Seamless Transport for Wireless IPTV Video Streaming

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Abstract. Prior investigation of video streaming over wireless networks has assumed a single access point and a homogeneous wireless technology. With multi-homed devices and the harmonizing influence of IEEE 802.21, it is now becoming possible to offer seamless video streaming. The paper presents a video streaming transport scheme that is more capable of exploiting the expected reduced latencies of IPTV content distribution networks. Simulations show the consistent superiority in terms of reduced packet loss and video quality of a simple NACK-based scheme over a traditional congestion controller and baseline UDP transport. Results are shown without and with vertical handover between a WiMAX base station and a WiFi access point.

Keywords: IEEE 802.16e, IEEE 802.21, media transport, mobile IPTV, vertical handover, WiMAX

1 Introduction

The contribution of the paper is a transport method better tuned to the needs of IPTV (Internet Protocol TV), 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.

In this paper we have concentrated on mobile WiMAX [3], which provides broadband wireless access independently of a pre-existing cellular system. It is not reliant on hardware authentication[1]. Furthermore, it can also deliver data in a cost-effective way [4] at approximately 3–4 times the rate of current 3G cellular systems. One additional short-term advantage is that mobile WiMAX is currently deployed, rather than in development or waiting ratification. Thus, mobile WiMAX is a promising alternative way to deliver multimedia services, supported as it is by leading companies such as Intel and Freescale. Intel have completed the specification of IEEE 802.11m [5], which will fulfill the requirements of 4G International Mobile Telecommunications-Advanced (IMT-Advanced), namely streaming throughput of 2-50 Mbps, with live and non-live latencies of 100 ms and 1 s respectively.

In the proposed 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 also compared with selective NACKs which are reserved for lost anchor frame packets from the video stream. Both are assessed for their performance during vertical handover and compared to a traditional congestion controller and unembellished UDP transport. For ease of reference in the following, the NACK enhancement to UDP is called broadband video streaming (BVS).

The remainder of this paper is organized as follows. Section 2 relates the background to IPTV streaming over IEEE 802.16e broadband wireless access, as well as a general introduction to vertical handovers during streaming. In the context of vertical handovers, the emerging IEEE 802.21 standard is introduced. Section 3, describes the proposed NACK-based transport scheme. The simulation model in Section 4 is applied in Section 5, which compares streaming behaviour without a vertical handover to that with. Finally, Section 6 draws some conclusions.

2 Background

In this Section background to the study is provided, including the IPTV architecture for mobile WiMAX and vertical handover mechanisms.

2.1 WiMAX technology for IPTV

Mobile WiMAX was introduced in 2007, as part *e* of the IEEE 802.16 standard, to strengthen the fixed WiMAX part d standard of 2004. Like many recent wireless systems, part *d* utilized Orthogonal Frequency Division Multiplexing (OFDM) as a way of increasing symbol length to guard against multi-path interference. The sub-carriers inherent in OFDM were adapted for multi-user usage by means of Orthogonal Frequency Division Multiple Access (OFDMA), allowing subsets of the lower data-rate sub-carriers to be grouped for individual users. Sub-channel spectral allocation can range from 1.25 MHz to 20 MHz. Adaptive antenna systems and Multiple Input Multiple Output (MIMO) antennas can improve coverage. Basic Multicast and Broadcast Services (MBS) are supported by mobile WiMAX. IEEE 802.16m [5] is expected to increase data rates to 100 Mbps mobile and 1 Gbps fixed delivery.

[1] Rather than a SIM card, WiMAX employs digital certificates in a Public Key Infrastructure.

However, 802.16m is not backwards compatible with 802.16e, though it does support joint operation with it.

For mobile WiMAX, capacity studies [4] suggest up to 16 mobile TV users per cell in a ‘lossy’ channel depending on factors such as the form of scheduling and whether MIMO antennas are activated. However, given the predicted increase in data rates arising from IEEE 802.16m, the number of unicast video users [3] with 4×2 Multi User (MU)-MIMO antennas, will be 44 at 384 kbps and 22 at 768 kbps in an urban environment. For a similar configuration but using 20 MHz rather than 80 MHz channels, the authors of [6] reported the number of unicast video users to be 11 and 6 depending on datarates.

An overview of how an IPTV system with WiMAX fixed or mobile delivery is presented in [7]. The system takes advantage of WiMAX’s point-to-multipoint (PMP) mode for the broadcast of TV channels. MPEG-2-transport system packets are multiplexed onto RTP/UDP/IP packets. Header suppression and compression techniques reduce the overhead. IPTV delivery has been tested [8] on a WiMAX testbed for downlink streaming of TV channels and uplink delivery of either TV news reports or video surveillance; refer to Fig 1a. However, that research did not consider the impact of the intervening wired network connecting the WiMAX BSs. In [9], 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, Fig. 1b, assumes a super head-end (SHE) distributor of content across a core network to regional video hub offices (VHOs).

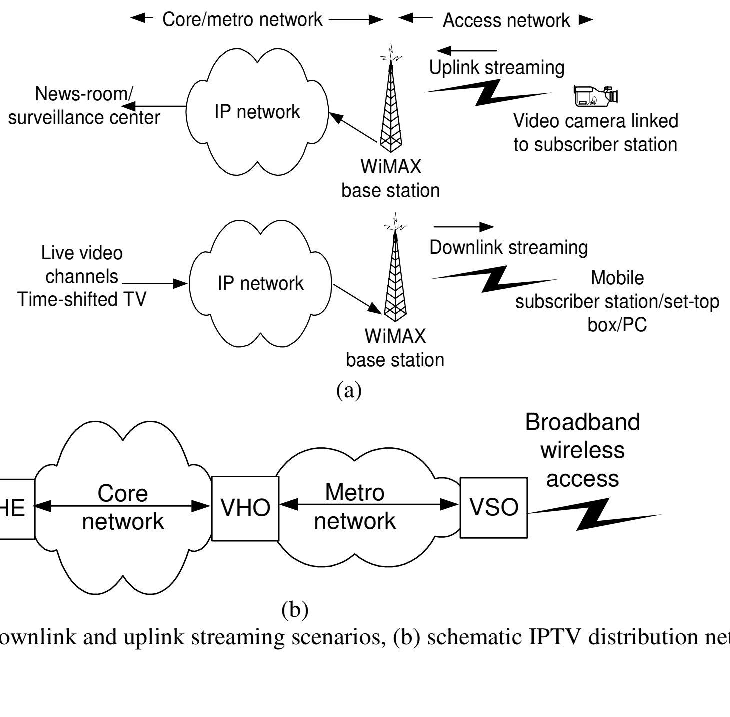


Fig. 1. (a) Downlink and uplink streaming scenarios, (b) schematic IPTV distribution network.

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.

2.2 Handover mechanisms

It is expected that these mechanisms for vertical handover, will be subsumed in the emerging IEEE 802.11.21 [10] 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 Fig. 2. 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 latency of the network selection process [11] and/or the mobility management [12]. 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 [12] at the client is possible. In [13], 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 [14] 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.

3 Transport schemes

This Section introduces the proposed NACK-based transport scheme but before that considers direct UDP transport and after that a traditional congestion controller, namely TCP-friendly Rate Control (TFRC).

3.1 Operation of UDP

Various methods of improving upon UDP offer the possibility of improved media transport without the overhead of application layer congestion control superimposed upon UDP transport. This is particularly the case if the latency between the streaming server and the mobile device is relatively small. Because content management can bring the server closer to the access network [9] reduced latencies are likely to occur.

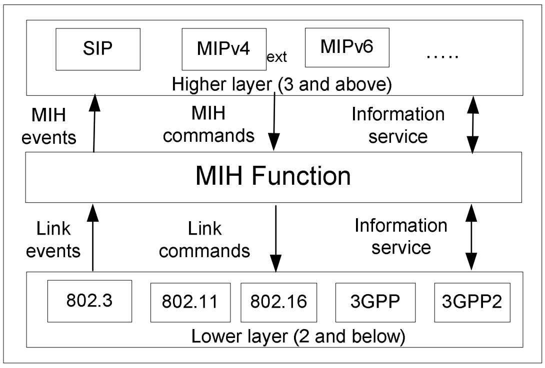


Fig. 2. Architecture of IEEE 802.21's MIH.

Cumulative delay for IPTV should not exceed 50 ms according to the stringent conditions mentioned in the industry report of [15] and ideally should be as low as 9 ms.

Unembellished UDP has been used for IPTV transport over IEEE 802.16e systems [8]. However, UDP packet losses can seriously harm a compressed video stream. Bell Laboratories introduced a reliable form of UDP, R-UDP, see [16], and there is also a coincidently-named R-UDP protocol employed by Microsoft for IPTV service delivery over multicast networks.

3.2 Operation of proposed NACK-based scheme

Fig. 3 is a general representation of the processing involved in the scheme, which for convenience of reference, as previously mentioned, for convenience we name Broadband Video Streaming (BVS), showing the NACK response of the receiver. The following describes the operation assuming downlink streaming from to an MS. At an 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 may be transmitted to a BS or Access Point (AP) for forwarding to the video server. The MS 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. This variety of BVS is named BVS-I when only intra-coded I-pictures are retransmitted. The results inspect the trade-off between reduced retransmissions, latency of handover, but some reduction in video quality.

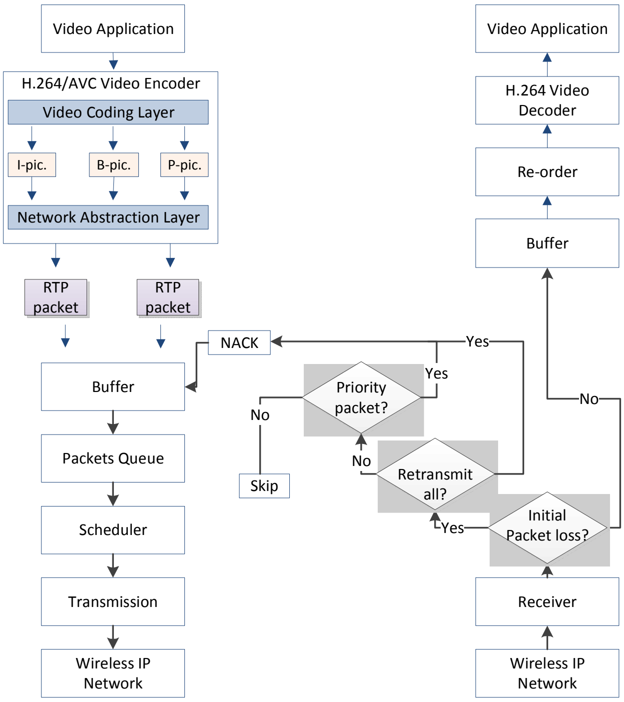


Fig. 3. Operation of BVS NACK enhancement to UDP.

3.3 Operation of TFRC

A common alternative to UDP is a standards-based form of transport. TFRC [17] have been used directly [18] or in cross-layer form [19] for video streaming over wireless networks. In TFRC [17], the packet loss rate is found at the receiver and returned to the sender via a feedback channel. Acknowledgment messages, once per packet, are returned in reliable fashion by TCP transport. The round-trip time estimate is subsequently calculated at the sender. The sending rate is then adjusted accordingly. A throughput equation models TCP New Reno to find the sending rate in (1):

$$TFRC\,(t_{rtt}, t_{rto}, s, p) = \frac{s}{t_{rtt}\sqrt{\dfrac{2bp}{3}} + t_{rto}\,\min\left(1, 3\sqrt{\dfrac{3bp}{8}}\right) p(1+32p^2)}$$

where t_{rtt} is the round-trip time, t_{rto} is TCP's retransmission timeout, s is the segment size (TCP's unit of output) (herein set to the packet size), p is the normalized packet loss rate, w_m is the maximum window size, and b is the number of packets acknowledged by each ACK. b is normally set to one and $t_{rto} = 4t_{rtt}$. It is important to notice that t_{rto} comes to dominate TFRC's behavior in high packet loss regimes. Clearly packet loss and round-trip time cause the throughput to decrease in (1), whereas other terms in the denominator are dependent on these two variables.

4 Simulation model

Simulations were conducted for IEEE 802.16e in two scenarios: 1) streaming without handover; and streaming with vertical handover from and to an IEEE 802.11 WLAN. The simulations were conducted with the National Institute of Standards and Technology (NIST) IEEE 802.21 mobility add-on for the well-known ns-2[2] simulator, together with the NIST add-on for IEEE 802.16 which includes IEEE 802.16e scanning and handover support. The NIST 802.21 add-on is tied to the IEEE 802.11b model built into ns-2 (hence the use of this version of IEEE 802.11 in simulations). 802.11b operated at 11 Mbps. 25 runs per data point were averaged (arithmetic mean) and the simulator was first allowed to reach steady state before commencing testing.

In Fig. 4's vertical handover scenario, a remote server at the Video Head Office (VHO) (see Fig. 1b) 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. Node A also sinks a continuous TCP FTP flow sourced at node B. Node B sources an FTP flow to the BS and CBR data at 1.5 Mbps with packet size 1 kB. The Video Serving Office (VSO) then transmits the video stream towards the appropriate access point (see Fig. 1b). The MS moves in parallel to the BS and the AP, which are separated by 1.9 km.

The behavior when *no* handover from a WiMAX BS occurs was also modeled, when Fig. 4's topology was again adopted but without movement of the MS to the AP.

4.1 Wireless configuration

To evaluate the proposal, transmission over WiMAX was carefully modeled. The PHYsical layer settings selected for WiMAX simulation are given in Table 1. The antenna heights and transmit power levels are typical ones taken from the Standard [2]. 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 802.16 Time Division Duplex (TDD) frame length was set to 5 ms, as only this value is supported in the WiMAX forum adaptation of the Standard. The data rate results from the use of one of the mandatory coding modes [2] 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 physical-layer coding rate are required to achieve a datarate of 10.67 Mbps over the downlink. Buffer sizes were set to 50 packets (a

[2] Available from http://w3.antd.nist.gov/seamlessandsecure/ [accessed July 2010]

single MAC Service Data Unit with a MAC Protocol Data Unit). This buffer size was selected as appropriate to mobile, real-time applications for which larger buffer sizes

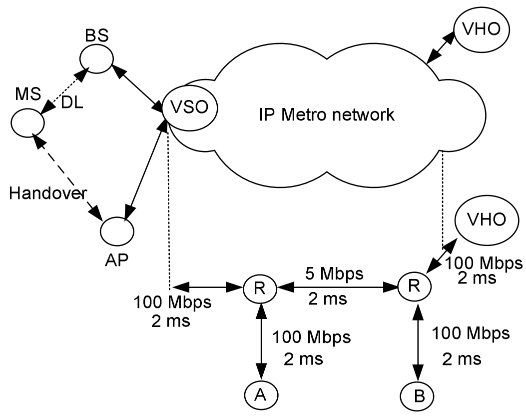


Fig. 4. IPTV video streaming vertical handover scenario, R = router.

might lead both to increased delay and larger memory energy consumption in mobile devices. Settings for the IEEE 802.11 AP are given in Table 2, while the operation of 802.11 is assumed to be well-known.

'Bursty' errors as occur during fast fading were generated by the Gilbert-Elliott model [20], which is a form of hidden Markov model with internal good and bad states. If *PGB* and *PBG* are the probabilities of going from the good to bad state and from going from the bad to good state respectively, then

$$\pi_G = PBG/(PBG + PGB) \qquad (2)$$

$$\pi_B = PGB/(PBG + PGB) \qquad (3)$$

are the steady state probabilities of being in the good and bad states. The probability of remaining in the good state, *PGG*, was set to 0.95 and of remaining in the bad state, *PBB*, was 0.94, with both states modeled by a Uniform distribution. The packet loss probability in the good state, *PG*, was fixed at 0.01 and the bad state probability, *PB*, was defaulted to 0.05. The Gilbert-Elliott model was selected, as it is the presence of burst errors [21] that mostly affects the quality of compressed video.

4.2 Video settings

The H.264/AVC (Advanced Video Coding) codec [25] 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 [26] 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.

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

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

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

DSSS=Direct-Sequence Spread Spectrum

5 Findings

In this Section, the behavior of the transport schemes with and without handover is analyzed.

5.1 Behavior without handover

The first findings presented concern streaming without handover as occurs in Fig. 4's topology but without the presence of the additional BS or AP. In Fig. 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 Fig. 4 are the losses after retransmissions and do not directly show packet losses across the transmission paths. For TFRC downlink streaming, the majority of packet losses occur across the wireless link, as this protocol is able to respond to congestion across the metro 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. 6 shows the breakdown for downlink streaming. The number of packets dropped is greater in uplink streaming for TFRC, as most packets are dropped over the wireless link, which is encountered first.

From Table 3, the percentages of packet losses for UDP transport for downlink streaming are much higher than the other methods. Though TFRC is 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 control 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 between the ITU's mean opinion score rankings and PSNR [27]). However, BVS uplink streaming results in 'good' quality video (just). The mean end-to-end delay of TFRC is lower again than UDP and BVS. This is because TFRC reduces its sending rate, resulting in less queuing time. From Table 3, UDP and BVS's sending period is approximately the same and close to the duration of the *Paris* sequence. However for TFRC, packet losses on the wireless link cause excessive delay, as TFRC introduces large inter-packet gaps. 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 TFRC decreases congestion by increasing the inter-packet gap to too high a duration.

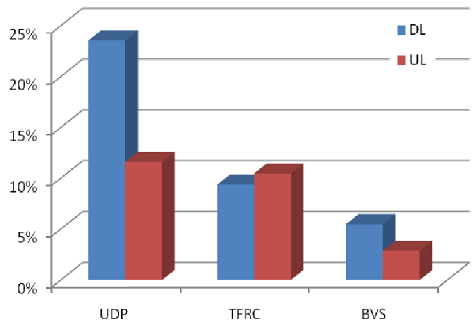


Fig. 5. Packet loss from streaming according to transport scheme and streaming direction for the *Paris* sequence.

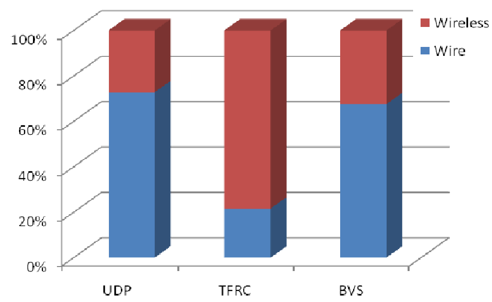


Fig. 6. Proportion of wired/wireless network packet losses for downlink streaming.

5.2 Behavior with vertical handover

This Section considers the overall effect of a single vertical handover between an IEEE 802.16e BS and an IEEE 802.11 AP using IEEE 802.21 mechanisms. Because the range of communication to the AP is much shorter (compare Tables 1 and 2) than for communication to the BS, the impact of channel error is reduced in the mean. In turn this means that the mean video quality is improved when a vertical handover occurs. Therefore, relative effects between the schemes are of interest and in Fig. 7, the packet loss statistics are analyzed by picture type. In this Section all streaming is via the downlink to the MS.

Fig. 7 shows that UDP and TFRC both incur high packet losses. They are also prone to drop the bigger I-pictures. Though TFRC is able to reduce its rate to reduce the effects of congestion compared to the BVS schemes it is not able to do as much to reduce these effects. Moreover as the speed of the MS increases to just 3 and 4 mps (6.7 mph, 10.8 kmh and 8.9 mph, 14.4 kmh respectively) packet losses increase considerably. In fact, all results are very sensitive to MS speed. The packet losses for the BVS schemes are after a single attempted retransmission, either all packets or only I-pictures in the case of BVS-I. If the retransmission does not succeed the packet is dropped. It is apparent that BVS-I does not help much in that respect.

Table 3. Mean performance metrics when downlink streaming *Paris* over an IPTV delivery network, DL = downlink, UL = uplink.

	UDP	**TFRC**	**BVS**
	Packets lost (%)		
DL	23.4	9.37	5.49
UL	11.6	10.46	2.87
	PSNR (dB)		
DL	18.01	24.55	27.62
UL	24.81	25.46	31.18
	End-to-end delay (s)		
DL	0.029	0.018	0.042
UL	0.049	0.016	0.062
	Sending period (s)		
DL	35.63	139.18	36.32
UL	35.62	134.00	35.77
	Jitter (s)		
DL	0.0097	0.0349	0.0079
UL	0.0084	0.0314	0.0076
	Throughput (kbps)		
DL	627	189	773
UL	751	197	809

The effect on video quality (PSNR) is shown for 4 mps in Fig. 8 according to increasingly adverse channel conditions. The gain of TFRC over UDP transport is small and quality remains fair. The gain from BVS over all schemes is large. For lower speeds, its advantage BVS maintains its video quality during better channel conditions, Fig. 9a, whereas other schemes are affected by MS speed. For worse channel conditions, Fig. 9b, BVS is unable to maintain video quality with increase in speed but is still superior. Speed of MS represents the impact of the vertical handover and it can be seen that its effect can be severe. Thus, a scheme that minimizes packet loss and, hence, video quality is to be preferred. From Fig. 10, the impact of retransmissions on packet end-to-end delay is not large and can be erratic. TFRC reduces packet end-to-end delay by reducing congestion but its loss from channel errors is too large to compensate for the reduction.

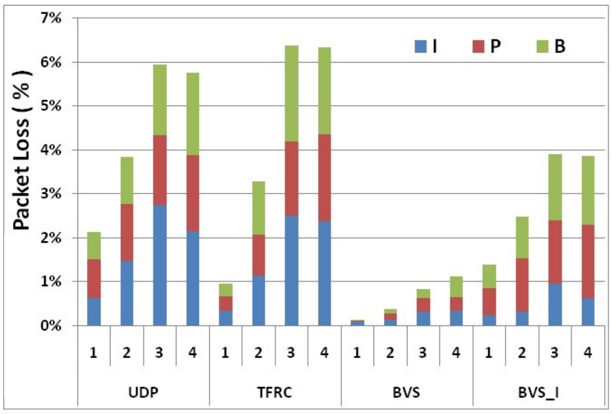


Fig. 7. Packet loss for different speeds (1= 1 mps, 2 = 2 mps, 3= 3 mps, 4 = 4 mps) and PB = 0.15when streaming *Paris* during a single handover from IEEE 802.16e to IEEE 802.11. Losses are arranged by picture type, I = I-picture, B = B-picture, P = P-picture.

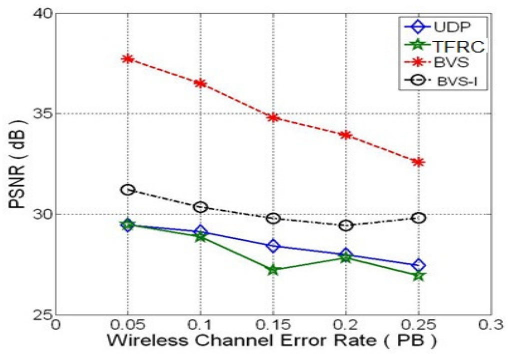
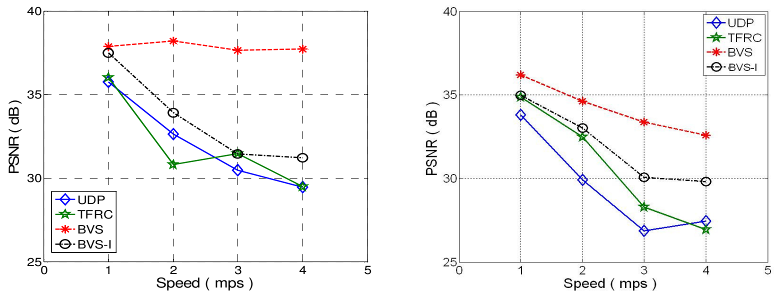


Fig. 8. Mean video quality (PSNR) when streaming Paris for MS speed 4 mps, with varying channel condition and a single handover from IEEE 802.16e to IEEE 802.11.

6 Conclusion

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 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



(a) **(b)**

Fig. 9. Mean video quality (PSNR) for a) PB= 0.05 b) PB= 0.25 and differing MS speeds when streaming *Paris* and a single handover from IEEE 802.16e to IEEE 802.11.

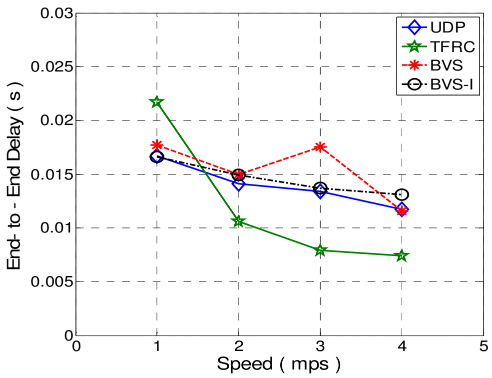


Fig. 10. Mean packet end-to-end delay for PB= 0.15 and differing MS speeds when streaming *Paris* and a single handover from IEEE 802.16e to IEEE 802.11.

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 during handover and characterize more clearly the nature of the impact of MS speed.

References

1. Wang, Q., Atkinsn, R., and Dunlop, J.: Design and Evaluation of Flow Handoff Signalling for Multihomed Mobile Nodes in Wireless Overlay Networks. Computer Networks. vol. 52, no. 8, pp. 1647–1674 (2008)
2. 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)

3. Jain, R., So-In, C., and Al-Tamimi, A.-K.: System-level Modeling of IEEE 802.16e Mobile WiMAX networks. IEEE Wireless Communications, vol. 15, no. 5, pp. 73–79 (2008)
4. So-In, C., Jain, R., and Tamimi, A.-K.: Capacity Evaluation for IEEE 802.16e Mobile WiMAX. J. of Comp. Syst., Networks, and Commun., [online journal] 12 pages, (2010)
5. Ahmadi, S.: Mobile WiMAX: A Systems Approach to Understanding IEEE 802.16m Radio Access Technology, Academic Press, Amsterdam (2011)
6. Oyman, O., Foerster, J., Tcha, Y.-j., and Lee, S.-C.: Towards Enhanced Mobile Video Services over WiMAX and LTE. IEEE Commun. Mag., vol. 48, no. 8, pp. 68–76 (2010)
7. Uilecan, I.V., Zhou, C., and Atkin, G.E.: Framework for delivering IPTV services over WiMAX wireless networks. In: IEEE Int. Conf. on Elec./Info. Technol., pp. 470–475 (2007)
8. Issa, O., Li, W., and Liu, H.: Performance Evaluation of TV over Broadband Wireless Access Networks. IEEE Trans. Broadcast., vol. 56, no. 2, pp. 201–210 (2010)
9. Degrande, N., Laevens, K., and De Vleeschauwer, D.: Increasing the User perceived Quality for IPTV Services. IEEE Commun. Mag., vol. 46, no. 2, pp. 94–100 (2008)
10. Taniuchi, K. et al.: IEEE 802.1: Media Independent Handover: Features, Applicability, and Realization. IEEE Commun. Mag., vol. 47, no. 1, pp. 112-120, (2009)
11. Lee, S.K., Sriram, K., Kim, K., Kim, Y.H., and Golmie, N.: Vertical Handoff Decision for Providing Optimized Performance in Heterogeneous Networks. IEEE Trans. Veh. Technol., vol. 58, no. 2, pp. 865–881 (2009)
12. Kong, K.-S., Lee, W., Han, Y.-H., and Shin, M.-K.: Handover Latency Analysis of a Network-based Localized Mobility Management Protocol. In: IEEE Int. Conf. on Commun., pp. 5838–5843 (2008)
13. Lee, D., Kim, J.W., and Sinha, P.: Handoff-aware Adaptive Media Streaming in Mobile IP Networks. In: Int. Conf. on Info. Networking, (2006)
14. Ishihara, S., Koyama, K., Miyamoto, G., and Kuroda, M.: Predictive Rate Control for Realtime Video Streaming with Network Triggered Handover. In: IEEE Wireless Commun. and Networking Conf., pp. 1335–1340 (2005)
15. Agilent Technologies: Validating IPTV Service Quality under Multiplay Network Conditions. White Paper (2008)
16. Partridge, C., and Hinden, R.: Version 2 of the Reliable Data Protocol. IETF, RFC 1151 (1990)
17. Handley, M., Pahdye, J., Floyd, S., and Widmer, J.: TCP-Friendly Rate Control (TFRC): Protocol Specification. IETF, RFC 3448 (2003)
18. Tappayuthpijam, K., Liebl, G., Stockhammer, T., and Steinbach, E.: Adaptive Video Streaming over a Mobile Network with TCP-Friendly Rate Control. In: Int. Conf. on Wireless Commun. and Mobile Computing, pp. 1325–1329 (2009)
19. Görkemli, B., Sunay, M.O., and Tekalp, A.M.: Video Streaming over Wireless DCCP. In: IEEE Int. Conf. on Image Processing, pp. 2028–2031 (2008)
20. Haßlinger, G., and Hohlfeld, O.: The Gilbert-Elliott Model for Packet Loss in Real Time Services on the Internet. In: 14th GI/ITG Conf. on Measurement, Modelling, and Evaluation of Computer and Commun. Systs., pp. 269–283 (2008)
21. Liang, Y.J., Apostolopoulos, J.G., and Girod, B.: Analysis of Packet Loss for Compressed Video: Effect of Burst Losses and Correlation between Error Frames. IEEE Trans. Circuits Syst. Video Technol., vol. 18, no. 7, pp. 861–874, (2008)
25. Wiegand, T., Sullivan, G.J., Bjontegaard, G. and Luthra, A.: Overview of the H.264/AVC Video Coding Standard. IEEE Trans. Circuits Syst. Video Technol, vol..13, no. 7, pp. 560–576 (2003)
26. Agboma, F., and Liotta, A.: Addressing User Expectations in Mobile Content Delivery. Mobile Information Systems, vol. 3, nos. 3/4, pp. 153–164 (2007)
27. Muntean, G., Perry, P., and Murphy, L.: A New Adaptive Multimedia Streaming System for all-IP Multi-Service Networks. IEEE Trans. Broadcast. vol. 50, no.1, pp. 1-10 (2004)