Vertical Handover Efficient Transport for Mobile IPTV

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Abstract. The success of IPTV suggests that an expansion to mobile devices is likely. A key 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. This paper proposes a lightweight form of IPTV transport based on negative acknowledgments. The performance of the scheme is analyzed in comparison to an industry standard congestion controller and baseline UDP transport. A selective acknowledgment variation of the scheme is also examined. The paper shows both proposed schemes result in better mean video quality (by as much as 4 dB) but that the non-selective scheme is better in the presence of vertical handovers. The paper presents a case study which emulates an IPTV streaming architecture with handovers between IEEE 802.16e (WiMAX) broadband wireless and an 802.11 network.

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

1 Introduction

Next Generation Mobile Networks (NGMN) [1] will consist of a number of overlapped heterogeneous networks, allowing the user to seamless pass between them through the process of vertical handover (VHO). VHO can be accomplished with the Media Independent Handover (MIH) part of the IEEE 802.21 standard [2] or through all-IP framing as integrated within the IP Multimedia Subsystem (IMS) initiative [3]. MIH is a looser way of approaching the goal of NGMN through optimized signaling between access points (APs) and/or base stations (BSs). This paper considers the MIH approach, which appears now to be gaining ground in the marketplace relative to IMS. The scenario considered is a mobile device crossing between an IEEE 802.16e (mobile WiMAX) [4] network to an IEEE 802.11 WLAN and *vice versa*, representing movement from indoors to outdoors reception. However, the general principles of efficient transport extend to transitions between other types of network. In this scenario, even though IEEE 802.21 attempts to harmonize signaling, the extent of signally differs between the two network types because of the relative complexity of WiMAX, which adds quality-of-service management to the underlying transmission system. The contribution of the paper is a transport method tuned to the needs of

IPTV (Internet Protocol TV) [5], which paves the way for new services such as time-shifted TV and video-on-demand. The proposed transport method improves upon standardized TCP-friendly Rate Control (TFRC) [6]. TFRC forms the basis of the video phone extension to mobile Google Talk [7], and as such improved performance with a different, more VHO sympathetic approach to video transport will have a practical significance.

There appear to be two ways to improve handover management for real-time services. The first is to reduce the latency of the network selection process [8] and/or the mobility management [9] [10]. The second is to improve performance at higher layers of the protocol stack. The two methods are not incompatible, though the concentration herein is to adapt the transport scheme to the needs of handover and video streaming. It is also possible to act at the application layer through increased protection against packet loss and delay [11] [12][13], the consequences of service interruption during handover. However, it appears that little attention has been given to transport layer improvements for VHO, which is the novelty of this paper's contribution. Because

The proposed scheme is a modification of UDP transport which is applied at the application layer of the protocol stack. Though UDP streaming has been used for broadband wireless access [14], UDP packet losses can seriously harm a compressed video stream. This is due to the predictive nature of video coding which operates through motion compensation and entropy coding. Bell Labs introduced a reliable form of UDP, R-UDP, see [15], and there is also an R-UDP protocol employed by Microsoft in their MediaRoom product for IPTV service delivery over multicast networks. In the present 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 contrasted with selective NACKs which are reserved for lost anchor frame packets from the video stream. Both are assessed for their performance during VHOs and compared to TFRC and raw 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 presents essential background knowledge to understand the results of the paper before Section 3 outlines the proposed scheme, Details of the simulation settings used to evaluate the proposal are given in Section 4. Section 5 presents those results, while Section 6 presents conclusions and future work, based on the findings of Section 45

2 Background

2.1 Handover procedures

While horizontal handover is concerned with migration between homogeneous networks, vertical handover is more intricate as it involves signaling 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 handoff mechanisms, but only the mandatory Hard Handover (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 subscriber station (SS) monitors signal strength from adjacent BSs, employing an hysteresis mechanism to avoid thrashing between BS. The SS 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 802.11.21 [2].

2.2 IPTV video streaming

A typical delivery chain [5], refer to Fig. 1, is from a Video Hub Office (VHO) (not to be confused with VHO for vertical handover), perhaps receiving Digital Video Broadcasting (DVB) from a satellite, then over an IP-framing based network. A Video Serving Office lies at the edge of the IP network to distribute video to different types of access network. In the Figure, a WiMAX BS and an IEEE 802.11 Access Point (AP) act as potential delivery routes for a multi-homed mobile SS device. Notice that to facilitate fast channel swapping [5] it is likely that through intelligent management, content will be placed relatively close to the receiver. This implies that a feedback channel for acknowledgments will not experience long delays, which is one reason why reinforcing UDP with NACKs seems appropriate for IPTV.

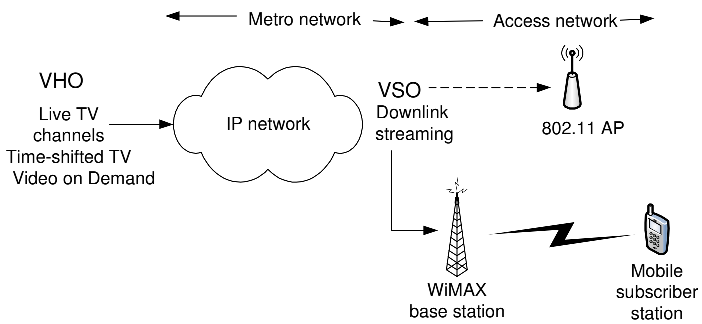


Fig. 1. IPTV video content delivery architecture

As a test, the *Paris* sequence H.264/AVC (Advanced Video Coding) codec [16] 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 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. Consequently, the risk of loss of synchronization between encoder and decoder was reduced. *Paris* consists of two figures seated round a table in a TV studio setting, with high spatial-coding complexity and moderate motion. Quality-of-experience tests show [17] 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. In the prioritized version of 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. The presence of I-pictures rather than gradual decoder refresh enables channel swapping for IPTV and in [5] a means of accelerating critical channel zapping time from network caches was demonstrated. This further encourages the use of the proposed NACK scheme.

2.3 IPTV transport

As mentioned in Section 1, 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 reduced latencies are likely to occur. In fact, Agilent recommend [18] a maximum cumulative delay factor of between 9 and 50 ms for IPTV delivery over a network. Raw UDP has been used for IPTV transport over IEEE 802.16e systems [14] but the principle interest in this paper, as the proposal improves upon UDP, is to use UDP as a baseline. The alternative selected at the industry level is a standards-based form of transport. TFRC [6] or its variant Datagram Congestion Control Protocol (DCCP) [19], which adds connection set-up, have been used directly [20] or in cross-layer form [21] for video streaming in research work on wireless networks.

In the simulations, the inter-packet sending time gap was varied according to the TFRC equation [6]. As described in [6], TFRC is a receiver-based system in which the packet loss rate is found at the receiver and fed-back to the sender in acknowledgment messages (through TCP in the simulations). The sender calculates the round-trip time from the acknowledgment messages and updates the packet sending rate. A throughput equation models TCP New Reno to find the sending rate:

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

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 are dependent on these two variables in the denominator.

3 NACK-based Scheme

Fig. 2 is a general representation of the processing involved in the scheme, which for convenience of reference, as previously mentioned, 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 a mobile SS 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 SS only

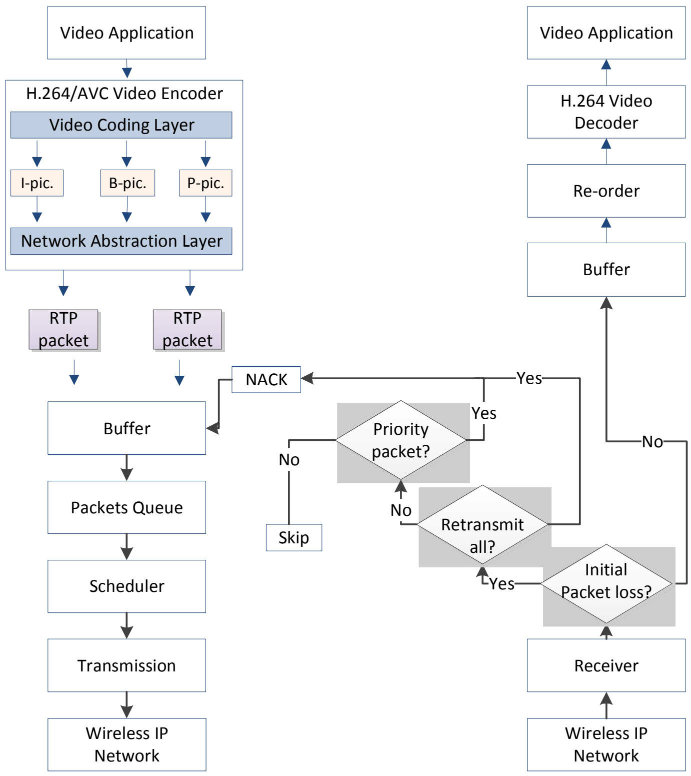


Fig. 2. Operation of BVS NACK enhancement to UDP

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.

4 Simulation Scenario

Simulation studies have been conducted for IEEE 802.16e (mobile WiMAX) with vertical handover to an IEEE 802.11 WLAN. To establish the behavior of VHO under WiMAX the well-known ns-2 simulator augmented with a module from the Chang Gung University, Taiwan [22] that has proved an effective way of modeling IEEE 802.16e's behavior. IEEE 802.21 was modelled with the NIST handover module for ns-2 [23] which is tied to the IEEE 802.11b model built into ns-2 operating at 11 Mbps (hence the use of this version of IEEE 802.11 in simulations). Simulation settings for 802.11b are given in Table 1. 25 runs per data point were averaged (arithmetic mean) and the simulator was first allowed to reach steady state before commencing testing.

A Gilbert-Elliott channel model [24] modeled 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. We are aware that the channel could also be modelled more closely for propagation effects but it is the presence of burst errors [25] that mostly effect's compressed vidoe quality. In fact, after testing for the impact of burst errors during VHO, future work will consist of assessment under the channel conditions specified in the Standard [4], i.e. propagation loss modelled by the COST 231 Walfisch-Ikegami model with Jake's model for fast fading. Slow fading as a result of changes in the wireless envirnoment such as increased reflection from buildings or interior walls as result of motion can be provided for by a shadowing model, such as log-normal.

Table 1. 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

To evaluate the proposal, transmission over WiMAX was carefully modeled. The PHYsical layer settings selected for WiMAX simulation are given in Table 2. The antenna heights and transmit power levels are typical ones taken from the Standard [4]. 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. 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 simplification of the Standard. The data rate results from the use of one of the mandatory coding modes [4] 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 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 single MAC Service Data Unit with a MAC Protocol Data Unit [26]). This buffer size was selected as it seems 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.

Fig. 2 shows a vertical handover scenario, in which streaming is from the VSO 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

Table 2. 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 SS	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

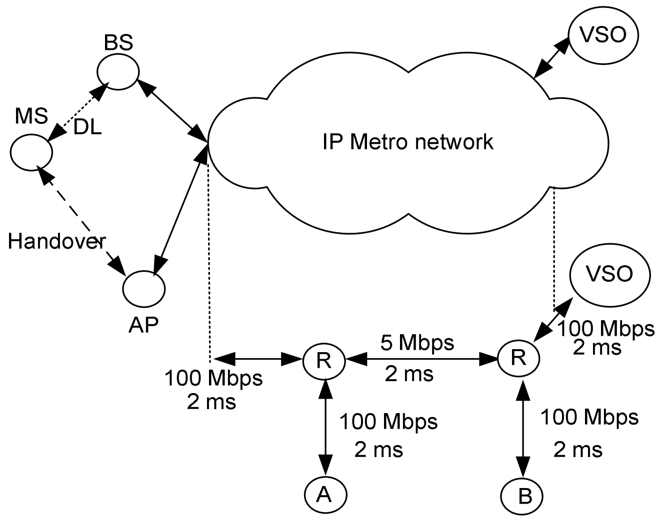


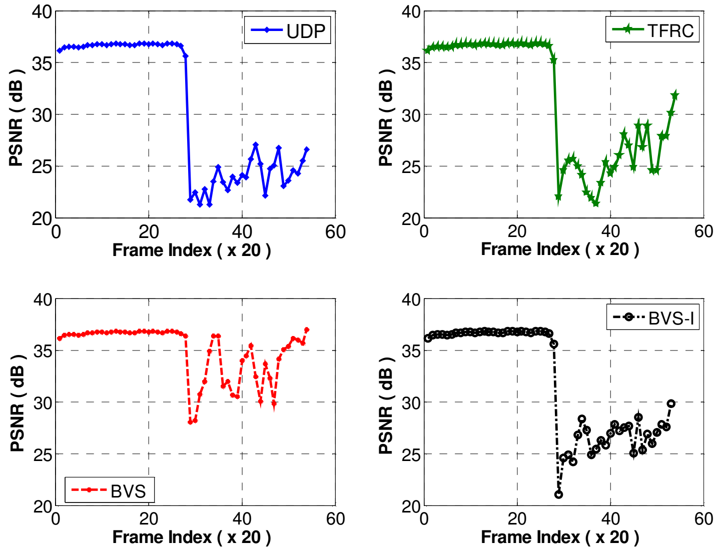
Fig. 3. Video streaming during the vertical handover scenario, MS = mobile subscriber station

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 effect of traffic sourced from B is to cause some congestion to the returning TFRC acknowledgement.

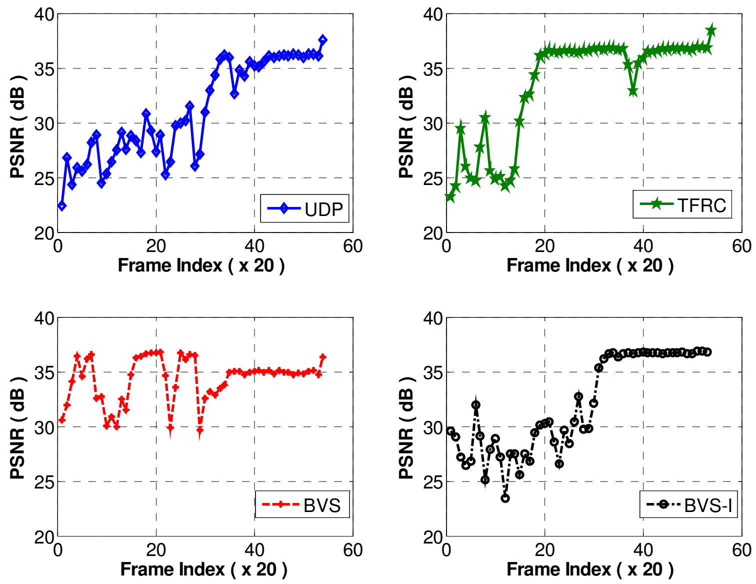
5 Evaluation

Frame-by-frame results in Fig. 3 isolate the effect on video quality (PSNR) of vertical handovers from IEEE 802.16e to 802.11b and *vice versa*. When the mobile SS loses connection to the IEEE 802.16e BS it is at 0.93 km distance from the BS, whereas when it connects to the IEEE 802.11b AP it is at 70 m distance from the AP. For ease of analysis, no other station is present in either the IEEE 802.16e or the 802.11b network. By default, the speed of movement of the mobile SS was 1 m/s (2.2 mph), i.e. as appropriate for somebody strolling between the BS and AP while using a mobile device. Notice that in a short study of the impact of different speeds [27] when going between these networks it was found that IEEE 802.18e's performance actually improved at higher speeds (up to 10 m/s) because of IEEE 802.16e's superior mobility management.

Plots shown are raw UDP, TFRC, and the two varieties of the lightweight NACK scheme: that is BVS when all initially lost packets are NACKed and 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 an IEEE 802.16e connection than a connection to 802.11. BVS is clearly better able to cope



(a)



(b)

Fig. 4. Frame-by-frame video quality (a) during vertical handover from IEEE 802.11b to 802.16e (b) from IEEE 802.16e to 802.11b

with the transition in Fig. 4a, while BVS-I only results in a limited gain over UDP. TFRC begins to recover quality after the handover but it does not compete with BVS. Notice that during the stable period between the handovers, re-sending 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. 4b, BVS-I also results in lower quality.

In the extended test, the mobile SS journeys from the BS to the AP and back again. An approximately equal time was spent under IEEE 802.16e and IEEE 802.11b streaming. As Table 3 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, as its throughput declines. Both the proposed schemes result in reduced packet loss, i.e. the packets lost even after a single attempt at retransmission. A consequence of retransmission is greater end-to-end packet delay, when delay is the aggregate of an initial transmission and any delay from resending a packet after a NACK. However, the delay in both cases is less than the 50 ms noted in Section 2.3. The main deficiency of BVS appears to be the delay that occurs during handovers, as the maximum end-to-end delay is high. However, it is actually the maximum period of interruption caused to frame display rather than the impact of delay from individual packets making up the frame that is significant. In this respect all systems appear to behave similarly.

Table 3. Summary at speed 2 mps of different transport schemes after two vertical handovers — the first from IEEE 802.16e to 802.11b and the second from IEEE 802.11b to IEEE 802.16e with data the mean of 25 simulation runs, PB = 0.15, speed 2 m/s

	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	0.007 5	0.0076
Mean packet end-to-end delay (s)	0.0141	0.0106	0.015 0	0.0149
Max. packet end-to-end delay (s)	0.0680	0.0520	0.298 0	0.0790
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

An important point to note is that packet loss during handover is heavily dependent on speed, even at slow walking speeds. The packet loss reflects itself in the overall video quality which is shown for different speeds in Fig. 5. Notice that 2 mps or 4.5 mph is already fasting than walking speed of about 3.3 mph. Likewise, see Fig. 6, variation in channel condition can notably affect video quality, especially when

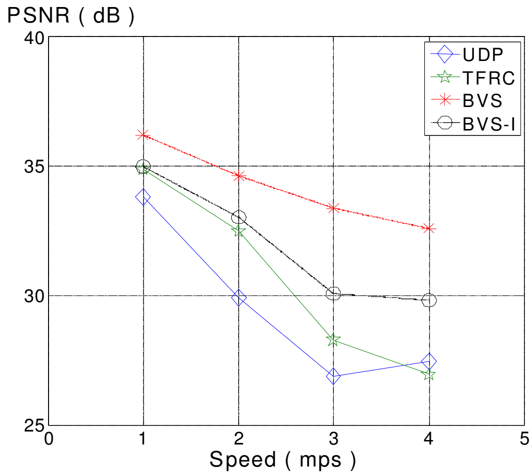


Fig. 5. Variation of video quality for the different schemes with differing speed

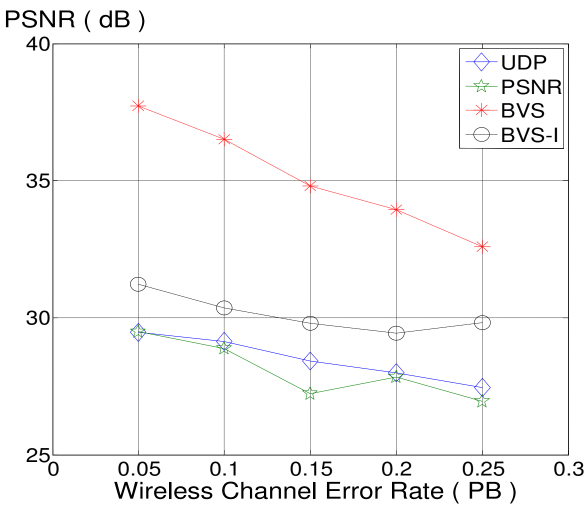


Fig. 6 Variation of video quality for the schemes with differing channel quality

communication to IEEE 802.11e because of the longer transmission times involved (as previously mentioned). However, though BVS suffers from poor channel conditions its video quality overall remains superior.

6 Conclusion

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. 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. An interesting point is that 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. An interesting observation from the simulations is that the selective NACK version of BVS tested, i.e. BVS-I, did not perform as well as BVS itself if there were vertical handovers. However, during streaming to the IEEE 802.11 AP good video quality resulted with less mean packet delay and reduced throughput. This suggests that handover detection at the mobile SS will make a hybrid BVS/BVS-I scheme effective. Future work will investigate the value of such a combined scheme.

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