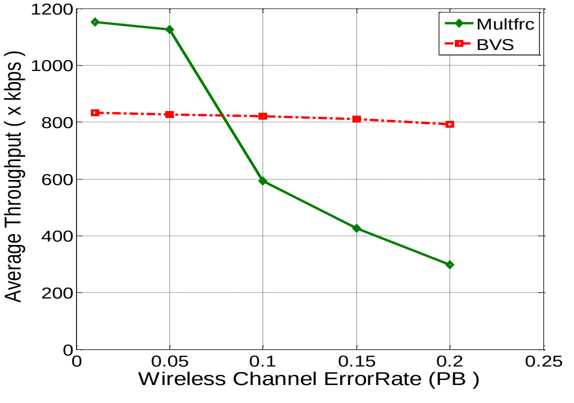


Figure 12 Packet loss percentage with multiple connection TFRC for the *Paris* sequence compared to the simple NACK scheme (BVS) according to WiMAX channel conditions

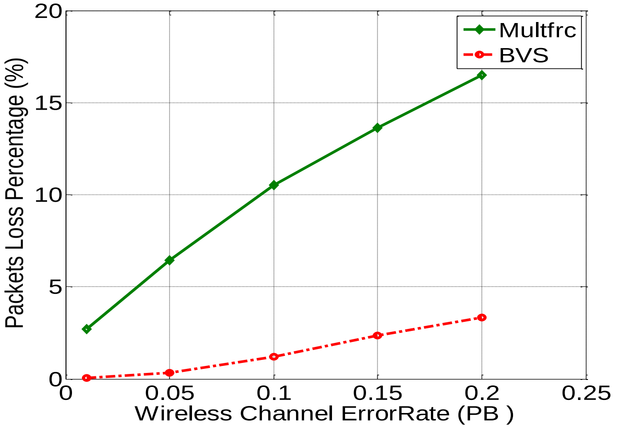
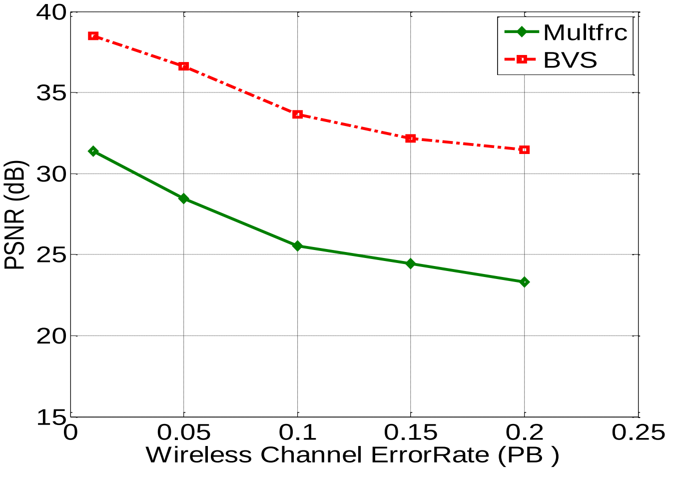
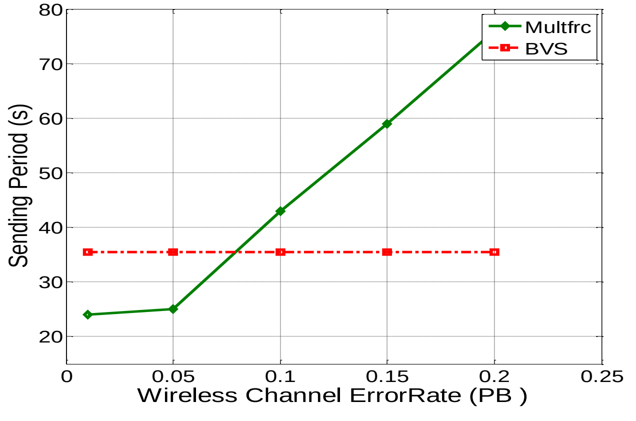


Figure 13 Video quality with multiple connection TFRC for the *Paris* sequence compared to the simple NACK scheme (BVS) according to WiMAX channel conditions



It may be asked "how does multiple TFRC represent an improvement?" In Figure 14, the sending periods for multiple TFRC/DCCP and the NACK scheme are compared. The NACK scheme maintains the sending period of the sequence despite some retransmissions but multiple TFRC/DCCP greatly exceeds the sending period once the error rate increases. In sending periods below that of the NACK scheme, multiple TFRC/DCCP (if streaming from the base station) would require buffering (and a reorder buffer as well (***, 2010)), while for PB= 0.1 there would be tolerable start-up delay. Thus, in lower error rate regimes, multiple TFRC/DCCP may result in a 'fair' video quality (above 25 dB) and a passable sending period.

Figure 14 Sending period with multiple connection TFRC for the *Paris* sequence compared to the simple NACK scheme (BVS) according to WiMAX channel conditions

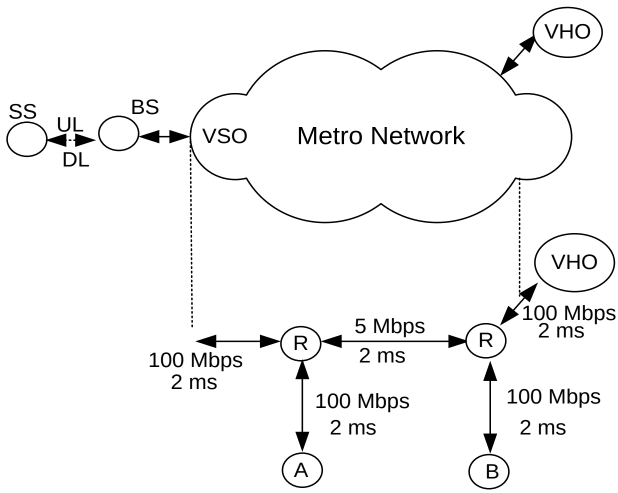


4.4 Performance over a heterogeneous network

This Section considers the NACK scheme's performance when there is a remote video server, that is the Video Hub Office (VHO) in Figure 15, connected over a metro network to the WiMAX radio access network, where access to the base station (BS) is via the Video Serving Office (VSO) (for VHO and VSO refer back to Figure 1). The BS streams video to the MS. We also considered uplink streaming in the reverse direction from the MS to the VHO. Uplink streaming might be used for live TV news reporting. There is also much interest (Oyman et al., 2010) in uplink streaming for future interactive video services.

In the Figure, all links except a bottleneck link within the core network are set to 100 Mbps to easily accommodate the traffic flows entering and leaving the network. The link delays are minimal (2 ms) in order in the results to avoid confusing propagation delay with re-ordering delay. A bottleneck link with capacity 5 Mbps was set up between the two routers. The buffer size in each router was set to 50 packets. This arrangement is not meant to physically correspond to a network layout but to represent the type of bottleneck that commonly lies at a Metro network edge. Node A sources to node B a CBR stream at 1.5 Mbps with packet size 1 kB and sinks a continuous TCP File Transfer Protocol (FTP) flow sourced at node B. Node B also sources an FTP flow to the BS and a Constant Bit-Rate (CBR) stream at 1.5 Mbps with packet size 1 kB. These traffic flows represent traffic congestion within the Metro network.

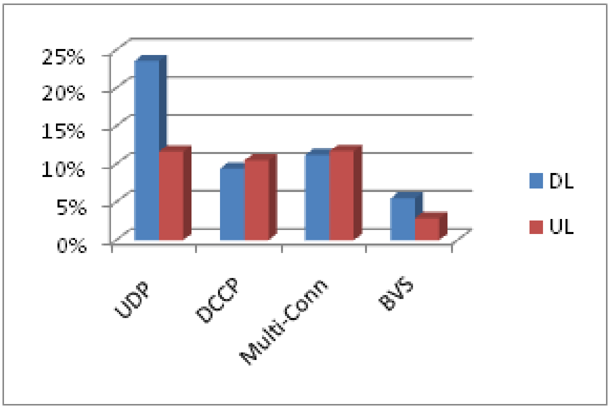
Figure 15 Network scenario with inset showing routing across the core network, A, B and the VHO being sources and sinks, and R = router



An interesting feature of the results is the asymmetry between uplink and downlink performance that can arise. This is because in uplink streaming more packets may be lost traversing the wireless link compared to downlink streaming across a congested core network. Once the wireless link is crossed, for uplink streaming the stream is less likely to suffer loss from congestion. This is because its packet arrival rate has already been reduced by losses arising from wireless channel conditions.

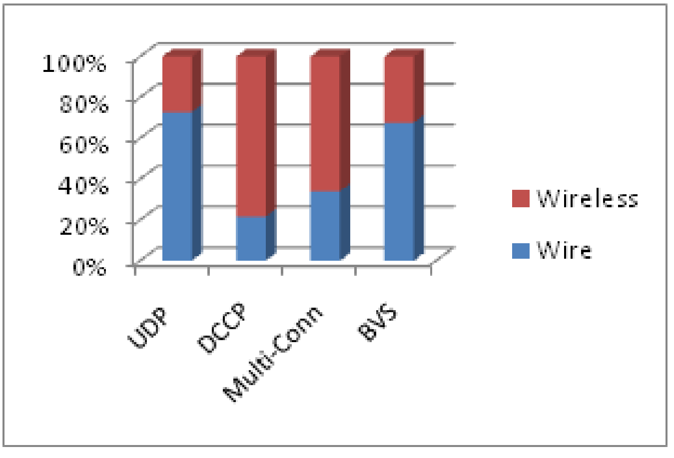
In Figure 16, UDP streaming suffers unacceptable packet losses (above 10%) in the downlink streaming direction because the stream not only suffers some losses due to congestion as it enters the buffers of the two routers but this further loss occurs across the WiMAX link. For uplink streaming packet losses from congestion are reduced, for the reasons previously discussed. The NACK scheme exhibits a similar asymmetric packet loss, as it is essentially an improved version of UDP. Notice that the NACK scheme’s totals in Figure 16 are the losses after retransmissions.

Figure 16 Overall packet losses according to streaming direction for the various transport methods



For DCCP and multi-connection TFRC, downlink streaming, the majority of packet losses occur across the wireless link, as these protocols are able to respond to the congestion across the core network to some extent but cannot prevent wireless channel losses. However, the number of packets available to be dropped at the wireless stage is reduced because of earlier losses from congestion. Figure 17 shows the breakdown.

Figure 17 Proportion of wired/wireless network packet losses for downlink streaming with the various transport methods



From Table 3, the percentages of packet losses for UDP transport for downlink streaming are much higher than the other methods. Though DCCP and multi-connection TFRC are able to reduce the packet loss levels, in this IPTV distribution network, the levels are too high, as they are around 10%. This implies that only the introduction of application-layer FEC or some form of error resilience could improve the situation. The net result of these packet losses, Table 3, is that UDP transport results in poor video quality. Only uplink streaming video quality passes above 25 dB when quality is ‘fair’ (according to an approximate mapping (Klaue et al., 2003) between the ITU’s mean opinion score rankings and PSNR).

Table 3. Mean performance metrics when streaming *Paris* over an IPTV delivery network for the various transport methods

	UDP	*DCCP*	*Multi-Conn*	*BVS*
	Packets lost			
DL	23.4	9.37	11.18	5.49
UL	11.6	10.46	11.67	2.87
	PSNR (dB)			
DL	18.01	24.55	24.18	27.62
UL	24.81	25.46	25.02	31.18
	End-to-end delay (s)			
DL	0.029	0.018	0.029	0.042
UL	0.049	0.016	0.020	0.062
	Sending period (s)			
DL	35.63	139.18	91.18	36.32
UL	35.62	134.00	69.81	35.77
	Jitter (s)			
DL	0.0097	0.0349	0.0097	0.0079
UL	0.0084	0.0314	0.0071	0.0076
	Throughput (kbps)			
DL	627	189	271	773
UL	751	197	360	809

However, NACK-based uplink streaming results in ‘good’ quality video (just). The mean end-to-end delay of DCCP and multi-connection

TFRC is lower again than UDP and the NACK-based scheme. This is because both DCCP and multi-connection TFRC reduce their sending rate, resulting in less queuing time. From Table 3, UDP and NACK-based scheme's sending period is approximately the same and close to the duration of the *Paris* sequence. However for DCCP, packet losses on the wireless link again cause excessive delay, as DCCP introduces large inter-packet gaps. Multi-connection TFRC is able to increase wireless utilization but this can be at a cost of greater packet losses across the connections. The NACK scheme still almost matches the sending period of the video sequence, by virtue of reduced end-to-end delay, despite sending more packets through retransmissions than UDP. The levels of inter-arrival-time packet jitter confirm that DCCP decreases congestion by increasing the inter-packet gap to too high a duration. Multi-connection TFRC can reduce the jitter but not sufficiently compared to UDP and the NACK-based scheme. Similarly, multi-connection TFRC with four connections increases throughput but greater net throughput is achievable with the NACK-based scheme.

5 Conclusions and implications

The paper has presented results that support the use of the Internet TV variant of IPTV when there is intelligent content placement and a limited path from the regional content cache across a Metro network. The results show that across a range of performance metrics the proposed simple NACK-based scheme is, in fact, better, than raw UDP transport, industry-standard DCCP (TFRC with connection set-up), and multi-connection TFRC. As the latter two have either been used in a commercial product (Google VideoTalk) or have been experimented at the highest level within the academic literature (refer to Sections 1 and 2), these results represent a counter-balance or alternative insight to perceived ways of providing video streaming.

It is likely that in the new phase of wireless network growth, video traffic will begin to dominate traffic flows, compared to the previous dominance of first voice traffic and then data traffic. Video services, such as the Internet TV variant of IPTV considered in this paper, may well transfer to mobile systems, which explains the development of WiMAX IEEE 802.16m and LTE networks. The fact that the video server is likely to be closer at hand through intelligent content management implies that NACK requests for retransmission will not introduce too much latency to an end-to-end solution. The proposed scheme achieves effective wireless channel utilization without the level of packet losses (after retransmissions) that can arise from raw UDP transport and that can badly affect fragile compressed video streams. For IPTV with intelligent placement of content close to the access network then TFRC/DCCP shares the problems that TCP is known to suffer from, such as long pauses in transmission. Poor wireless channel utilization is partially solved by multiple-connection TFRC so long as channel conditions are not poor. However, video quality is still reduced in comparison to the proposed NACK-based scheme and buffer management is required at the mobile station. Therefore, the proposed NACK-based scheme seems to be a simpler way forward than multi-connection TFRC.

When TFRC or its enhanced DCCP version is adopted for wireless networks it is because: a) it is an industry-standard form of congestion control and b) TFRC is a survivor of the academic debate that produced many solutions to the need to avoid congestion collapse in the wired Internet by ensuring that congestion control was TCP friendly. It is understandable that commercial developers wish to avoid a further set of comparative studies and, at the same time, wish to select a congestion control mechanism that can work across a number of products. However, this study has raised doubts about its use as an end-to-end congestion controller for heterogeneous networks consisting of wired and wireless path elements, subject to both congestion-induced packet drops and packet loss across the wireless element. Even the multi-connection form of TFRC, which was designed to address some of these doubts, may not be the most practical solution. Given that the alternative scheme presented in this paper is relatively simple, it is probably time for another look at what constitutes a suitable application-layer controller for Internet TV.

6 Limitations and future research

This paper has charted the implications of what is in effect an enhanced form of UDP. However, there is certainly scope for more extensive tests in terms of differing video content, differing network test scenarios, and differing WiMAX configurations, to name but a few degrees of testing freedom. Mobility implies that either vertical or horizontal handoffs will eventually occur and the response during such events was not considered. The proposed scheme has been considered in its most simple form in which all packets of whatever content type are retransmitted. However, a packetized video stream does not consist of packets that are all of equal importance to the reconstruction at the decoder. There are a number of ways to prioritize single-layer video packets, such as according to the picture type (I-, B-, or P-picture) or according to data-partition type (A-, B-, C-partition in an H.264/AVC codec). Apart from selective NACK-based retransmission of single-layer video, scalable video naturally introduces priority layers that could be exploited by the scheme. The advantage of such selective NACK-based schemes is that they reduce overall latency, compared to the situation when all lost packets are NACKed. It is also recommended that comparative evaluation also takes place on a heterogeneous network testbed, such as the one being constructed on the University campus of the authors. Though these testbeds are unable to model a full Internet TV system, they raise the level of confidence that commercial developers have in the robust working of a system.

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