Journal of Mobile Multimedia, Vol. 7, No.3 (2011) 177-193

ADAPTIVE BROADBAND VIDEO STREAMING FOR IPTV WIRELESS ACCESS

SALAH S. MAJEED MARTIN FLEURY

University of Essex, Colchester, UK

ssaleha@essex.ac.uk *fleury.martin55@gmail.com*

Received March 1, 2011
Revised August 29, 2011

Broadband wireless access supports mobile applications, which may soon extend to Internet Protocol TV (IPTV). IPTV streaming from a localized server is likely to be across a wired path through a metro network before crossing the wireless link. In this scenario, the paper proposes Broadband Video Streaming (BVS), which enhances UDP transport with a single, negatively acknowledged, lost packet retransmission. Taking IEEE 802.16e access as an example, results demonstrate that BVS is sufficiently able to compensate for packet losses without overly increasing delay and without the overhead of application forward error correction, whereas unembellished UDP, and two alternative congestion controllers (single- and multi-connection versions of TCP-Friendly Rate Control (TFRC)) are unable to both reduce packet loss and streaming delay. The paper exposes asymmetrical streaming behavior between downlink and uplink streaming and finds that, for downlink streaming, packet reordering by video picture-type packet is sensible. The paper then extends the classic BVS scheme with an adaptive scheme that takes into account whether packet losses are from congestion or wireless channel conditions or a mixture of both. To cope with this, adaptive BVS adopts differentiated lost packet retransmission according to the picture type of a lost packet. It is found that for greater packet loss, adaptive BVS achieves equivalent objective video quality to BVS but with reduced delay and bandwidth consumption.

Key words: broadband video streaming, IPTV, mobile WiMAX, multi-connections TFRC
Communicated by: D. Taniar & E. Pardede

1 Introduction

Broadband wireless access in the form of IEEE 802.16d,e (WiMAX) [13] continues to be rolled out in many parts of the world that do not benefit from existing wired infrastructures or cellular networks. In rural and suburban areas in some developed countries [6], WiMAX is also cost effective. The transition to the higher data rates of IEEE 802.16m (see [2] written by Intel's chief technology officer) indicates the competiveness of WiMAX. One of the drivers of WiMAX's development is its suitability for Internet Protocol TV (IPTV), including mobile IPTV [19]. The success of IPTV services such as the BBC's iPlayer in the UK indicates the likely demand. The iPlayer allows TV programs to be streamed on demand, either 'start over' live programs or time-shifted TV. However, this service appears to be primarily aimed at Asymmetric Digital Subscriber Line (ADSL) receivers and may be ill-adapted to mobile wireless broadband delivery. Though currently based on Adobe Flash Player technology, it could be that, because of the limitations of TCP transport (unbounded delays and fluctuating bitrates) required for Flash Player, TCP will be superseded by other transport protocols.

This paper contributes Broadband Video Streaming (BVS), which is a simple, single retransmission scheme aimed at improving IPTV video streaming. Because it employs a single Negative Acknowledgement (NACK), it is most suitable for situations where the roundtrip time is not long. However, it now seems likely [7] that intelligent content management will result in local caching of frequently-requested content reducing the roundtrip time. It is possible to elaborate this simple scheme by assigning priorities to the compressed video data and varying the number of retransmissions accordingly. In a practical scheme, priority decisions can be based on: picture type[a], or display and coding deadlines [22]. The paper explores the potential of priority retransmission based on picture type. It subsequently demonstrates an adaptive scheme, in which retransmission of lost packets carrying selected video picture types depends on packet loss statistics. Depending on the estimated nature of the packet losses, whether through congestion or adverse wireless channel conditions or a mixture or both, different selections of video picture types are retransmitted if their packets are lost. Estimation is based on the trend of a relative measure of one-way delay.

The proposed scheme is a modification of UDP transport which is applied at the application layer of the protocol stack. Though UDP streaming has been trialed for broadband wireless access [14], UDP packet losses can seriously harm a compressed video stream. This is due to the predictive nature of video coding which operates through motion compensation and entropy coding. Bell Labs introduced a reliable form of UDP, R-UDP, see [21], and there is also an R-UDP protocol employed by Microsoft in their *MediaRoom* product for IPTV service delivery over multicast networks. In the present scheme, UDP is supplemented with negative acknowledgments (NACKs) whenever a packet is lost for the first time. To avoid additional latency, the receiver only requests retransmission once. In the paper, this non-selective scheme is adaptively combined with selective NACKs, which are reserved for lost anchor picture packets from the video stream.

By comparison with BVS, we consider the performance of several alternative end-to-end video transport methods, and test them over a WiMAX channel. Thus, the paper compares BVS with directly-applied UDP-based streaming, the industry-standard Datagram Congestion Control Protocol (DCCP) [18], and multi-connection streaming. DCCP adds connection handling to TCP-Friendly Rate Control (TFRC) [12]. For example in [23], DCCP was used over a Long Term Evolution (LTE) [8] but to do so required packet losses to be disguised from the application by repeated retransmissions at the data-link layer. TFRC itself modifies TCP's congestion control mechanism for UDP transported video streams, smoothing the bitrate but retaining the same average bitrate over time of TCP. To further increase wireless channel utilization, multi-connection TFRC [5] can be considered. In single-connection TFRC, wireless channel packet loss is misinterpreted as congestion, causing the congestion controller to reduce its sending rate, resulting in poor utilization and lengthened streaming periods. To avoid this, a stream can be split between several connections, allowing the aggregate rate to compensate for the slowdown of individual connections. This option was developed for tandem networks in which there is a wired and wireless component and our paper also considers a tandem network.

---

[a] The terms picture and frame are used interchangeably in this paper as progressive rather than interlaced TV is assumed.

The remainder of this paper is organized as follows. Section 2 initially describes alternative methods of IPTV transport before turning to the proposed BVS scheme. The classic and extended versions of the BVS scheme are described in this Section. The simulation mode is described in Section 3, giving the WiMAX settings, the channel model for 'bursty' errors, and the video configuration. Section 4 is an evaluation of both classic BVS and the extended adaptive BVS. Finally, Section 5 draws some conclusions.

2 Video streaming transport

This Section initially considers the IPTV network architecture that is assumed in the paper. Section 2.1 examines some video transport alternatives that appear in the literature, before Sections 2.2 and 2.3 introduce classic BVS and extended BVS respectively.

2.1 IPTV architecture

In [14], IP/UDP/Real Time Protocol (RTP) IPTV streaming was evaluated on a WiMAX testbed for downlink delivery of TV channels and uplink delivery of either TV news reports or video surveillance; refer to Fig 1a. However, that research [14] did not consider the impact of the intervening core wired network connecting the WiMAX base stations. In [7], ways to improve IPTV quality were discussed with the assumption that intelligent content management would bring popular video content nearer to the end viewer. The typical IPTV architecture considered in [7], Figure 1b, 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.

After connection negotiation has taken place, DCCP adopts TFRC for streaming purposes. However, in the wireless domain, several attempts, e.g. [5] [9], have taken place to improve TFRC's utilization of the wireless channel, which reduces sharply when packet loss occurs. The result can be that considerable interruption to the stream may occur if packet losses, causing TFRC to reduce its streaming rate without reducing video quality. In [9] [10], cross-layer intervention occurs in one way or another to mask channel packet loss from TFRC. Alternatively in [23], it is assumed that the data-link layer transparently retransmits packets until successful receipt occurs. The potential problem of this approach [23] is that the application loses control of packet latency, which implies that the mobile device will require large buffers to compensate, with resulting lengthy start-up times.

In this paper, the various streaming protocols are tested through simulation across the path over the metro network to the user subscriber station (SS). Figure 2 shows the tandem network simulated in which node C represents the source or sink of downlink or uplink streaming according to Figure 1. The WiMAX channel is between the base-station (BS) and SS shown. In the Figure, all links except a bottleneck link within the metro network are set to 100 Mbps to easily accommodate the traffic flows entering and leaving the network. The link delays are minimal (2 ms) to avoid confusing propagation delay with re-ordering delay in the results. A bottleneck link with capacity set to 5 Mbps is set up between the two routers. This arrangement is not meant to physically correspond to a network layout but to represent the type of bottleneck that commonly lies at the 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 FTP flow sourced at node B. Node B also sources an FTP flow to the BS and a CBR stream at 1.5 Mbps with packet size 1 kB.

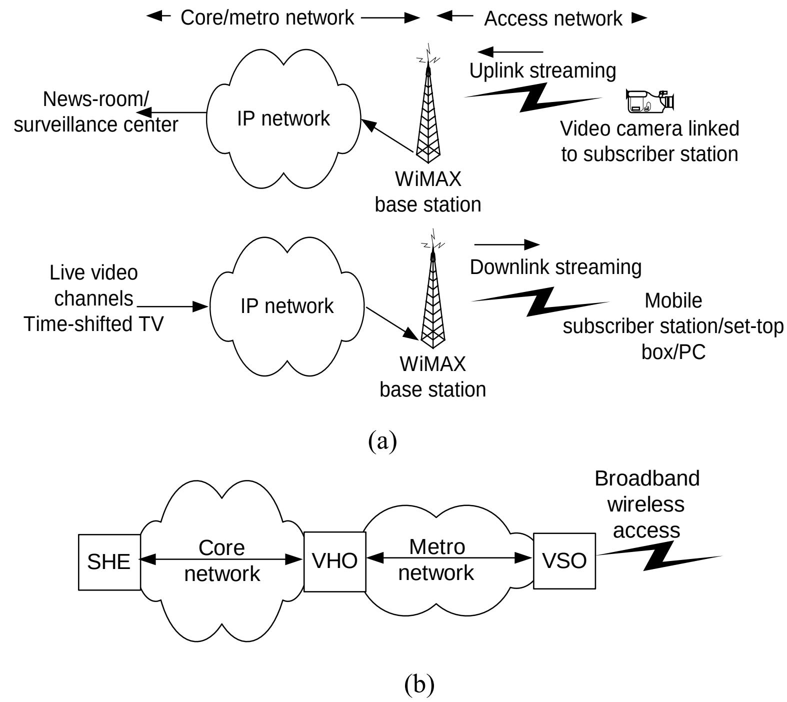

Figure 1 (a) Downlink and uplink streaming scenarios, (b) schematic IPTV distribution network.

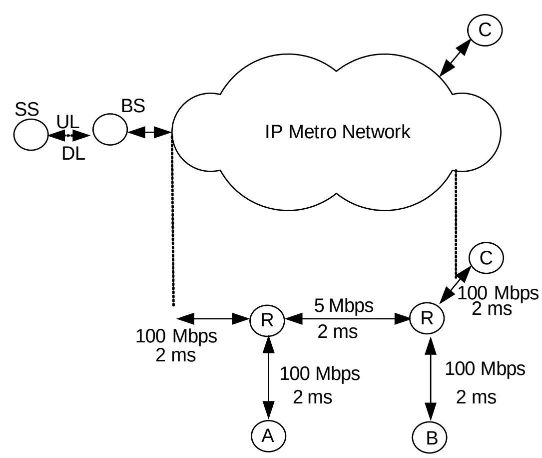

Figure 2 Video streaming test topology for IPTV.

Another approach, as mentioned in Section 1, is to employ multi-connection TFRC [5]. In the experiments reported in Section 4, the number of connections was set to four, as this value has been found to be [3] the point when the value of the multi-connection approach is maximized. It is possible that other numbers of connections could improve the performance of multi-connection TFRC. However, finding the correct number of connections then becomes a further problem.

2.2 BVS approach

In contrast, after observing that UDP at least succeeds in good wireless channel utilization [14], without any protection from channel loss, the simple BVS scheme introduces a single negative acknowledgment (NACK) to UDP. At the receiver, a record is kept of packet sequence numbers available through the RTP header and if an out-of-sequence packet arrives a NACK is transmitted to the sender. The video source prevents transmission from its input buffer until a single retransmission of the missing packet in the sequence has taken place. Further retransmissions do not take place, because waiting packets could be delayed and because the failure of one retransmission may indicate continuing poor channel conditions at the broadband wireless link. Thus the sender only transmits a NACK if this is the first time that particular packet has been lost. If it is the first time and the non-selective NACK version of BVS is in operation then a NACK is sent. However, if prioritized operation is in use then a decision is made according to the picture type of the video packet that has been lost, reflecting the importance to the reconstruction of the video of that packet.

Figure 3 is a general representation of the processing involved in the scheme, showing the NACK response of the receiver. The following describes the operation assuming downlink streaming from a BS to an SS, with the protocol stacks for both video source and mobile receiver shown. At the video source, compression is through an H.264/AVC (Advanced Video Coding) codec [26]. H.264/AVC conceptually separates source coding in the Video Coding Layer (VCL) from network adaptation in the Network Abstraction Layer (NAL). Figure 2 shows a distinction between intra-coded Instantaneous Decoding Refresh (IDR) picture NALs which for consistency with earlier codecs we call I-picture NALs. Predictively-coded P-pictures are motion compensated and compared with a previous reference picture within a Group of Pictures (GoP) bounded by an I-picture. Bi-predictive B-pictures are predicted from two reference pictures, which in traditional usage are not other B-pictures. Thus there is no effect on predictive coding from the loss of a B-picture carrying packets. In H.264/AVC, a NAL unit forms a virtual packet which can be encapsulated by an RTP header. At the receiver, if the packet is not lost, it enters a buffer before reordering prior to decoder playout. If a packet is lost then there is a check whether all lost packets are to be NACKed. If not the priority of the packet, is checked and a selective NACK takes place.

2.3 Adaptive extension to BVS

BVS can be extended to adaptively react to congestion and channel conditions. We have used BVS according to the Spike scheme, first reported in [24] and then employed in [4]. However, application of BVS to the Spike scheme is the contribution of this paper's authors. In the Spike scheme, a peak or spike in the Relative One-way Trip Time (ROTT) indicates the presence of congestion. Because of the problem of clock synchronization, one-way trip time can only be relative to the clock rate at the sender and receiver and is not an absolute measure of end-to-end-delay. When the ROTT passes above a given threshold, packet loss is definitely from congestion. When it passes below a threshold, it is assumed to be definitely from wireless channel conditions. In Figure 4, classic BVS operates in the bad channel condition zone which can exist at various points in time. In this situation, packets from all picture types are re-transmitted when necessary, in order to adequately reconstruct the video sequence. However, if there is limited congestion and moderate problems within the wireless channel then only I- and P-picture packets need be re-transmitted in order to reduce delay arising from retransmissions. If congestion increases then within the high congestion zone, only I-picture packets are re-transmitted to avoid further adding to the congestion. B-picture packets can be neglected as they have no effect on predictive decoding. Also shown in Figure 4 are two shaded areas where there is a margin of measurement error as to which zone is occupied.

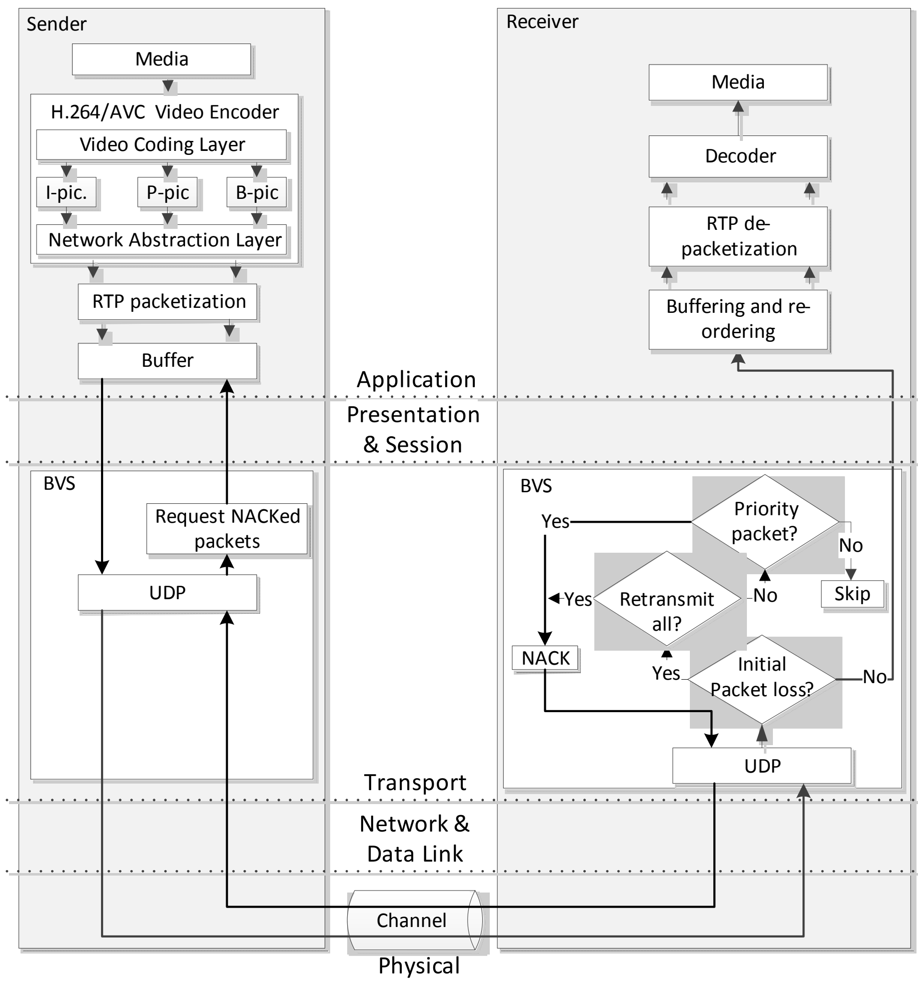

Figure 3 Operation of BVS NACK enhancement to UDP, including adaptation.

To judge the zone that is occupied the ROTT is calculated. Two thresholds, α and β, mark the extent of the zones in the adaptation of the Spike scheme for this paper. To find α the receiver continually monitors ROTT as recorded by the time stamped RTP packets to find the minimum ROTT. Clearly over time this estimate will be refined. The threshold is modified by a small percentage of the ROTT, which again is modified by a truncated normalized Gaussian random variable, P = G (-0.5, 0.5). The point of including this modification is to model the impact of measurement noise in the delay estimate, as in the simulations of Section 3 measurement error would not otherwise occur.

$$\alpha = \text{Min. Delay Threshold} \pm (\text{ROTT} \times 5\% \times P) \tag{1}$$

In a similar manner, β is found as in (2). The maximum end-to-end delay threshold was set to 50 ms in tests, as this is the maximum recommended value for IPTV recommended by the ITU-T.

$$\beta = \text{Max.-End-to-End-Delay-Threshold} \pm (\text{ROTT} \times 5\% \times P) \tag{2}$$

Notice that adopting a constant value for the maximum end-to-end delay threshold avoids the problem of defining the separation between the upper and lower thresholds [4] that otherwise arises.

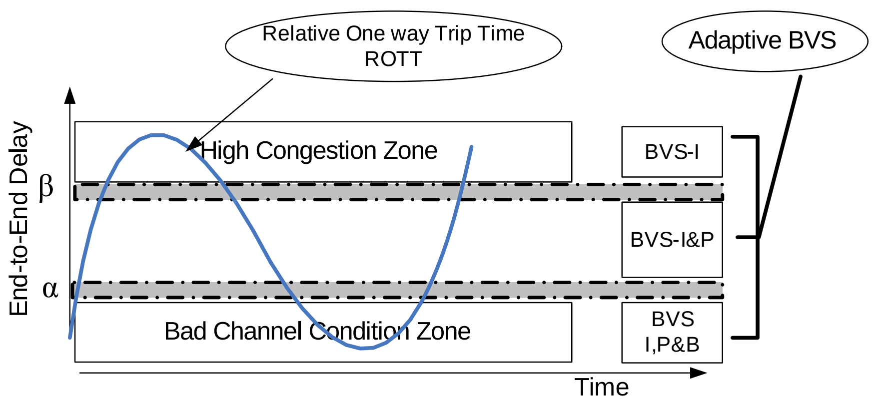

Figure 4 Adaptive BVS NACK enhancement to UDP.

3 Simulation model

The WiMAX system operating in point-to-multipoint mode was simulated by well-known ns-2 simulator (v. 2.29) augmented by a WiMAX module [25] from which values not mentioned below were defaulted. Mean data points 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 20s before commencing video streaming. A trace file was input to ns-2 and packet losses recorded in the output. The output serves to calculate the objective video quality (PSNR). Video quality comparisons were made under the EvalVid environment [17].

3.1 WiMAX configuration

The PHY settings selected for WiMAX simulation are given in Table 1. The antenna heights and transmit power levels are typical ones taken from the Standard [13]. The antenna is modeled for comparison purposes as a half-wavelength dipole, whereas a sectored set of antenna on a mast might be used in practice to achieve directivity and, hence, better performance. Similarly, multiple-input multiple-output (MIMO) antennas are not modeled. The IEEE 802.16e Time Division Duplex (TDD) frame length was set to 20 ms (one of the higher values available in the Standard) to suit the real-time nature of video streaming. Shorter frame lengths can lead to waiting in WiMAX service queues. The data rate results from the use of one of the mandatory coding modes for a TDD downlink/uplink sub-frame ratio of 3:1. The BS was assigned more bandwidth capacity than the uplink to allow the WiMAX BS to respond if necessary to multiple mobile devices. Thus, the parameter settings in Table 1 such as the modulation type and PHY coding rate are required to achieve a datarate of 10.67 Mbps over the downlink. The buffer sizes at the BS and MS were set to fifty packets, as it is unlikely that mobile stations will support large buffers as these cause delay and contribute to energy consumption. In a WiMAX setting, a packet corresponds to a MAC Service Data Unit (MSDU) within a MAC Protocol Data Unit (MPDU).

3.2 Channel modeling

A Gilbert-Elliott two-state, discrete-time, ergodic Markov chain [27] modeled the wireless channel error characteristics at the ns-2 physical layer. The probability of remaining in the 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. These values were *not* chosen because they represented the underlying physical characteristics of a particular channel but because they represent error statistics [16] seen by an application. Burst errors can be particularly damaging to compressed video streams, because of the predictive nature of source coding. Therefore, the impact of 'bursty' errors [20] should be assessed in video streaming applications. In this case, we were interested in settings that implied significant levels of packet loss.

3.3 Video configuration

As a test, we used the *Paris* sequence H.264/AVC Variable Bit-Rate (VBR)-encoded at a display rate of 30 frame/s with Common Intermediate Format (CIF) (352×288 pixel/frame) with quantization parameter (QP) set to 26 (from a range 0 to 51). The common 4:2:0 chroma sub-sampling configuration was used. The video quality (PSNR) for this sequence without packet loss is 38 dB. The slice size was fixed at the encoder at 900 B. In this way the risk of network segmentation of the packet was avoided, which could result in loss of synchronization at the decoder.

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
Channel model	Gilbert-Elliott
MS transmit power	250 mW
BS transmit power	20 W
Approx. range to MS	0.7 km
Antenna type	Omni-directional
Antenna gains	0 dBD
MS antenna height	1.5 m
BS antenna height	32 m

OFDMA = Orthogonal Frequency Division Multiple Access, QAM = Quadrature Amplitude Modulation, TDD = Time Division Duplex

Paris consists of two figures seated round a table in a TV studio setting, with significant spatial-coding complexity and moderate motion. Quality-of-Experience tests show [1] that this type of content is favored by users of mobile devices as it does not stretch the capabilities of the screen display (as, for instance, sport sequences would do). Moreover the clip also approximates to mobile video conferencing video streams. The Intra-refresh rate was every 15 frames with an IPBB…I coding structure. 1065 frames were transmitted resulting in a video duration of 35.5 s. Many commonly available test sequences are less than 10 s in length, which, though suitable for testing source coding innovations, are not necessarily suitable for testing network behavior. Simple previous frame replacement was set for error concealment at the decoder as a point of comparison with others' work.

4 Evaluation

This Section initially examines classic BVS streaming before turning to the adaptive extension proposed in Section 2.3.

4.1 BVS without adaptation

Evaluation first considered downlink and uplink streaming in the scenario of Figure 1. The test topology was as in Figure 2. Strictly, uplink modeling is not required but tests revealed some interesting contrasts.

In Figure 5, 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 further losses occur across the WiMAX link. For uplink streaming packet losses from congestion are reduced. 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 and consequently self-congestion in the intervening router buffers is reduced. BVS exhibits a similar asymmetric packet loss pattern between downlink and uplink, as it is essentially an improved version of UDP. Notice that the BVS totals in Figure 5 are the losses after retransmissions and do not directly show packet losses across the transmission paths. 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 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 6 shows the breakdown. The number of packets dropped is greater in uplink streaming for these two protocols, as all packets are dropped over the wireless link, which is encountered first.

From Table 2, the percentages of packet losses for UDP transport from 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 forward error correction or some form of error resilience could improve the situation. The net result of these packet losses, Table 2, 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 between the ITU P.1010's recommended Mean

Opinion Score (MOS) rankings and PSNR (see [11] for a conversion table). However, BVS uplink streaming results in 'good' quality video using an objective-MOS scale.

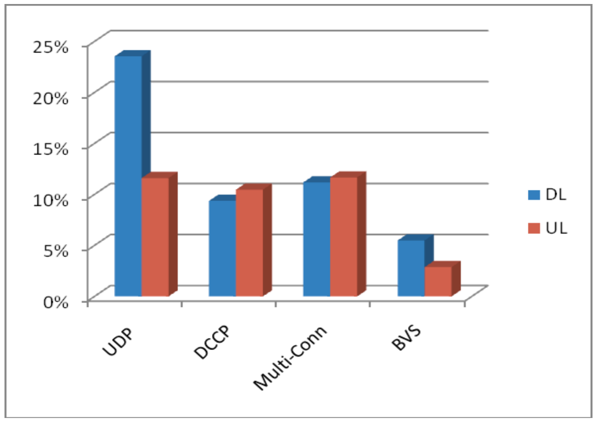

Figure 5 Percentage overall packet loss according to streaming direction.

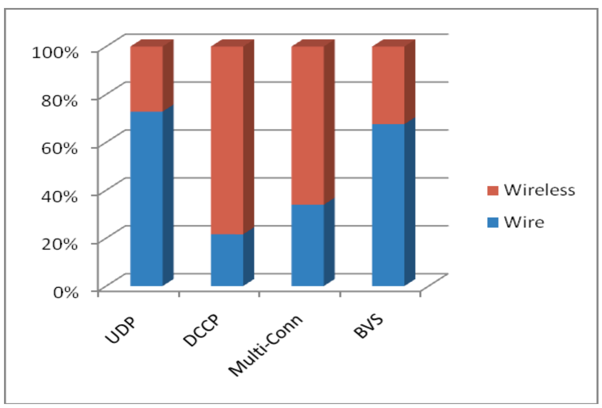

Figure 6 Proportion of wired/wireless network packet losses for downlink streaming.

The mean end-to-end delay of DCCP and multi-connection TFRC is lower again than UDP and BVS. This is because both DCCP and multi-connection TFRC reduce their sending rate, resulting in less queuing time. From Table 2, UDP and BVS'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. BVS 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 enough compared to UDP and BVS. Similarly, multi-connection TFRC with four connections increases throughput but greater net throughput is achievable with BVS.

Table 2 Mean performance metrics when streaming *Paris* over an IPTV delivery network.

	UDP	DCCP	Multi-Conn	BVS
	Packet loss rate (%)			
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

An interesting feature of our analysis, Figure 7, was that in downlink streaming proportionally more of the larger intra-coded I-picture packets are lost in UDP and BVS streaming. Notice from Figure 6 that for UDP and BVS, essentially without congestion control, more losses occur in the wired portion of the IPTV delivery network than occur in the wireless part. Larger I-picture packets on average contribute to 26 packets after segmentation at the encoder, all arriving together at a router buffer. In contrast, predictively-coded P-pictures on average are broken into three packets at the encoder. The total number of P-pictures in a 15-frame GOP is 12, but it is the arrival pattern that is significant. A bi-predictively-coded B-picture contributes two packets in the mean, leading to bursts of four packets (and 20 packets per GOP).

The pattern of wireless packet channel losses is not so selective of I-picture packets, as the breakdown in Figure 8 by picture type for uplink streaming illustrates. Notice again that in Figure 8, the BVS packet losses include retransmissions and, therefore, do not directly reflect the packet loss pattern. From Figures 7 and 8, it is apparent that video quality for BVS streaming can be further improved by avoiding bursts of I-picture packets. In fact, if packet loss levels increased compared to the comparatively low levels of Table 2, then this pattern of packet losses would be a problem for BVS downlink streaming. At a cost in delay, this can be achieved by packet reordering between the picture type packets, as occurred in [15] as a form of video smoothing. However, in the analysis of adaptive BVS that follows this option is not taken up to make a comparison with BVS possible.

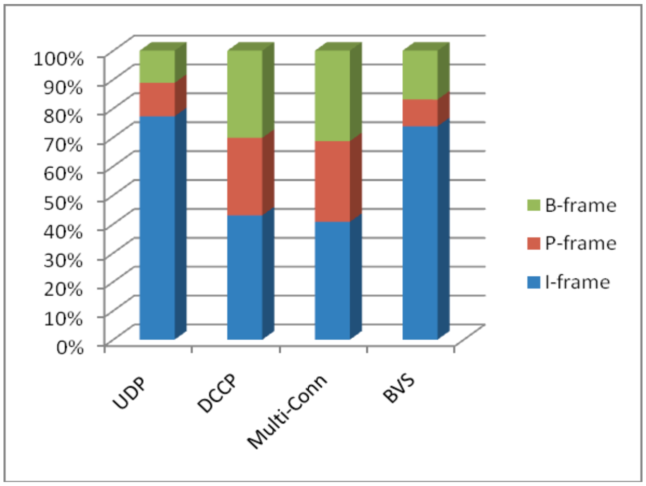

Figure 7 Breakdown by frame type of packet losses when downlink streaming.

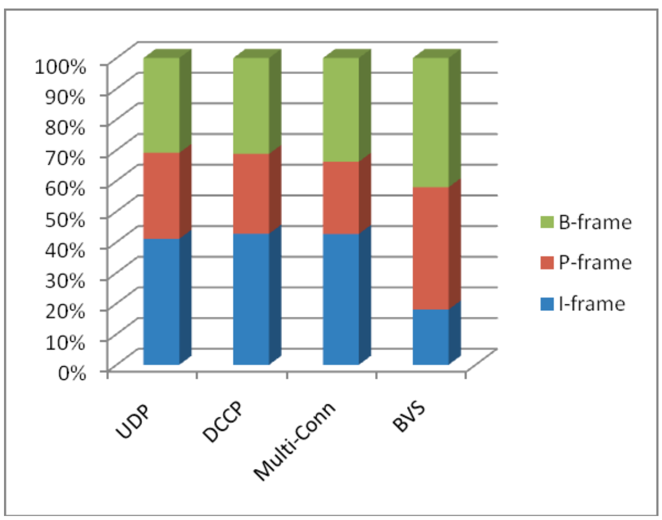

Figure 8 Breakdown by frame type of packet losses when uplink streaming

4.2 BVS with adaptation

This Section is confined to downlink streaming. In Figure 9, the packet loss performance of UDP and TFRC is contrasted with that of several varieties of BVS without adaptation. It can be seen that there is not much to separate UDP and TFRC except at the higher error rate (according to the Gilbert-Elliott model), when TFRC is able to reduce its sending rate sufficiently. 'BVS' in Figure 9 is classic BVS in which all lost packets are retransmitted just once. BVS-I in the Figure is when only I-picture packets are retransmitted and BVS-A represents BVS with adaptation. Thus, the rankings of aggregate packet loss at the receiver are not surprising given the retransmission policies. Figure 10 is a breakdown of packet loss by picture type. Examination of the relative loss of I-picture packets lost indicates that relatively more of these packets are lost in BVS than for the other BVS schemes. Consequently, despite the higher total packet losses for BVS-A, the video quality is similar to BVS in Figure 11. Thus, BVS-A can achieve similar video quality with less retransmissions. BVS-I is better than the two other schemes examined but its video quality is reduced.

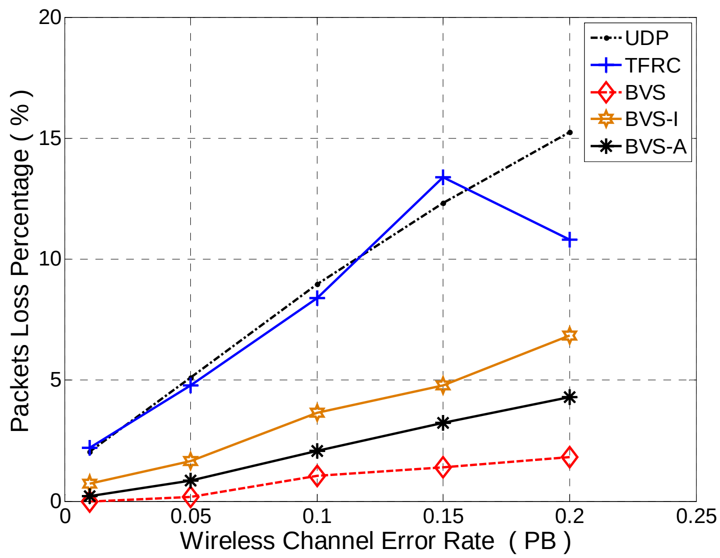

Figure 9 Packet loss according to wireless channel condition.

Table 3, amends the PSNR findings according to the impact of measurement noise. BVS-A represents accurate estimation of the ROTT in equations (1) and (2). BVS-A- represents a subtraction of the adjustment in (1) and (2), while BVS-A+ represents addition of the adjustment. The difference in PSNR is only once just over 1 dB. Turning to delay in Figure 12, it can be seen that BVS-A's packets are delayed less that in the classic BVS scheme while achieving equivalent video quality. BVS-I remains as a lower latency version of BVS. In all cases, end-to-end delay meets IPTV's stringent delay requirements in the scenario examined.

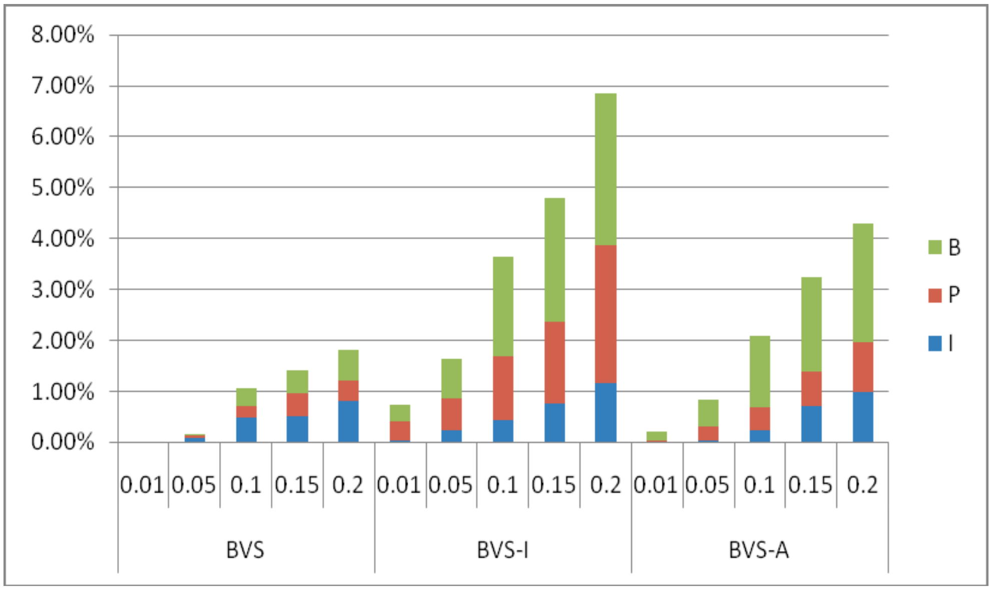

Figure 10. Packet loss according to wireless channel condition and picture type.

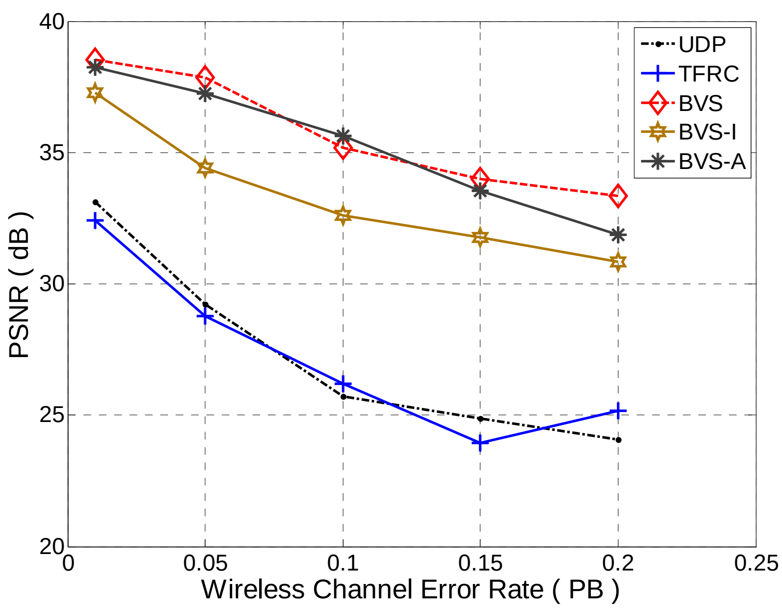

Figure 11 Video quality according to transport scheme and channel condition.

Table 3 Video quality in dB according to the effect of measurement noise.

PB:	*0.01*	*0.05*	*0.10*	*0.20*	*0.25*
BVS-A-	37.81	37.00	35.10	32.91	30.82
BVS-A	38.23	37.25	35.64	33.55	31.87
BVS-A+	38.10	37.35	35.70	33.50	31.95

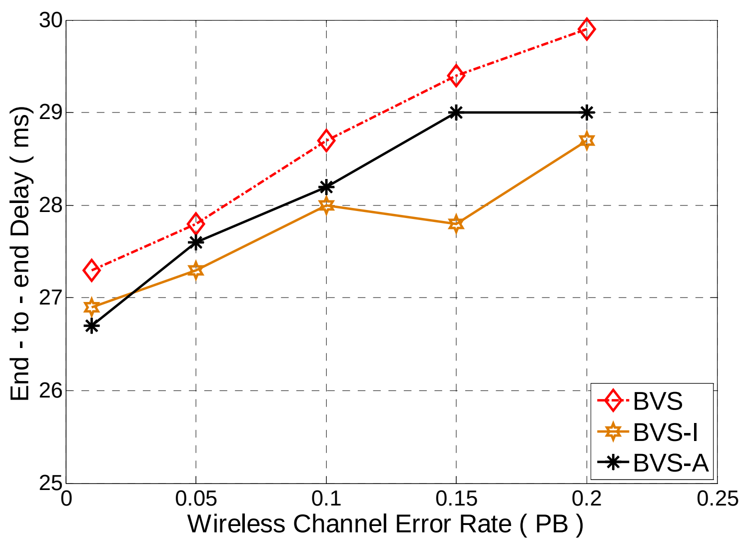

Figure 12 End-to-end delay according to BVS scheme and channel condition.

5 Conclusion

For IPTV, with intelligent placement of content closer to the access network, we have demonstrated BVS which is a simple video transport scheme based on negative acknowledgments. Furthermore, with BVS with adaptation and selective acknowledgments, good video quality results both during periods of high congestion and bad channel conditions. In addition the adaptive version of BVS reduces packet end-to-end delay and improves wireless channel utilization. In comparison to BVS, TFRC/DCCP shares to a lesser degree the problems that TCP is known to suffer from when traversing a tandem network. Poor wireless channel utilization is partially solved by multiple-connection TFRC/DCCP in better channel conditions but video quality is reduced in comparison to BVS and buffer management is required at the mobile station. Adaptive BVS is flexible and while in this paper we have opted for a practical scheme based on selective retransmission of lost packets according to picture type priority, other prioritizations are possible such as based on video layering or coding priority.

References

1. Agboma, F., and Liotta, A., Addressing user expectations in mobile content delivery. Mobile Information Systems, 3(3/4), 153–164, 2007.
2. Ahmadi, S. Mobile WiMAX: A systems approach to understanding IEEE 802.16m radio access technology. Academic Press, London, UK, 2010.
3. Al-Majeed, S. S., and Fleury, M. Options for WiMAX uplink media streaming. Int. Journal of Mobile Computing and Multimedia Communications, 2(2), 49-66, 2010.
4. Cen, S., Cosman, P.C., and Voelker, G.M. End-to-end differentiation of congestion and wireless losses. IEEE/ACM Transactions on Networking, 11(5), 703-715, 2003.
5. Chen, M., and Zakhor, A. Multiple TFRC connections based rate control for wireless networks. IEEE Transactions on Multimedia, 8(5), 1045-1061, 2006.
6. Cicconnetti, C., Lenzini, L., Mingozzi, E., and Eklund, C., Quality of service support in IEEE 802.16 networks. IEEE Network, 20(2), 50-55, 2006.
7. Degrande, N., Laevens, K., and De Vleeschauwer, D., Increasing the user perceived quality for IPTV services. IEEE Communications Magazine, 46(2), 94-100, 2008.
8. Ekstrom, K. et al. Technical solutions for the 3G Long-Term Evolution. IEEE Communs. Mag., 44,(3), 38-45, 2006..
9. Fu, Y., Hu, R., Tian, G., and Wang, T. TCP-Friendly Rate Control for streaming service over 3G Network. In Proceedings of the Int. Conf. on Wireless Comms., Networking and Mobile Computing, (2006).
10. Görkemli, B., Sunay, M.O., and Tekalp, A.M., Video streaming over wireless DCCP. In Proceedings of the IEEE Int. Conf. on Image Processing, (San Diego, CA, 2008) 2028–2031.
11. Gross, J., Klaue, J., Karl, H., and Wolisz, A., Cross-layer optimization of OFDM transmission systems for MPEG-4 video streaming. Computer Communications, 27(4), 1044-1055, 2004.
12. Handley, M., Pahdye, J.,Floyd, S., and Widmer, J., TCP-Friendly Rate Control (TFRC): Protocol specification. IETF, RFC 3448, 2003.
13. IEEE, 802.16e-2005.. IEEE Standard for Local and Metropolitan Area Networks. Part 16: Air Interface for Fixed and Mobile Broadband Wireless Access Systems, 2005.
14. Issa, O., Li, W. , and Liu, H., Performance evaluation of TV over broadband wireless access networks. IEEE Transactions on Broadcasting, 56(2), 201-210, 2010.
15. Jammeh, E., Fleury, M., and Ghanbari, M. Smoothing transcoded MPEG-1 video streams for Internet transmission. IEE Proceedings: Vision, Image & Signal Processing, 151(4), 298-306, 2004.
16. Jiao, Ch., Schwiebert, L., and Xu, B. 2002. On modelling the packet error statistics in bursty channels. In Proceedings of IEEE Conf. on Local Computer Networks, (2002) 534- 541.
17. Klaue, J., Rathke, B., and Wolisz, A. 2003. EvalVid - A framework for video transmission and quality evaluation. In Proceedings of the Int. Conf. on Modeling Techniques and Tools for Computer Performance, (Sept. 2003) 255–272.
18. Kohler, E., Handley, M., and Floyd, S. Datagram Congestion Control Protocol. IETF, RFC 4340, 2006.
19. Lee, J. M., Park, H.-J., Choi, S. G., and Choi, J.K. Adaptive hybrid transmission mechanism for on-demand mobile IPTV over WiMAX. IEEE Transactions on Broadcasting, 55(2), 468-477, 2009..

20. Liang, J., Apostolopoulos, J.G., and Girod, B., Analysis of packet loss for compressed video: Effect of burst losses and correlation between error frames. IEEE Transactions on Circuits and Systems for Video Technology, 18(7), 861-874, 2008.
21. Partridge, C., and Hinden, R. Version 2 of the Reliable Data Protocol. IETF, RFC 1151, 1990.
22. Razavi, R., Fleury, M., and Ghanbari, M., Unequal protection of video streaming through adaptive modulation with a tri-zone buffer over Bluetooth enhanced data rates. EURASIP Journal on Wireless Communications and Networking, [online volume], 16 pages, 2008.
23. Tappayuthpijam, K., Liebl, G., Stockhammer, T., and Steinbach, T. Adaptive video streaming over a mobile network with TCP-Friendly Rate Control. In Proceedings of the Int. Conf. on Wireless Communs. and Mobile Computing, (Leipzig, Germany, 2009) 1325–1329.
24. Tobe, Y., Tamura, Y., Molano, A., Ghosh, S., and Tokuda, H., Achieving moderate fairness for UDP flows by path-status classification. In Proceedings of 25th Annual IEEE Conference on Local Computer Networks, (Tampa, FL, 2000) 252–61.
25. Tsai, F. C.-D. et al., The design and implementation of WiMAX module for ns-2 simulator. In Proceedings of the Workshop on NS2: The IP network simulator, (Pisa, Italy, 2006), article no. 5.
26. Wiegand, T., Sullivan, G.J., Bjontegaard, G., and Luthra, A. Overview of the H.264/AVC video coding standard. IEEE Transactions on Circuits and Systems for Video Technology, 13(7), 560-576, 2003.
27. Zorzi, M., and Rao, R., On the statistics of block errors in bursty channels. IEEE Trans. on Communications, 45(6), 660–667, 1997.