Video Streaming Protocols for Broadband Wireless Access to IPTV

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ABSTRACT
IPTV delivery networks are likely to consist of a wireless access network and a wired path across a metro network. Downlink and uplink video streaming may take place. In this scenario, the paper compares two congestion controllers with UDP and broadband video streaming (BVS), the proposed single negative acknowledged packet retransmission. The results demonstrate that in this environment the congestion controllers (single- and multi-connection versions of TFRC) are unable to both reduce packet loss and overall delay, whereas BVS is sufficiently able to compensate for packet losses without overly increasing delay and without the overhead of application forward error correction. The paper exposes asymmetrical streaming behavior between uplink and downlink streaming and finds that for downlink streaming packet reordering by video frame type is sensible.

Categories and Subject Descriptors
C.2.1 [**Computer-Communication Networks**]: Network Architecture and Design -*Wireless communication.*

General Terms
Algorithms, Performance, Experimentation.

Keywords
Broadband video streaming, DCCP, IEEE 802.16, multi-connection.

1. INTRODUCTION
Despite impending competition with cellular systems such as 3GPP's Long Term Evolution, broadband wireless access in the form of IEEE 802.16d,e (WiMAX) [1] 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, WiMAX is also cost effective. The success of IPTV services such as the BBC's iPlayer in the UK suggests that delivery of video streams to the user will be important. The iPlayer allows TV programs to be streamed on demand, either live programs or time-shifted TV. Though currently based on Adobe Flash Player technology, it could be that because of the limitations of TCP transport (unbounded delays and fluctuating bitrates) this will be superseded by other transport protocols, especially if mobile TV is supported on WiMAX [2].

The contribution of this paper is a single negative acknowledged video streaming scheme for IPTV. It is shown that the scheme is sufficiently able to compensate for packet losses without overly increasing delay and without the overhead of application forward error correction. The proposed Broadband Video Streaming (BVS) 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. In comparison, the paper examines directly-applied UDP-based streaming and the industry-standard Datagram Congestion Control Protocol (DCCP) [3]. For streaming purposes, DCCP adds connection handling to TCP-Friendly Rate Control (TFRC) [4]. TFRC modifies TCP's congestion control mechanism for UDP transported video streams, smoothing the bitrate but retaining the 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 [5] was developed for heterogeneous networks in which there is a wired and wireless component and our paper also considers this scenario.

Figure 1 shows the tandem network simulated in which node C represents the source or sink of downlink or uplink streaming. The WiMAX channel is between the basestation (BS) and subscriber station (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

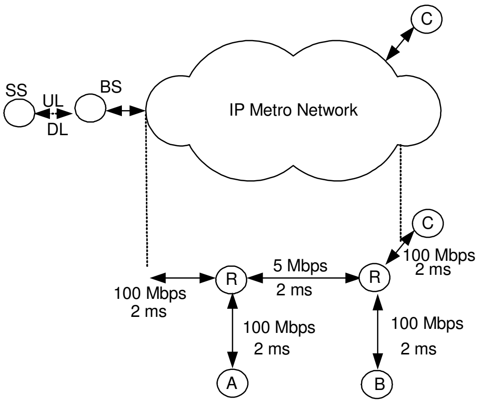

Figure 1. Video streaming scenario for IPTV

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.

2. 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 [6]. 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.

The PHY settings selected for WiMAX simulation are given in Table 1, with additional MAC settings defaulted from [6]. 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 MS by permitting more data to be removed from any queues at each polling time. The value of 20 ms is at the high end of the available durations in the Standard [1] in order to reduce this source of queuing delay. The buffer sizes at the base station and mobile station were set to fifty packets, as it is unlikely that mobile stations will support large buffers. Similarly, router buffers were also set to fifty packets. In a WiMAX setting, a packet corresponds to a MAC Service Data Unit (MSDU) within a MAC Protocol Data Unit (MPDU).

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). As a test, we used the *Paris* sequence H.264/AVC Variable Bit-Rate (VBR)-encoded at 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 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. *Paris* consists of two persons seated round a table in a TV studio setting, with high spatial-coding complexity and moderate motion. Quality-of-Experience tests show [7] that this type of

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 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

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). 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. Simple previous frame replacement was set for error concealment at the decoder as a point of comparison with others' work.

A Gilbert-Elliott two-state, discrete-time, ergodic Markov chain [8] modeled the wireless channel error characteristics at the ns-2 physical layer. A two-state model reproduces conditions experienced during fast fading but does not model slow fades, implying the model is valid for nearby mobile nodes. 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.

3. EVALUATION

In Fig. 2, 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

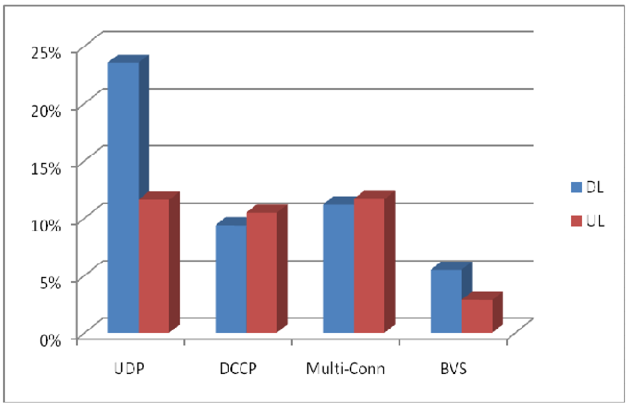

Figure 2. Percentage overall packet loss according to streaming direction.

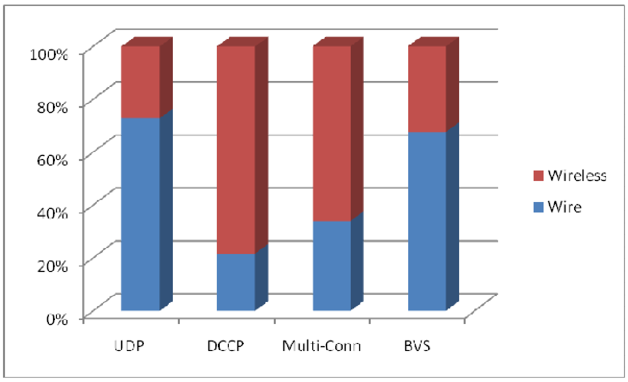

Figure 3. Proportion of wired/wireless network packet losses for downlink streaming.

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 Fig. 2 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. Fig. 3 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 fro 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

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

	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

application-layer forward error control 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's mean opinion score rankings and PSNR). However, BVS 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 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.

An interesting feature of our analysis, Fig. 4, was that in downlink streaming proportionally more of the larger intra-coded I-frame packets are lost in UDP and BVS streaming. Notice from Fig. 3 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-frame packets on average contribute to 26 packets after segmentation at the encoder, all arriving together at a router buffer. In contrast, predictively-coded P-frames on average are broken into three packets at the encoder. The total number of P-frame packets in a 15 frame group-of-pictures (GOP) is 12, but it is the arrival pattern that is significant. Bi-predictively-coded B-frame contribute 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-frame packets, as the breakdown in Fig. 5 by frame type for uplink streaming illustrates. Notice again that in Fig. 5, the BVS packet losses include retransmissions and, therefore, do not directly reflect the packet loss pattern. From Figs. 4 and 5, it is apparent that video quality for BVS streaming can be further improved by avoiding bursts of I-frame 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 frame type packets.

4. CONCLUSION

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. For example, there may be long pauses in transmission when handoff occurs unless intervention occurs such as a fast retransmission scheme. 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.

Therefore, in this paper we have demonstrated BVS which is a simple broadband wireless scheme based on negative acknowledgments. BVS achieves effective wireless channel utilization without the scale of packet losses (after retransmissions) that badly affect fragile compressed video streams. The paper has presented results supporting the feasibility of the BVS approach for IPTV with intelligent content placement and with a limited path over a metro network. When downlink streaming, measures to equalize packet losses across frame types will further improve the effectiveness of a BVS solution. Possible future directions to go are testing on other wireless technologies such as LTE, and comparison with other wireless specific congestion control protocols.

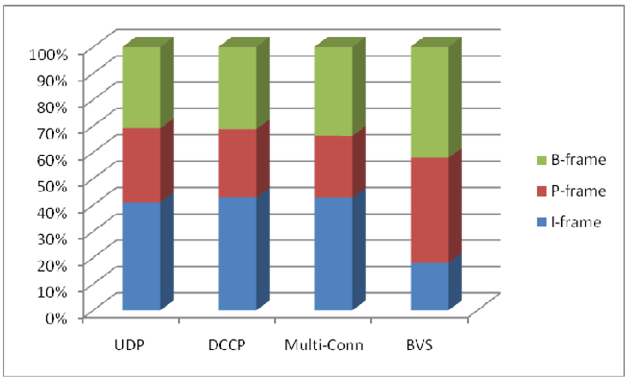

Figure 4. Breakdown by frame type of packet losses when downlink streaming

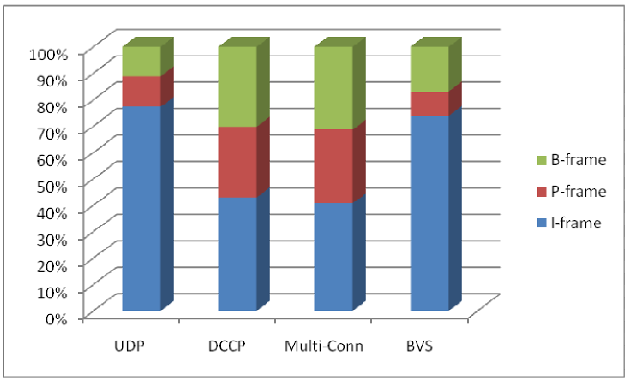

Figure 5. Breakdown by frame type of packet losses when uplink streaming

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