Dual Handover vs. QoS for Real Time Broadband Video Streaming Over WiMAX Networks

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Abstract— **The consistent superiority in terms of reduced packet loss to enhance video quality it is what this paper proposing. Prior investigation of video streaming over wireless networks has assumed a single access point and a homogeneous wireless technology. With different channel effects and conditions a long side the Hard Handover (HHO) consequences on mobile broadband video streaming , IEEE 802.21, it is now becoming possible to offer seamless video streaming and harmonizing that influence. The paper presents a video streaming transport scheme that is more capable of exploiting the expected reduced latencies of real time video streaming content distribution networks. Broadband Video Streaming (BVS) with adaptive packet retransmission promises better video quality during an (HHO) than both raw UDP transport and traditional congestion-controlled streaming, making it attractive to mobile broadband video streaming services. It achieves this by distinguishing between high congestion and poor channel conditions, the latter of which an HHO induces, and by prioritized retransmission according to picture type.**

Keywords-component; QoS; Mobile IPTV; Handover ; Broadband Video Streaming; Media Transport; Wireless Networks; WiMAX

I. INTRODUCTION

The contribution of the paper is a transport method better tuned to the needs of IPTV (Internet Protocol TV) or any other real time video streaming application, paving the way for an extension to wireless networks of services such as time-shifted TV and video-on-demand. A key deficiency of some streaming schemes for wireless networks is that they do not account for the movement of the user between different access networks. However, seamless video streaming in which the video stream follows a user's multi-homed device [1] will increase the attraction of mobile IPTV. In fact, it is intuitively unlikely that a mobile user will stay close to the vicinity of one base station (BS) or access point during the course of a streaming session. For example, a typical commuter to work carrying some form of portable display device may stream via a broadband wireless BS such as IEEE 802.16e WiMAX [2], while outdoors, but once indoors will transfer to streaming from an IEEE 802.11 access point (AP). Therefore, this paper considers direct transport of an IPTV video stream to a WiMAX BS and transport in the course of which a vertical handover occurs.

An important difference between IPTV delivery to mobile devices and broadband access is the possibility of vertical handovers, which can cause disruption to real-time video streaming, due to factors such as: route setup delay; signalling message overhead and processing time; and packet loss. This paper proposes a lightweight form of IPTV transport based on negative acknowledgments, which, during video streaming of time-shifted TV, aims to improve video quality over that of raw UDP transport and traditional congestion controllers such TCP-Friendly Rate Control (TFRC) [3]. The Broadband Video Streaming (BVS)-adaptive (A) scheme is simulated across the delivery path from a remote video server on an unmanaged wired core network to either an IEEE 802.11 access point or an IEEE 802.16e (mobile WiMAX) base station (BS). An underlying IPTV content delivery network is assumed to reduce the video delivery path length, thus reducing the latency of the single negative acknowledgments employed.

BVS-A, by virtue of its adaptive structure, is designed to react both to traffic congestion and to poor channel conditions. It does this by selecting packets by their video picture type according to traffic conditions. Consequently, when a vertical handover occurs, BVS-A can react as if poor channel conditions have occurred rather than traffic congestion. By contrast, TFRC has only one mode of response, reacting to traffic congestion, as a result of its provenance as a wired Internet congestion controller. Nevertheless, TFRC is a standardized controller that has been widely adopted. For example, in [4] it was tested as a controller for streaming over a Long Term Evolution (LTE) link.

II. HANDOVER MECHANISMS

It is expected that these mechanisms for vertical handover, will be subsumed in the emerging IEEE 802.11.21 [5] standard. IEEE 802.21 specifies tools to exchange information, commands, and events but does not standardize the execution mechanism. The architecture of IEEE 802.2's MIH appears in Figure 1. In this paper for mobility management, mobile IP (MIP) is assumed rather than the Session Initiation Protocol (SIP). Mobile IP acts as an upper layer client of 802.21's MIH function (MIHF). The MIHF itself lies between layer 2 (Datalink — Medium Access Control (MAC)) and layer 3. Layers 3 and above can obtain information, receive event notifications, and issue commands via MIH, while the MIHF provides a Service Access Point to layer 2 and below. Network information includes MAC addresses, security information, and channel information. Events include link parameter changes and link status changes.

There are several ways to improve handover management for real-time services. The first way is to make structural changes to the way a handover operates such as reducing the

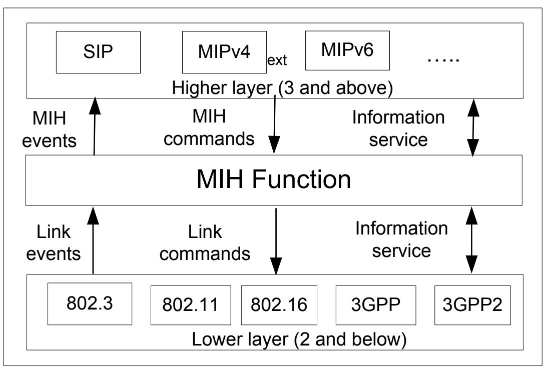

Figure 1. Architecture of IEEE 802.21's MIH.

latency of the network selection process [6] and/or the mobility management [7]. It is also possible to act at the application layer through increased protection against packet loss and delay. If the handover can be anticipated then pre-buffering [8] at the client is possible. In [8], it is noted that, receiver notification of increased packet losses and round-trip times are insufficient handover indicators, because they occur after the event. Instead, in [9] information about an impending handover is passed up the protocol layers. Alternatively, this paper seeks to adapt the transport scheme to the needs of handover and video streaming. The advantage of this second way is that it neither alters the way handovers are controlled nor requires special intervention for video applications.

III. ADAPTIVE BROADBAND VIDEO STREAMING

The BVS-A scheme introduces a single negative acknowledgment (NACK)[10] to User Datagram Protocol (UDP) transport. At the receiver, a record is kept of packet

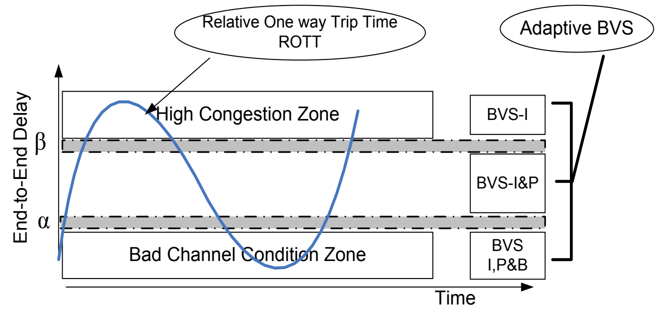

Figure 2. The Spike scheme applied to BVS-A

sequence numbers 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 across the broadband wireless link. During prioritized operation a decision is made to resend a video packet according to the picture type of the packet that has been lost, reflecting the importance to the reconstruction of the video of that packet's picture type.

BVS-A has been applied according to the Spike scheme [13]. In the Spike scheme, a peak or spike in the Relative One-way Trip Time (ROTT) indicates the presence of congestion. 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 2, in the bad channel zone, packets from all picture types are re-transmitted when necessary, in order to reconstruct the video sequence. However, if there is limited congestion and moderate problems within the wireless channel then only intra-coded I- and inter-coded P-picture packets are re-sent 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. I-pictures are always re-transmitted in whatever zone as they affect the reconstruction of the rest of a Group of Pictures (GoP).

IV. VIDEO AND NETWORK SETTINGS

The H.264/AVC (Advanced Video Coding) codec [11] was employed. The Paris video sequence was chosen as a test, as it is sufficiently long to judge the impact of network conditions. The encoding settings were as follows. Variable Bit-Rate (VBR)-encoding at 30 frame/s was used with Common Intermediate Format (CIF) (352 × 288 pixel/frame) and the quantization parameter (QP) set to 26 (from a range 0 to 50). The Peak-Signal-to-Noise Ratio (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. Thus because each slice's header was contained in the same packet as the matching slice, decoder loss of synchronization is avoided. The Paris clip contains a bookcase in the background with high spatial coding complexity.

On the other hand, the two seated TV studio commentators contribute moderate motion and hence reduced temporal coding complexity. Quality-of-experience tests show [12] 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). The Intra-refresh rate was every 15 pictures 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. Other forms of error concealment increase decoder complexity.

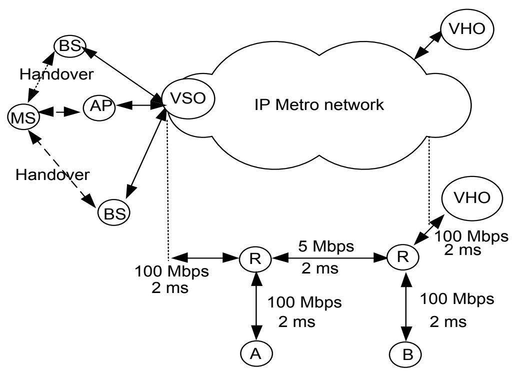

Figure 3. Video streaming scenario with dual handovers as video is streamed from the VHO

Table 1. IEEE 802.16e parameter settings.

Parameter	*Value*
PHY	OFDMA
Frequency band	5 GHz
Bandwidth capacity	10 MHz
Duplexing mode	TDD
Frame length	5 ms
Max. packet length	1024 B
Raw data rate (downlink)	10.67 Mbps
IFFT size	1024
Modulation	16-QAM 1/2
Guard band ratio	1/16
MS transmit power	245 mW
BS transmit power	20 W
Approx. range to MS	1 km
Antenna type	Omni-directional
Antenna gains	0 dBD
MS antenna height	1.2 m
BS antenna height	30 m
Receiving threshold	7.91e-15 W

Table 2. IEEE 802.11b parameter settings.

Parameter	*Value*
PHY	DSSS
Frequency band	2.4 GHz
Bandwidth capacity	20 MHz
Max. packet length used	1024 B
Raw data rate (downlink)	11 Mbps
AP transmit power	0.0025 W
Approx. range	100 m
Receiving threshold	6.12e-9 W

A Gilbert-Elliott two-state channel model modeled error bursts during 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 modeled 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. Table 1. The WiMAX PHYsical layer 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. Vertical handover was modelled with the NIST IEEE 802.21 module for the ns-2 simulator, which is tied to the IEEE 802.11b model (see Table 2) built into ns-2 operating at 11 Mbps.

(Available from http://w3.antd.nist.gov/seamlessandsecure/ [accessed Jul. 2010].)

In Figure 3's dual handover scenario, a remote server at the video head office (VHO) streamed video over the IP network to the video serving office (VSO) in the content delivery network, while node A sourced to node B constant bitrate (CBR) data at 1.5 Mbps with packet size 1 kB and sank a continuous TCP FTP flow sourced at node B. Node B also sourced an FTP flow to the BS and CBR data at 1.5 Mbps with packet size 1 kB. The MS moved in parallel to the first BS then to the wireless access point (AP) and on to a second WiMAX BS, each of which transmitters were separated by 0.825 km.

V. EVALUATION

In this Section, the behavior of the transport schemes have been analyzed from the packet loss perspective and the final outcome of the video streaming (PSNR).

At low speeds, packet loss for TFRC is comparatively higher than that for UDP With increasing speed, UDP exhibits greater packet loss than TFRC. BVS-A consistently shows a lower packet drop with increasing speeds, Figure 4.

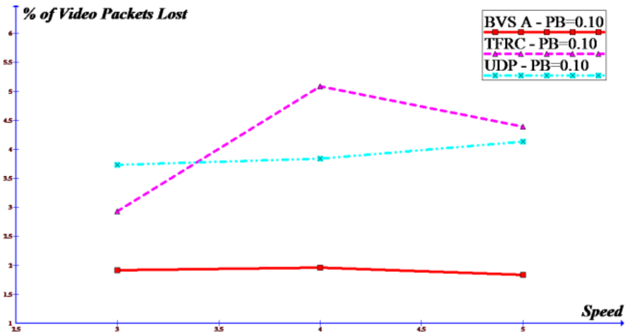

Figure 4. Packet Loss for Different Mobile Speed

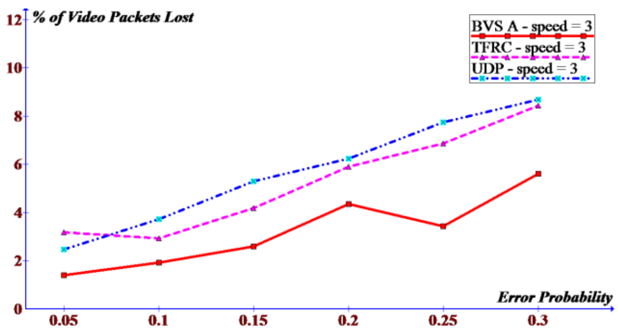

Figure 5 Packet Loss for Different Channel Conditions

Figure 5. shows the different in percentage with changing of the channel conditions, where the BVS-A without fail shows a lower packet drop with increasing in horrific channel conditions

From Figure 6, one observes a decline in objective video quality as the speed of the user increases. The BVS-A quality remains good (above 30 dB) throughout, whereas TFRC offers less than raw UDP at the same bad-state channel setting (PB = 0.10). In fact, TFRC's sending time for the entire clip is longer than UDP or BVS-A, as it reacts to congestion by lengthening the inter-packet gap. At a speed of 3 mps, Figure 7, one sees the response as channel conditions worsen.

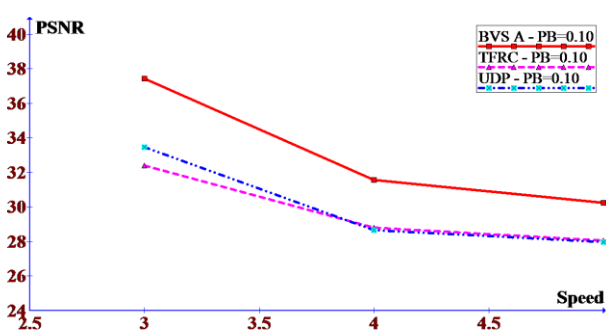

Figure 6. Video quality (Y-PSNR) of BVS-A for varying mobile device speeds.

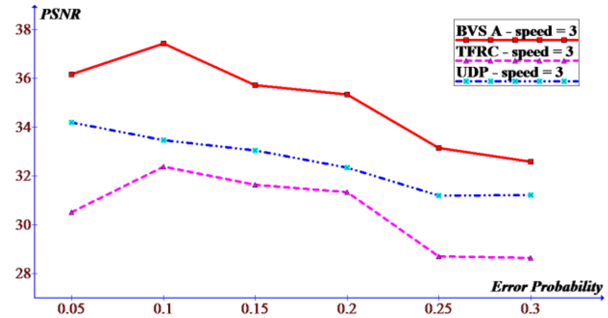

Figure 7 Video quality (Y-PSNR) of BVS-A for different channel conditions.

That this is not a monotonic decline is due to the type of packets that happen to be lost, as Figure 8, illustrates. Recall that I-pictures generate more packets than P- and B-pictures. While I-picture packets are dropped in a similar ratio to the other types with UDP transport, TFRC's mode of control actually discriminates against I-pictures leaving them exposed to the channel for longer periods, especially during handovers. Consequently, video quality is reduced.

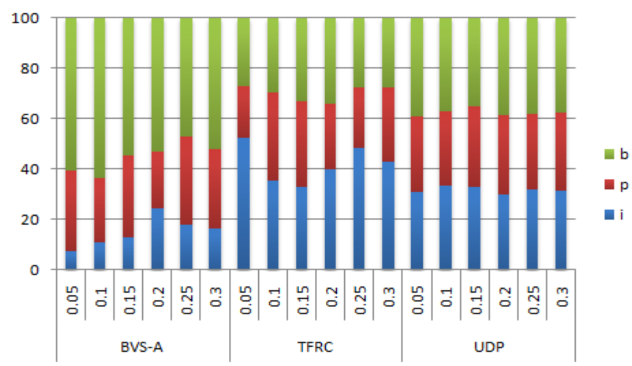

Figure 8 Frame type packet loss percentage for different channel conditions (PB = 0.05 ... 0.3) with a mobile speed of 3 mps.

VI. CONCLUSION

Adaptive broadband video streaming, by preserving anchor frames during handover, improves upon traditional congestion control, which seems ill suited to realistic scenarios when handovers take place.

In low latency conditions, this paper has proposed a lightweight transport method to minimize the impact of congestion control delays. In fact, the method seems to be sufficient in the presence of network congestion affecting the path from the video server to the mobile device. In comparison, TFRC, which requires an acknowledgment after every packet transmission can be more affected by congestion in the feedback path than the BVS-A scheme which only uses acknowledgments after the first packet loss. TFRC is also affected by its inability to distinguish between those packet losses due to congestion (on the streaming path) and those due to packet drops on the wireless channel. It was also found that when vertical handover takes place results are sensitive to the speed of motion of the user. However, if the user is walking from outdoor communication with a WiMAX BS to indoor communication with a WiFi AP, the speed effect is less. Next generation mobile networks will support seamless motion across heterogeneous networks, thus raising user expectations that mobile IPTV will be able to follow the mobile device. Future work will investigate the temporal behavior different wireless technology (like LTE advance) and characterize more clearly the nature of the impact of MS speed.

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