Lightweight IPTV Transport Option for Wireless Broadband Access

Salah Al-Majeed, Martin Fleury, *Member*, *IEEE*, Ismail Ali, and Mohammed Ghanbari, *Fellow, IEEE*

ABSTRACT

As intelligent content management reduces the latency between IPTV video serving office and mobile device, it becomes feasible to introduce a limited form of negative acknowledgments. Scalable video lends itself to selective acknowledgment, as does selection by video picture type. The defining characteristic of mobile video transport schemes is the response to handover. In this paper, it is demonstrated that the scalable selective negative acknowledgment transport scheme proposed decreases throughput but can retain acceptable video quality during vertical handover from an outdoors WiMAX system and an indoors IEEE 802.11 system.

***Index Terms*—H.264/SVC, vertical handover, video streaming, WiMAX**

I. INTRODUCTION

As IPTV extends to mobile devices [1], suitable transport methods are sought that can adapt streaming of time shifted or live Internet Protocol TV (IPTV) to broadband wireless access networks from a remote content server. However, the problem cannot just be reduced to providing a solution to congestion and error control over a heterogeneous network, because mobile devices participate in handover between wireless networks, which in the case of vertical handover will be between different wireless technologies. Though enhanced signalling and mobility management [2] has a role, the transport mechanism can reduce delays arising from handovers by limiting application-layer messaging at the time of handover. In addition, because content-delivery networks have brought the point of video distribution nearer to the mobile user [3], feedback latency to the video serving office will be reduced. This implies that a transport mechanism can exploit a limited form of feedback without undue impact on video display deadlines.

Taking the need to support vertical handover and the reduced feedback latency leads to what we term a semi-reliable version of UDP acts as a congestion controller over UDP, which is imposed at the application layer in a similar way to that of the TCP-Friendly Rate Control (TFRC) [4] acts as a congestion controller over UDP and has now been used as such in Google Talk’s videophone application. Similarly, Bell Labs introduced a reliable form of UDP, R-UDP [5] and there is also an R-UDP protocol employed by Microsoft in their Media Room product for IPTV service delivery over multicast networks. In the proposed lightweight transport 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. For ease of reference the scheme is called Broadband Video Streaming (BVS). The scheme comes in several different flavours, as it is possible to employ selective NACKs depending on the priority of the compressed video data type. Experiments have been conducted with scalable video, namely the Scalable Video Extension (SVC) [6] of the H.264 codec standard. In receiver-based selective NACK using SVC, BVS-s, the scheme limits NACKs to base-layer packets and can be extended to those designated packets from the Medium Grain Scalability (MGS) layer that other packets’ data are dependent on. This is compared to the video quality arising from single layer H.264/AVC (Advanced Video Codec). In the single layer variety, the selective variety NACKs intra-coded pictures when these are lost, which is designated BVS-I.

In the time division duplex (TDD) form of multiplexing favoured by broadband wireless access networks such as IEEE 802.16e (mobile WiMAX) [7], NACKs come almost for free by virtue of the return sub-frame. Consequently, the lightweight transport scheme’s behaviour during vertical handover was tested for a transition between outdoor WiMAX and indoors IEEE 802.11. The feasibility of broadband video streaming has been established [8] by means of a live WiMAX testbed. However, that study concentrated on varying the configuration parameters and used UDP-transported streams seemingly without congestion or error control. Other work on WiMAX in [9] was primarily concerned with combining true and near-video-on-demand, providing a solution to how content should be allocated between the two services. Therefore, it further illustrates a scheme to bring content nearer the user. In [10], adaptive multicast streaming was proposed using H.264/SVC. Fixed WiMAX (IEEE 802.16d) channel conditions were monitored in order to vary the bitrate accordingly. Unfortunately, the subsequent decision of the JVT standardization body for H.264/AVC not to support fine grained scalability (FGS) implies that it will be harder to respond to channel volatility, though coding efficiency is improved. Other work has also investigated combining scalable video, with multi-connections in [11] and in comparison with H.264/AVC in [12].

The remainder of this paper is organized as follows. Section II describes the background to the investigations in terms of vertical handover mechanisms and the scalable variety of H.264. Section III presents the proposed BVS algorithm along with the scalable variety of selective NACKs. Section IV details the simulation settings for the evaluation which appears later in this section. Vertical handover with single-layer and scalable video are presented in that Section Finally, Section V rounds off the paper with some concluding remarks.

II. CONTEXT

A. Vertical handover while horizontal handover is concerned with migration between homogeneous networks, vertical handover is more intricate as it involves signalling between heterogeneous networks. Handovers are either soft, in which the previous connection is kept alive until the new connection is made, or hard, in which the previous connection is broken before the current one is made. Handovers can be: entirely controlled by a mobile device; be assisted by the mobile device though executed by the network, based on connection information at the mobile; or initiated by the network, without any action by the mobile device. Handover consists of: detection of a new network and selection of that network based on channel conditions; resource allocation as a new connection is established; and the update of routes and forwarding of data over the new connection.

Mobile WiMAX supports three handover mechanisms but only the mandatory hard handover at layer 2 can be accomplished with a single channel at any one time, thus reducing equipment cost and improving base station capacity. Hard handover employs a break-before-make procedure which reduces signalling. As is normal, a mobile station monitors signal strength from adjacent base stations, employing a hysteresis mechanism to avoid thrashing between base stations. The mobile station 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, whether for horizontal or vertical handover, will be subsumed in the emerging IEEE 802.11.21 [2] 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. 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.

In the successor to IEEE 802.16e, IEEE 802.16m [13], handover is also hard and network controlled. The role of the mobile station is confined to suggesting alternative base stations if a connection to the target base station fails. Reduced authentication procedures may occur after negotiation between the base station and the target access point. Consequently, in seamless handover, a mobile station may exchange data packets with the target access point before a network re-entry control transaction. In addition, entry-before-break allows connection with the original base station while at the same time negotiating with the target access point. At no point are there two data paths at the mobile station.

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 [14] and/or the mobility management [15]. 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 [16] at the client is possible. In [16], 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 [17] 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.

B. H.264 scalability the scheme has been applied to H.264 scalability. Fig. 2 shows an illustrative size four SVC GOP structure with hierarchical B-pictures (or P-pictures) between the key frames. In tests, a Group of Pictures (GOP) size of 16 was employed. Partitioning the enhancement-layer transform coefficients into multiple slices, MGS, increases the granularity of quality scalability and enables packet-based quality scalable coding. Unfortunately, this arrangement introduces inter-packet dependency which together with the predictive structure introduces complex dependencies. If not preserved, these dependencies can result in video quality reduction when the decoder drops packets that depend on non-received or erroneous ones.

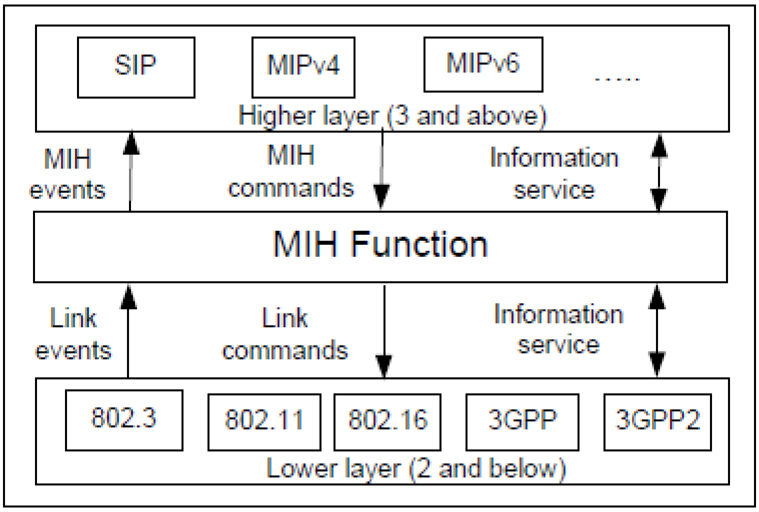

Fig. 1. Architecture of IEEE 802.21's MIH.

Fig. 3 is a representation of the processing involved in BVS in its scalable video variety, showing the NACK response of the receiver. The following describes the operation assuming downlink streaming from a vertical handover to a mobile station. At a mobile station a record is kept of packet sequence numbers available through the Real-Time Protocol (RTP) header. If an out-of-sequence packet arrives, a NACK may be transmitted to the base station in the next sub-frame for forwarding to the video serving office. The mobile station only transmits a NACK if this is the first time that particular packet has been lost. If it is the first time and a 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 in this case according to the type of SVC packet that has been lost, reflecting the importance of that packet to the reconstruction of the video. Thus, in the interest of reduced latency a mobile station only transmits a NACK if a prioritized packet has been lost for the first time.

The reorder buffer at the mobile station is a 'playout' buffer which may change the sending order of video data to fit the decode order. Upon receiving a NACK, 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. 3, is a holding buffer that retains sent packets in the case of the need for a retransmission. The variety of the scheme implemented for testing purposes negatively acknowledges the non-received or erroneous important packets shown shaded in Fig. 4. These are the base SNR (quality scalability) layer packets only in the selective variety, BVS-s. However, the scheme is capable of extension [18] to selected packets from the first MGS sublayer, assuming two SNR enhancement layers, as shown in a more general way in Fig. 3.

The scheme has also been applied with prioritization by picture type by way of comparison. In this case, in Fig. 3 the three outputs from the H.264/AVC (the single layer variety) become picture type I-, B-, and P-, with selective NACKs for I-pictures, i.e. BVS-I. In the single layer experiments reported in Section IV.B, a sending coding structure of IPBB…I was used with a 15 pictures refresh interval.

In the successor to IEEE 802.16e, IEEE 802.16m [13], handover is also hard and network controlled. The role of the mobile station is confined to suggesting alternative base stations if a connection to the target base station fails. Reduced authentication procedures may occur after negotiation between the base station and the target access point. Consequently, in seamless handover, a mobile station may exchange data packets with the target access point before a network re-entry control transaction. In addition, entry-before-break allows connection with the original base station while at the same time negotiating with the target access point. At no point are there two data paths at the mobile station.

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 [14] and/or the mobility management [15]. 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 [16] at the client is possible. In [16], 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 [17] 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.

B. H.264 scalability

The scheme has been applied to H.264 scalability. Fig. 2 shows an illustrative size four SVC GOP structure with hierarchical B-pictures (or P-pictures) between the key frames. In tests, a Group of Pictures (GOP) size of 16 was employed. Partitioning the enhancement-layer transform coefficients into multiple slices, MGS, increases the granularity of quality scalability and enables packet-based quality scalable coding. Unfortunately, this arrangement introduces inter-packet dependency which together with the predictive structure introduces complex dependencies. If not preserved, these dependencies can result in video quality reduction when the decoder drops packets that depend on non-received or erroneous ones.

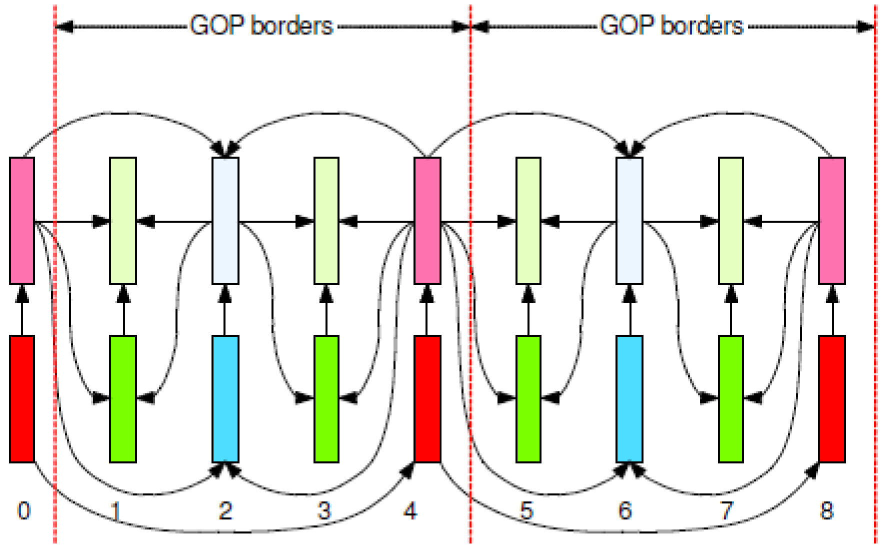

Fig. 2. Illustrative coding structure for GOP size of four with base and single enhancement layer, showing the prediction structure. Pictures appearing at 0, 4, and 8 are coded as key pictures

III. BVS ALGORITHM

Fig. 3 is a representation of the processing involved in BVS in its scalable video variety, showing the NACK response of the receiver. The following describes the operation assuming downlink streaming from a vertical handover to a mobile station. At a mobile station a record is kept of packet sequence numbers available through the Real-Time Protocol (RTP) header. If an out-of-sequence packet arrives, a NACK may be transmitted to the base station in the next sub-frame for forwarding to the video serving office. The mobile station only transmits a NACK if this is the first time that particular packet has been lost. If it is the first time and a 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 in this case according to the type of SVC packet that has been lost, reflecting the importance of that packet to the reconstruction of the video. Thus, in the interest of reduced latency a mobile station only transmits a NACK if a prioritized packet has been lost for the first time.

The reorder buffer at the mobile station is a 'playout' buffer which may change the sending order of video data to fit the decode order. Upon receiving a NACK, 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. 3, is a holding buffer that retains sent packets in the case of the need for a retransmission. The variety of the scheme implemented for testing purposes negatively acknowledges the non-received or erroneous important packets shown shaded in Fig. 4. These are the base SNR (quality scalability) layer packets only in the selective variety, BVS-s. However, the scheme is capable of extension [18] to selected packets from the first MGS sublayer, assuming two SNR enhancement layers, as shown in a more general way in Fig. 3. The scheme has also been applied with prioritization by picture type by way of comparison. In this case, in Fig. 3 the three outputs from the H.264/AVC (the single layer variety) become picture type I-, B-, and P-, with selective NACKs for I-pictures, i.e. BVS-I. In the single layer experiments reported in Section IV.B, a sending coding structure of IPBB…I was used with a 15 pictures refresh interval.

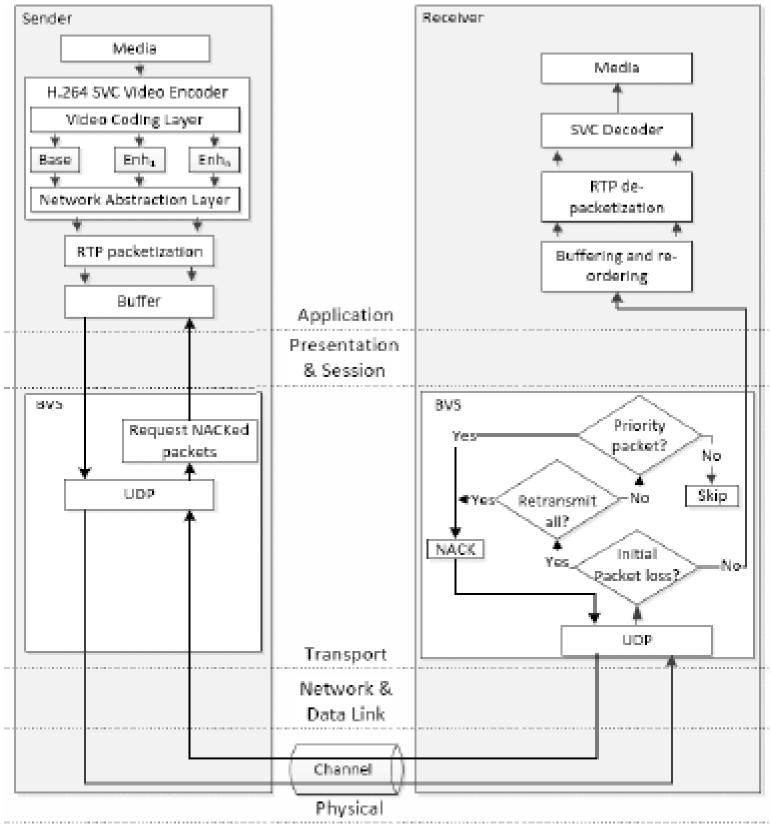

Fig. 3. Operation of the SVC version of the selective NACK scheme

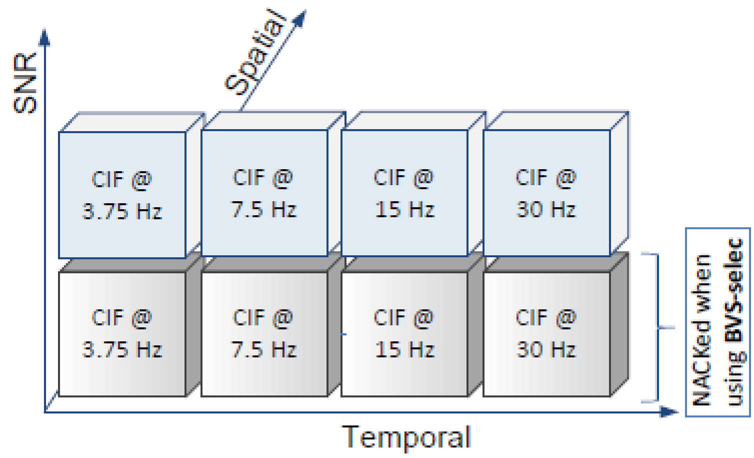

Fig. 4. H.264 SVC mapping to BVS-s. Packets belonging to darkly shaded layers are those that are negatively acknowledged.

IV. EVALUATION

A. Simulation scenario Fig. 5 shows a vertical handover scenario, in which streaming is from the video serving office across the metropolitan IP network showing intermediate routers (=R) along with link capacities and latencies on the streaming path simulated. Nodes marked A and B inject traffic into the bottleneck link between them, as sources of congestion. 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.

The mobile station (MS) traverses between a WiMAX base station (BS) (at 0.7 km distance) to an 802.11b access point (AP) (at 70 m distance) and returns back again. For ease of analysis, no other station is present in either the WiMAX or the 802.11b network (part b is used as a restriction of the ns-2 NIST 802.21 simulation module). Each data point was the mean PSNR of up to 20 independent runs of the well-known ns-2 simulator.

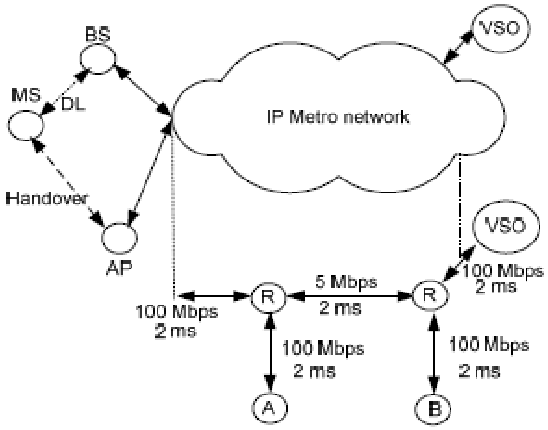

Fig. 5. Video streaming during vertical handover scenario

TABLE I. 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 II. 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

B. Wireless settings
To evaluate the proposal, transmission over WiMAX was carefully modelled. The PHYsical layer settings selected for WiMAX simulation are given in Table I. The antenna heights and transmit power levels are typical ones taken from the Standard [7]. The antenna is modelled 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 modelled. The IEEE 802.16 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 [7] 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 data rate of 10.67 Mbps over the downlink. Buffer sizes were set to 50 packets (a 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 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 II, while the operation of 802.11 is assumed to be well-known.

A 'bursty' Gilbert-Elliott channel model was applied to both WiMAX and IEEE 802.11 channels. In [19], it was shown that this model sufficiently approximates to Rayleigh fading, as occurs in urban settings during transmission from a base station to a mobile device. The Gilbert-Elliott model was selected, as it is the presence of burst errors [20] that mostly affects the quality of compressed video. *C. Single layer streaming results* Tests with the non-scalable H.264/AVC the video sequence used was 1035 frames of the *Paris* TV news clip, variable bitrate encoded as Common Intermediate Format (CIF)@30 Hz with IBBPBBP… (15 frame refresh interval). Temporal results in Fig. 3 isolate the effect of vertical handovers from WiMAX to 802.11b before an analysis of the complete scenario with two vertical handovers in Table III. Plots shown are raw UDP, TFRC [4], and two varieties of the lightweight NACK scheme: named broadband video streaming (BVS) when all initially lost packets are NACKed and named BVS-I when only I-picture bearing packets are NACKed. Because of the longer exposure of packets to channel conditions, video quality is generally worse during a WiMAX connection than a connection to 802.11.

BVS is clearly better able to cope with the transition in Fig. 3a, while BVS-I only results in a limited gain over UDP. Notice that during the stable period between the handovers, resending only I-frames (BVS-I) is sufficient to maintain quality, with reduced throughput and reconstruction delay at the decoder. However, during worse channel conditions, Fig. 3b, BVS-I also results in lower quality. This suggests that handover detection at the MS will make a hybrid BVS/BVS-I scheme effective. TFRC begins to recover quality after the handover but it does not compete with BVS. Furthermore, as Table III shows for the two handover scenario, the TFRC response to both congestion and channel errors is to increase the inter-packet gap such that the total sending period of the 34.5 s clip grows to an unacceptable level (about 8 s longer than the display time), though all transport methods suffer from interruptions (freeze frame effects). TFRC also suffers from poor wireless channel utilization.

TABLE III. H.264/AVC STREAMING OF PARIS: SUMMARY AT SPEED 2 MPS OF DIFFERENT TRANSPORT SCHEMES AFTER TWO VERTICAL HANDOVERS

	UDP	*TFRC*	*BVS*	*BVS-I*
Throughput (kbps)	838	687	851	855
Sending period (s)	35.48	43.08	35.79	35.58
Packet loss (%)	3.84	3.29	0.38	2.49
Packet jitter (s)	0.0077	0.0093	0.0075	0.0076
Mean packet end-to-end delay (s)	0.014	0.011	0.015	0.015
Max. packet end-to-end delay (s)	0.068	0.052	0.298	0.079
PSNR (dB)	31.29	32.31	37.43	33.01
Standard deviation (dB)	5.85	6.43	2.52	6.09
Max. interruption (s)	0.301	0.309	0.300	0.301

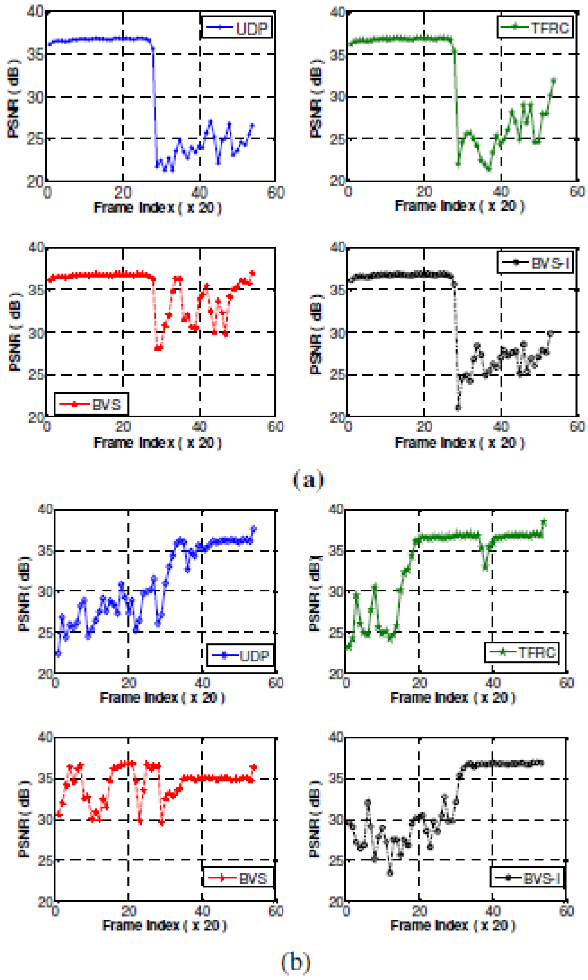

Fig. 6. Frame-by-frame video quality (a) during vertical handover from IEEE 802.11b to 802.16e (b) from IEEE 802.16e to 802.11b

D. Scalable streaming results

In this Section, streaming with the scalable version of H.264 is illustrated for the same two vertical handovers scenario as in Section IV.C. The H.264/SVC settings are those of Section II.B. Notice that no direct comparison between results in this Section and those of Section IV. B should be made as the video configurations have not been made equivalent. In Fig.7, one sees that transmitting NACKs for all lost SVC packets results in consistently better video quality. Selecting only base layer packets for retransmission is competitive in terms of video quality with the period during IEEE 802.11 transmission but is less so during WiMAX transmission. However, UDP transport is more likely to suffer deeper dips in received video quality than selective BVS. Table IV presents mean values. BVS-S presents lower throughput than unselective BVS. Therefore, it represents a possible choice as mean video quality over the sequence is still good and about 1 dB above that of direct UDP streaming. Besides, as Fig. 7 illustrated, video quality for IEEE 802.11 streaming is much closer to that of the unselective variety of BVS, again suggesting an adaptive scheme is attractive. It could turn out that H.264/SVC represents better quality than single layer streaming for reduced throughput. However, obviously the throughput for H.264/AVC streaming in Section IV.C is affected by the presence of I-pictures which because of the inefficiency of spatial coding are likely to have a heavy impact.

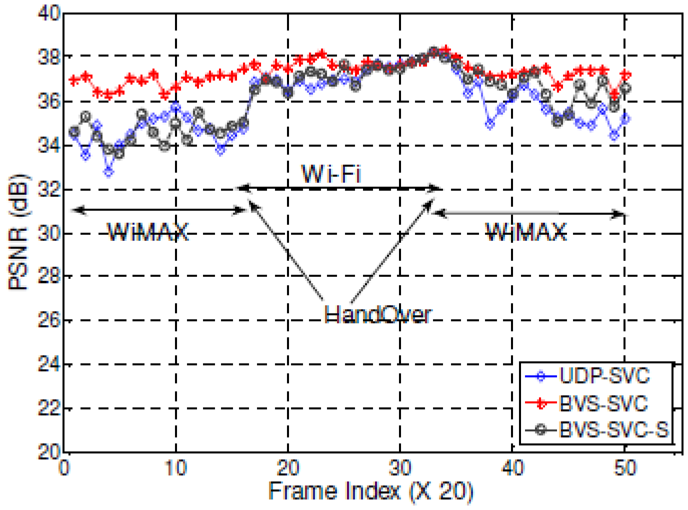

Fig. 7. Temporal GOP-by-GOP video quality two handover scenario for the *Paris* sequence

TABLE IV. H.264/SVC STREAMING OF PARIS: SUMMARY AT SPEED 2 MPS OF DIFFERENT TRANSPORT SCHEMES AFTER TWO VERTICAL HANDOVERS

	UDP	*BVS*	*BVS-S*
Throughput (kbps)	705	774	738
Packet jitter (s)	0.0076	0.0069	0.0072
Mean packet end-to-end delay (s)	0.014	0.019	0.018
PSNR (dB)	35.89	37.29	36.16
S.D. (dB)	1.99	1.02	1.90

throughput for H.264/AVC streaming in Section IV.C is affected by the presence of I-pictures which because of the inefficiency of spatial coding are likely to have a heavy impact.

V. CONCLUSION

This paper has presented a simple single NACK scheme designed for the low-latency paths that are likely to exist in IPTV content distribution networks. A key issue for such a transport system is what performance in terms of throughput (due to the capacity restrictions of broadband access systems) and video quality is possible during vertical handovers. It seems to the authors of this paper that unless handover is investigated then characterization of mobile IPTV is incomplete. Unfortunately industry standard TFRC fares badly in this situation as the time it takes to deliver a stream can far exceed the duration of the video sequence. Selective varieties of the proposed BVS scheme have been illustrated, as these allow trade-offs between lower through put with some reduction in video quality. They also represent a future adaptive scheme that chooses between unselective and selective BVS according to the underlying technology, which is easily determined at handover time.

REFERENCES

[1] S.Park, and S.-H. Jeong, “Mobile IPTV: Approaches, challenges, standards and QoS support,” *IEEE Internet Computing*, vol. 13, no 3, pp. 23–31, 2009.

[2] K. Taniuchi et al., “IEEE 802.21: Media Independent Handover: Features, applicability, and realization,” *IEEE Commun. Mag.,* vol. 47, no. 1, pp. 112-120, 2009.

[3] G N. Degrande, K. Laevens, and D. De Vleeschauwer, “Increasing the user perceived quality for IPTV services,” *IEEE Commun. Mag.*, vol. 46, no. 2, pp. 94-100, 2008.

[4] M. Handley, J. Pahdye, S. Floyd, and J. Widmer, “TCP-Friendly Rate Control (TFRC): Protocol Specification,” *Internet Engineering Task Force*, RFC 3448, 2003.

[5] C. Partridge, and R. Hinden, “Version 2 of the Reliable Data Protocol, ”*Internet Engineering Task Force*, RFC 1151, 1990.

[6] H. Schwarz, D. Marpe, and T. Wiegand, “Overview of the scalable video coding extension of the H.264/AVC standard,” *IEEE Trans. Circ. Syst. Video Technol*., vol. 17, no. 9, pp. 1103–1119, 2007.

[7] 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.

[8] O. Issa, W. Li, and H. Liu, “Performance evaluation of TV over broadband wireless access networks,” *IEEE Trans. Broadcasting*, vol. 56, no. 2, pp. 201-210, 2010.

[9] L.M. Lee, H.-J. Park, S.G. Choi, and J.K. Choi, “Adaptive hybrid transmission mechanism for on-demand mobile IPTV over WiMAX,” *IEEE Trans. Broadcasting*, vol. 55, no. 2, pp. 468-477, 2009.

[10] O.I. Hillested, A. Perkis, V. Genc, S. Murphy, and J. Murphy, “Adaptive H.264/MPEG-4 SVC video over IEEE 802.16 broadband wireless access networks*,”* in *Proc. Packet Video Workshop*, 2007.

[11] H.-H. Juan, H.-C. Huang, C.Y. Huang, and T. Chiang, “Cross-layer mobile wireless MAC designs for the H.264/AVC scalable video coding,” *Wireless Networks*, vol. 16, no. 1 pp. 113-123, 2010.

[12] J. Casampere, P. Sanshez, T. Villameriel, and J. Del Ser, “Performance evaluation of H.264/MPEG-4 Scalable Video Coding over IEEE 802.16e networks,” in *Proc. IEEE Int’l Symp. on Broadband Multimedia Systems and Broadcasting*, 2009.

[13] S. Ahmadi, *Mobile WiMAX: A Systems Approach to Understanding IEEE 802.16m Radio Access Technology*, Academic Press, Amsterdam, 2011.

[14] S.K. Lee, K. Sriram, K. Kim, Y.H. Kim, and N. Golmie, “Vertical handoff decision for providing optimized performance in heterogeneous networks,” *IEEE Trans. Veh. Technol.*, vol. 58, no. 2, pp. 865-881, 2009.

[15] K.-S. Kong, W. Lee, Y.-H. Han, and M.-K. Shin, “Handover latency analysis of a network-based localized mobility management protocol,” in *IEEE Int. Conf. on Commun*., 2008, pp. 5838-5843.

[16] D. Lee, J.W. Kim, and P. Sinha, “Handoff-aware adaptive media streaming in mobile IP networks,” in *Int. Conf. on Info. networking*, 2006.

[17] S. Ishihara, K. Koyama, G. Miyamoto, and M Kuroda, “Predictive rate control for realtime video streaming with network triggered handover,” in *IEEE Wireless Commun. and Networking Conf.*, 2005, pp. 1335- 1340.

[18] I. Ali, S. Al-Majeed, M. Fleury, and M. Ghanbari, “Semi-reliable transport protocol for IPTV over mobile WiMAX,” *EURCON: 8th Conf. on Telecomms.,* 2011.

[19] H. Wang and P. Chang, “On verifying the first-order Markovian assumption for a Rayleigh fading channel model,” *IEEE Trans. Veh. Technol.*, vol. 45, no. 2, pp. 353–357, 1996.

[20] Y.J. Liang, J.G. Apostolopoulos, and B. Girod, “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.