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Broadband Video Streaming During Backhaul WiMAX Handoff

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ABSTRACT

Broadband Video Streaming (BVS) with selective retransmission trades a reduced but acceptable video quality during IEEE 802.16e hard handoff (HHO) for improved end-to-end latencies. Both forms of BVS promise better video quality with an HHO than UDP transport or traditional congestion-controlled streaming.

INTRODUCTION

An IEEE 802.16e (mobile WiMAX) [1] can act as a backhaul network for an on-bus IEEE 802.11 network (as trialled in Stockholm, Sweden). As a bus travels along its fixed route, passengers can access the Internet via a set of WiMAX masts, with horizontal handoff occurring as the bus moves between masts in a point-to-multipoint access. Mobile TV is an attractive service for urban commuters but there is a possibility of interruption during handoff due to factors such as: route setup delay; signalling message overhead and processing time; and packet loss. Though the WiMAX forum specifies just 50 ms handoff delay, when combined with channel errors, simulations in this paper with the NIST handover module for ns-2 [2] suggest the overall effect [3], when combined with channel error on streamed video, may be unsatisfactory UDP transported video streams [4]. In fact, to guard against channel error this paper proposes a simple negative-acknowledgment (NACK) scheme which for convenience is called Broadband Video Streaming (BVS). In the variety examined (BVS-I), only intra-coded I-frame packets when lost are retransmitted, which has the effect of reducing streaming interruption during a hard handoff (HHO).

HANDOFF AND BVS-I

Mobile WiMAX supports three handoff mechanisms, but only the mandatory HHO at layer 2 can be accomplished with a single channel at any one time, thus reducing equipment cost and improving base station (BS) capacity. HHO employs a break-before-make procedure which reduces signalling. As is normal, a mobile station (MS) monitors signal strength from adjacent BSs, employing an hysteresis mechanism to avoid thrashing between BS. The MS must then: obtain uplink and downlink parameters; negotiate capabilities; gain security authorisation and exchange keys; register with the BS; and establish connections. It is expected that these mechanisms will be subsumed in the emerging IEEE 802.21.

In Fig. 1's scenario, a remote server at C streams video over the IP network, while node A sources to node B constant bitrate (CBR) data 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 CBR data at 1.5 Mbps with packet size 1 kB. The MS moves in parallel to the two BS, which are separated by 1.9 km. For BVS-I, in Fig. 2 at an MS a record is kept of packet sequence numbers available through the RTP header and, if an out of sequence packet arrives from the server, a NACK may be transmitted to the BS in the next 802.16e sub-frame for forwarding to the video server. The MS only transmits a NACK if this is the first time that particular packet has been lost. To reduce the overhead at the MS, the decision as to whether to retransmit a packet is left to the server. The server prevents transmission from its input buffer until a single retransmission of the missing packet in the sequence has taken place. Not shown in Fig. 2, is a holding buffer that retains sent packets in the case of the need for a retransmission. However, the server will only retransmit if the NACK refers to an Iframe packet. In Fig. 2, the packet is defined as of type A, as other forms of prioritized retransmit are possible, and, in fact, all packets are retransmitted in simple BVS.

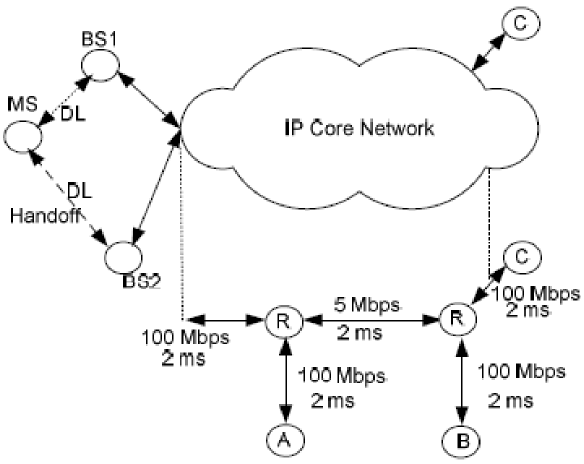

Fig. 1 Video streaming scenario.

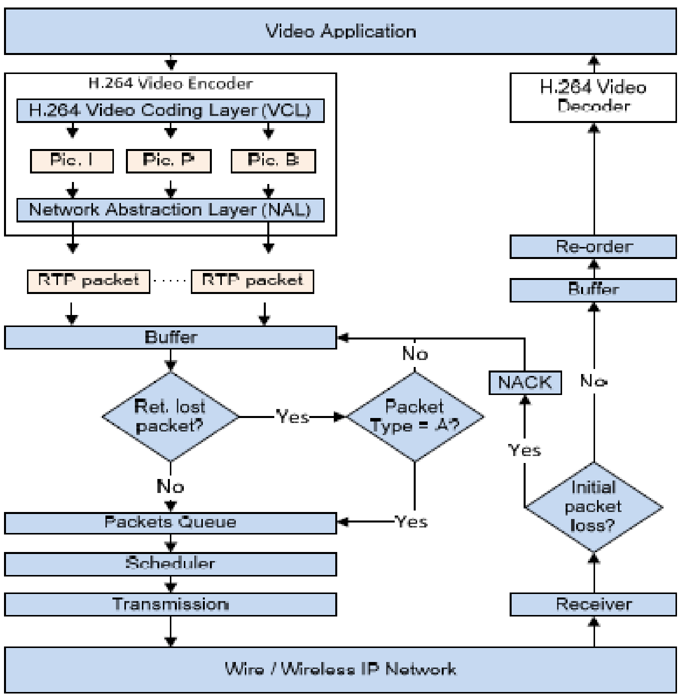

Fig. 2. Operation of BVS.

In tests, 35.5s of the reference *Paris* clip were H.264/AVC (Advanced Video Coding) variable bitrate encoded with Common Intermediate Format @ 30 Hz. Group of Pictures structure was IBBP… with an intra-refresh rate of 15. A Gilbert-Elliott channel model [5] modelled fast fading. 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 modelled by a Uniform distribution. The packet loss probability in the good state was fixed at 0.01 and the bad state probability (PB) was made variable. The WiMAX PHYsical settings were 5 ms Time Division Duplex (TDD) frame, 16-QAM ½, guard band 1/8, maximum packet length 1kB, raw data-rate 10.67 Mbps, range 1.0 km. Buffer sizes were set to 50 packets.

FINDINGS

In Figs. 3 and 4, BVS and BVS-I's objective video quality at the MS are compared for increasing speeds and variable channel conditions. At around 20 mps (45 mph), BVS-I's quality becomes unsatisfactory as its quality drops below 25 dB. At low bus speeds and shorter error bursts both BVS and BVS-I deliver 'good' quality video (above 31 dB). From Fig. 5, both BVS and BVS-I are superior to UDP-transport and TCP-Friendly Rate Control (TFRC) [7]. From the summary in Table 1, without retransmissions, packet loss approaches 10%. TFRC reduces its sending rate by increasing the inter-packet gap but only by doubling the sending period of *Paris*. BVS is better but its latencies are larger. However, by reduced retransmission BVS-I, end-to-end latencies are reduced. Simulation has shown that at speeds above 45 mps (100 mph), HHO latency grows rapidly.

CONCLUSION

Results show that at moderate speeds video quality during a WiMAX handover could improve by as much as 9 dB using BVS compared with UDP but there is a cost in end-to-end delay. This issue is resolved in BVS-I by selective retransmission by picture type, while video quality remains acceptable. The findings do not suggest acceptable streaming from a remote server at speeds much over 20 mps (45 mph), which is probably the maximum bus speed in large cities.

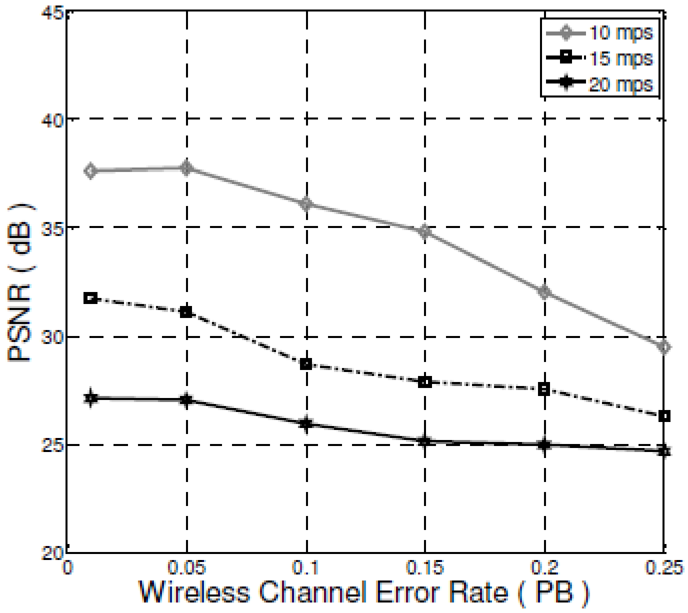

Fig. 3 Y-PSNR of BVS for different channel conditions.

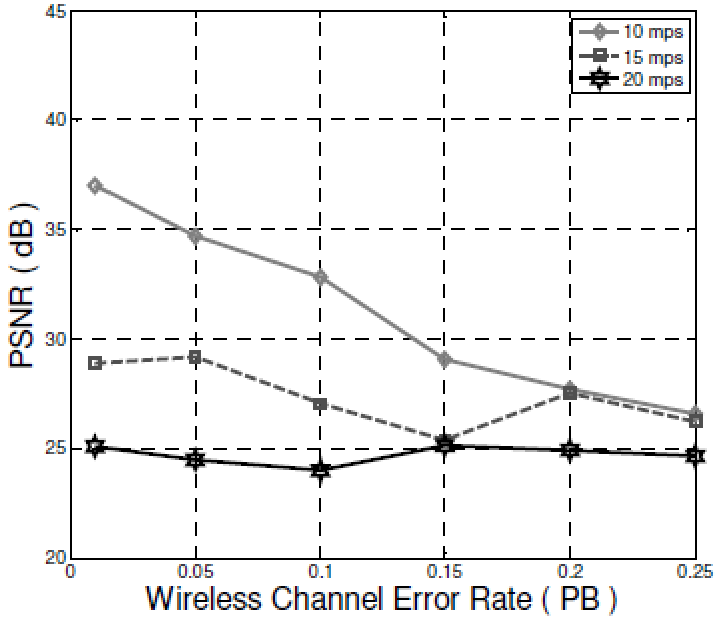

Fig. 4 Y-PSNR of BVS-I for different channel conditions.

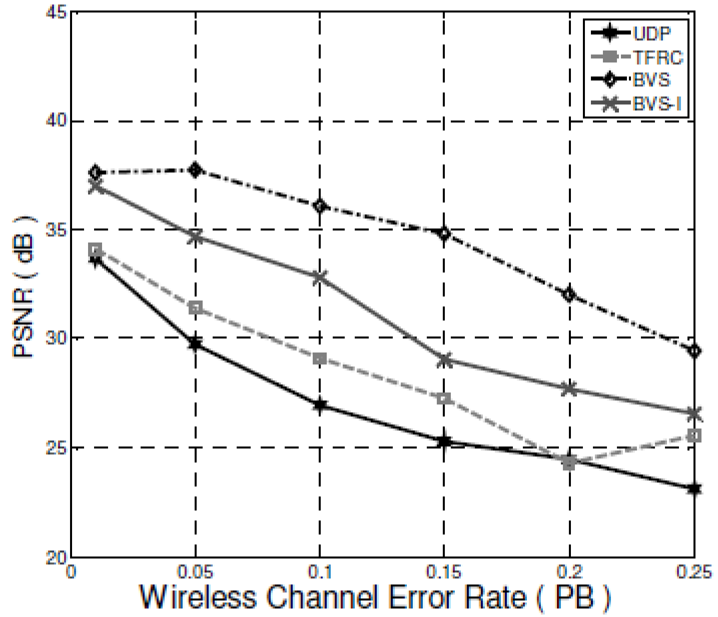

Fig. 5 Video quality from different protocols with a speed of 10 mps.

Table 1. Summary comparison at 10 mps (22 mph) with PB = 0.15.

	UDP	*TFRC*	*BVS*	*BVS-I*
Throughput (kbps)	762.9	371.1	819.2	766.4
Sending period (s)	35.43	72.97	35.70	37.49
Packet Loss (%)	9.75	7.29	1.69	3.81
Jitter (s)	0.008	0.066	0.007	0.008
Mean end-to-end delay (s)	0.008	0.007	0.015	0.011
Max. end-to-end delay (s)	0.256	0.247	0.631	0.329
PSNR (dB)	25.29	27.26	34.82	29.05
S.D. (dB)	3.28	3.31	3.42	4.91

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