Channel Adaptive Video Stream Switching for Broadband Wireless Links

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This paper exploits H.264/Advanced Video Codec smooth stream switching to achieve robust video delivery across an IEEE 802.16 (WiMAX) broadband wireless link. As the wireless channel conditions vary over time, dynamic selection of bitrate and corresponding video quality will reduce the risk of harmful packet loss. Choice of stream-switching with Secondary SP-frames or SI-frames is investigated relative to the selection of quantization parameter (QP) values. To control the switching points at the WiMAX server, a feedback mechanism that monitors packet loss is applied. An adaptive ARQ scheme is alternatively exploited to protect switching frames against packet loss. The broadband wireless streaming system gives more protection to higher quality video, reduces delay and packet loss, and improves received video quality. In particular, for the QP values selected, results show that increased quality primary switching frames with SI-frames bring a significant gain in video quality over the other switching schemes with secondary SP-frames, which in turn show an improvement over 'no switching' for a typical WiMAX channel with burst errors. Link delay is also reduced.

Keywords *broadband wireless, SP and SI switching frames, video coding, video streaming, WiMAX*

1 Introduction

The aim of this paper is to enhance unicast video streaming over a WiMAX broadband wireless link [1] through the application of encoded bitstream switching [2]. Broadband wireless is an alternative to access networks such as cable and Digital Subscriber Line, while WiMAX can also support mobile users. IEEE 802.16 (known as WiMAX) [3] allows rapid deployment of video services in rural areas in the world unlikely to benefit from extensions to Third Generation (3G) cellular systems and is also already widely deployed in the U.S. In areas where 3G is already present, Long Term Evolution (LTE) is likely to have similar broadband facilities to WiMAX for mobile users. Therefore, the results in this paper for unicast streaming are also broadly applicable to LTE, just as the proposed WiMAX multimedia broadcast and multicast service (MBMS) is similar for WiMAX and LTE [4].

There has been comparatively limited research published on the subject of unicast video streaming over WiMAX, despite (or possibly because of) the commercial attraction of what would be a value-added service. In [5], the Scalable Video Coding (SVC) extension to the H.264 codec was used to improve received video quality over a WiMAX network. Multiple connections operated to send different quality layers of a scalable video stream. However, the scheme was only tested with two layers and video quality testing does not appear to have accounted for lost packets, despite the interdependencies between the SVC layers that make it difficult to decode if the lowest temporal layer' packets are lost. IPTV is an attractive application of WiMAX [6]. Proposed solutions have often been at the physical layer rather than the application layer. In [6], two-level superposition multicast allowed some data to be received by all subscriber stations (SSs) and additional video data to be received by SSs experiencing better channel conditions, again a layered video scheme but one requiring adaptive modulation. Similarly in [7], unequal error protection through adaptive modulation was proposed. In [8], Multiple Description Coding in combination with SVC was suggested but this involves development of a specialist codec rather than utilizing an existing codec, as in our work. Multi-connection WiMAX video streaming [9] is intended for seamless congestion control over a concatenated network (one comprising Internet and broadband wireless link) rather than optimization of broadband video streaming. The attraction of IPTV Video-on-Demand was identified in [10] but dimensioning of the WiMAX radio system was the main concern and the wrork reported in [11] is in a similar vein.

To provide smooth switching between streams we employ switching frames, namely H.264/Advanced Video Codec (AVC)'s SP/SI-frames [2]. Because a feedback channel is required, this scheme is suited to a video-on-demand service and not to MBMS. The delay should also be limited across the feedback channel. Work on cellular wireless applications of stream switching, such as [12, 13] detailed in Section 2, concentrated on switching as a way of varying the bitrate to avoid congestion, while our paper proposes switching for an entirely different purpose, namely adjustment of the bit-rate to reduce the impact of adverse channel conditions leading to packet loss. To gain better received video quality acknowledgment messages are sent to identify when switching is necessary. The Time Division Duplex (TDD) frame structure of IEEE 802.16 provides a feedback channel for the streaming client. The streaming client records packet losses and sends this information back to the server. Each short acknowledgment packet is sent prior to the server sending a switching frame. Switching is performed both from high-to-low and from low-to-high quality video streams, depending on packet loss statistics. Trade-offs are investigated in the choice of predictively-coded or spatially-coded secondary switching frames (SP- or SI- frames). An extension to the switching scheme is adaptive monitoring of round trip times to determine whether an Automatic Repeat request (ARQ) for switching frames has arrived. If the ARQ acknowledgment does not arrive in time then a switching frame is retransmitted. When what we call 'adaptive ARQ' is deployed, there is a tendency to increase the average end-to-end delay. However, the delay is still tolerable for non-interactive applications and without the receipt of a switching frame, encoder-decoder mismatch occurs, leading to error propagation over time.

The main value of this work is as a demonstration of the value of switching frames for broadband wireless, which includes the possibility of 'bursty' channel errors. In particular, our results show a significant gain in video quality from employing SI-frames compared to the other schemes. This advantage may well outweigh the advantages of switching with secondary SP-frames. If switching occurs occasionally it is preferable to be able to prevent error propagation with SI-frames. SI-frames are able to reset the video stream so that errors do not propagate, even if the SI-frames are of moderate quality. The increase in throughput from employing in compensation higher-quality primary switching frames in the SI-frame switched streams is moderate.

Stream switching presents a natural progression from commercial simulcast streaming systems and allows these schemes to be adapted for use over a wireless channel. Compared to the complexity of SVC, their computational complexity is small. Compared to the use of intra-coded frames in simulcast, more potential switching points become available with much improved video quality if SI-frames are used. In the paper, we analyze the overhead from storing switching frames, which for secondary SP-frames is about 41% and for SI-frames about 30%. The difference is explained by the use of higher-quality secondary SP-frames, mandated by the need to optimize the overall rate distortion of the switching scheme.

The remainder of this paper is organized as follows. Section 2 examines related work involving stream switching in cellular wireless systems and commercial simulcast systems. Section 3 describes the stream-switching system and contains a detailed analysis of an H.264 stream switching encoder. Section 4 gives the WiMAX background and details of modeling the wireless channel. Section 4 also contains the feedback schemes. Section 5 evaluates the switched streaming schemes in terms of their network performance and the resulting video quality, indicating trade-offs between the schemes. Section 5 includes an analysis of the overhead in sending switching frames and the static storage overhead. Finally, Section 6 draws some conclusions.

2. Related work

Adaptive streaming has been of interest to wireless multimedia researchers. The adaptive element of the packet-switched streaming (PSS) system (audio and video) for the Third Generation Partnership Project (3GPP) is described in [12]. Adaptation in 3GPP PSS aims to change the throughput according to: 1) changing capacity across links formed by different wireless technologies, e.g. from WCDMA to GPRS (intersystem handover); 2) changing cell user population; and 3) lowered bandwidth as a result of channel conditions. Therefore, this generic system using a feedback channel that responds indirectly to channel conditions by reducing the bitrate. The main intention of adaptation in that scheme [12] is to avoid receiver device buffer underflow or overflow, as well as to avoid exceeding the link bitrate. As such the system in our paper is complementary to 3GPP PSS, as it could provide smooth bitrate transitions, without the need for extensive buffering. Increasing buffer sizes in mobile devices has a significant impact on active and passive energy consumption, leading to a reduced battery life and the need to recharge more frequently.

Work in [13] report refinements to the generic 3GPP PSS system. A frame type prioritization scheme is introduced, so that, if the bandwidth drops or there is an outage, the more important frames for decoding are available first. It is also possible to drop non-referenced predictively-coded (P)-frames in H.264/AVC if the available bandwidth requires this as a form of temporal scalability. The research in [11] adds H.264/AVC switching frames to the temporal scalability scheme for GPRS and EGPRS cellular wireless systems. In addition, H.264 generalized bi-predictive (B)-frames are added as a further temporal scalability feature, though, in fact, B-frames are not supported in 3GPP's adoption of the H.264 Baseline Profile. The scheme [13] relies on comprehensive feedback information, rather than the minimal feedback messages required by our scheme.

Research in [14] concerned itself with how to provide a seamless multicast streaming service in the event of a vertical handoff from one wireless technology to another. Multi-homed devices allow continuous wireless coverage when a currently available wireless network becomes unavailable. The paper [14] contains a detailed consideration of the signaling process that is required so that when a vertical handoff is anticipated streams of packets are duplicated at each network interface. As such this type of adaptive streaming would also complement our scheme.

In commercial streaming systems simulcast is favored for its relative simplicity. In simulcast, multiple streams are stored (or encoded online) at different rates and selected according to network conditions. For example, in Windows Media [15], intended for the wired Internet, the receiver detects the onset of congestion by monitoring its input buffer's occupation and packet loss. However, simply switching between streams may result in prediction mismatch, bearing in mind that to reduce temporal redundancy video compression relies on motion prediction from previous frames. 'SureStream' of RealNetworks and 'Intelligent Streaming' from Windows Media insert I-frames for switching. (Intra-coded I-frames are locally coded with purely spatial encoding.) These schemes are mainly intended for the wired Internet in which traffic congestion rather than 'lossy' channels are the principal threat. Using I-frames (key frames) has some drawbacks: the compression efficiency of I-frames is low, as their compressed size is approximately 5-10 times higher than that of P-frames. Therefore, inserting many switching points in a bit stream results in a significant increase in bandwidth. In fact, I-frames are normally placed every 30 s, which is actually a significant time to wait while visual artifacts caused by encoder decoder drift occur. In the case that switching does not occur, bearing this extra overhead and delay (in forming the larger I-frames in the output buffer and transmitting them) may be pointless. Switching frames can be used for stream switching but, because they rely on predictive coding, they can be coded more efficiently. Consequently, for the same bit-rate as a simulcast stream with key frames, more switching points are possible.

3. Smooth video switching

3.1 Overall system

Figure 1 is a diagram of the proposed system. At the streaming server, a set of pre-encoded videos are stored at various bitrates. After adaptively-switched selection of the stream, generic physical-layer Forward Error Control (FEC) is applied before IP packetization and transmission over the wireless packet loss channel. This paper assumes a WiMAX link with an Internet Protocol (IP) stack employed for messaging. In order to avoid delay, it is normal to transport video streams through unreliable User Datagram Protocol (UDP), without packet retransmission. FEC channel decoding is then applied at the receiver. IP packets are aggregated or fragmented into their coding units before error detection of remaining errors. At this point in time, packets are declared lost.

If a packet is lost then the process of error concealment (Error Con. in Fig. 1) takes place at the decoder, using a previously decoded frame. (If no error concealment is possible because of multiple lost frames, previous frame replacement (PFR) takes place.) Otherwise, the usual decoding processes for intra-coded frames of variable length decoding, dequantisation (Deq.), and inverse transformation, are applied to the residual predication data. If a frame is inter-coded (non-local encoding) then motion compensation prediction (MCP) from a previously decoded frame takes place using motion vectors to match the decoded residual data with a previously decoded frame. Notice that some macroblocks within a predictively-coded frame are in fact intra-coded, that is they are not predicted temporally but are predicted from spatially adjacent neighboring macroblocks[1]. However, the main point as far as this paper is concerned is that upon detection of packet loss a feedback route exists to the server in the event of a stream switch being required. An application-layer ACK from the receiving device to the server is sent utilizing WiMAX's TDD frame structure.

H.264/AVC consists of a Video Coding Layer (VCL) and a Network Abstraction Layer (NAL). The VCL is responsible for generating the source encoded bit streams, while the NAL adapts the bit stream for network transport. Sometimes there is not a one-to-one mapping between Real-Time Protocol (RTP)-headed packets and NAL unit, which is why aggregation and segmentation is mentioned in Fig. 1. Feedback commands might be encapsulated in Real Time Control Protocol (RTCP) packets running on top of UDP [16]. Though H.264 supports a number of error resilience techniques, switching frames are currently supported by the Extended Profile and not the low complexity Baseline Profile. However, as hardware implementations of H.264/AVC were quickly implemented [17], the authors expect that low-power Application Specific Integrated Circuit (ASIC) decoders supporting switching frames will be implemented should the need arise.

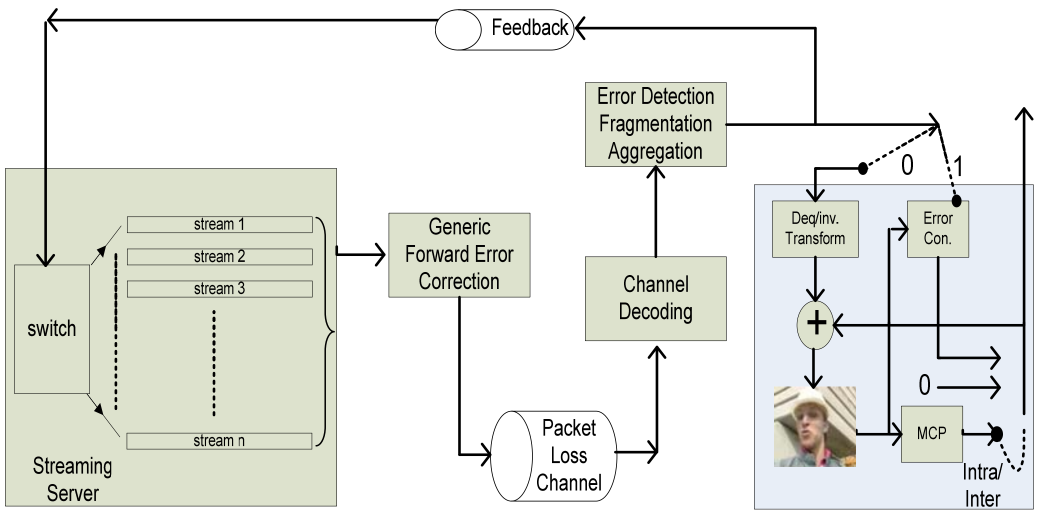

Figure 1. Switched video stream system for wireless channels

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[1] Some macroblocks are also not predicted but simply skipped, as they have not changed significantly since the last frame.

3.2 Switching frames

In H.264/AVC, new types of frame, namely Switching Predictive/Intra frames (SP/SI-frame), have been defined. SP- and SI-frames were proposed by Karczewiz and Kurceren [2]. This feature is introduced in the Extended Profile of the H.264 codec. These frames were designed to support different applications [2] such as stream-switching between bitstreams at different coding rates as herein, random access, VCR facilities such as fast forwarding, error resiliency and error recovery, as well as splicing between different sequences and switching between bitstreams coded at different frame rates. Notice that this paper refers to Variable Bit Rate (VBR) switching. Therefore, for any one stream I- and P-frames are encoded with the same Quantization Parameter (QP). The concept of bitstream switching in which the streams have a Constant Bitrate (CBR) occurs in [18]. The aim of the SP/SI frames [2] is to "enable reconstruction of identical frames using different reference frames". Thus, in stream switching between bitstreams at encoded at different rates, the reconstructed frame after the switching frame is the same as if it was reconstructed in the normal manner without switching.

Earlier research most probably contributed to the development of SP/SI frames. In [19], only H.263 P-frames were employed (except for an initial I-frame) in each of two streams. In going from the lower bitrate (Quarter Common Intermediate Format (QCIF)@15 Hz or frame/s) to the higher bitrate (CIF@30 Hz) at a switching point, a P-frame is re-sampled and the difference between corresponding P-frames in the two streams is spatially-encoded to form a Switch or S-frame. However, the temporary increase in bitrate was six times as much as the QCIF rate from using S-frames in this way because of the fine QP required. Compared to CIF I-frames, the temporary rate was double that of I-frames. Therefore, S-frames are suitable only if there is a way of accommodating a sudden increase in datarate. To reduce the overhead, the S-frames were predictively-encoded in [20] with a fine or lower-valued QP. Nevertheless, there still remains drift error between encoders and decoder, because of the different P-frame references. Therefore, the performance of the scheme in [20], which was applied to a digital cordless (DECT) phone wireless channel, can be improved with SSP-frames. In [20], selective ARQ was employed rather than the scheme in this paper.

In [2] it is shown that it is possible to support VCR functionality if switching frames are dynamically produced. However, a static alternative is to also store a periodic set of coarsely quantized I-frames that are only transmitted when VCR functions are required.

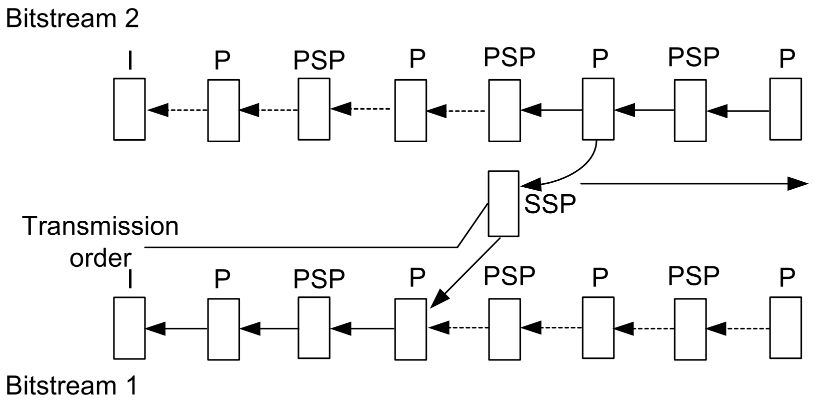
There are two types of SP-frame, namely primary and secondary SP-frames. In this paper, Primary SP frames are generally denoted as 'PSP-frames' and Secondary SP frames are generally denoted as 'SSP-frames'. The intra-coded version of the SSP frame will be called an SI-frame, while 'switching frames' will signify the overall concept. An SI-frame does not reference a previous frame as it does not use predictive coding, while an SSP-frame does require a reference frame. Therefore, in the event of a feedback message request, a robust option is to use an SI-frame to switch streams to prevent any possibility of error drift. If the quantization parameter of the SI-frame is appropriately set (refer to Section 3.3), the fact of using intra-coding for the SI-frame does not result in reduced coding efficiency at the point of switching.

PSP-frames are inserted at various pre-determined and matching periodic locations in the frame sequence in both streams. SSP- or SI-frames (or both) are created at the same periodic locations as the PSP-frames ready to be used should the need arise. The storage overhead from pre-coding SSP/SI-frames is analyzed in Section 5.4. If SSP/SI-frames were to be created dynamically this would cause delay which would restrict their application to interactive applications (probably in the CBR variety of stream switching [18]).

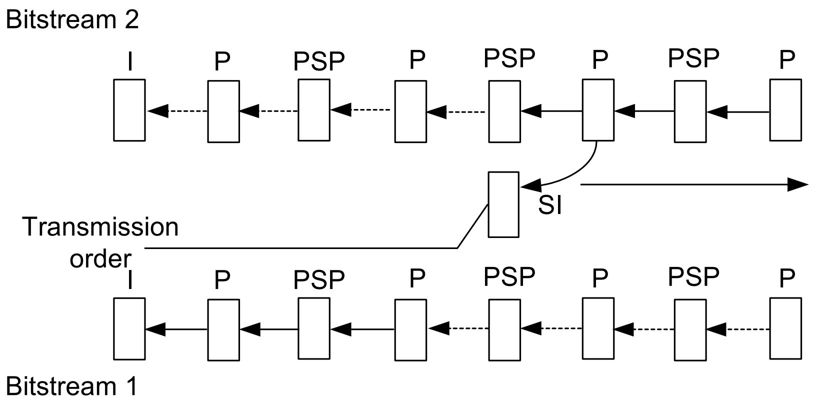
In the event of one or more packet losses in a normal video stream without switching, the loss of synchronization normally results in drift error until the next synchronization point, which is normally the next intra-coded I-frame. Notice though that a form of distributed intra-coding [21], Gradual Decoder Refresh, is also available in Baseline Profile streams without I-frames (except an initial I-frame). However, if a feedback channel is available the decoder can signal the presence of error to the server, and an SSP- or SI-frame can be transmitted without the need for I-frame synchronization. The main advantage of switching is that periodic PSP-frames replace I–frames. This is because PSP-frames exploit temporal redundancy with the result that they can be compressed more efficiently than I-frames.

In Fig. 2, to enable drift-free switching, the streaming server stores the same sequences encoded at different datarates and, therefore, different qualities resulting from different quantization parameters (QPs). As mentioned, these bitstreams are populated with PSP-frames at the locations where switching is allowed, as shown in Fig. 2. Notice though that Fig. 2 is an illustrative example only and the periodicity of PSP frames can vary just as the I-frames they replace can. As mentioned, if switching becomes necessary, an SSP- or SI-frame is transmitted instead of the PSP frame. The arrowed line in Fig. 2 indicates that transmission starts with bitstream one and that all frames before the second PSP-frame of bitstream one are transmitted, followed by an SI- or SSP-frame. From then onwards the rest of the transmitted frames are from bitstream two, omitting the PSP-frame in bitstream two, as the SSP-frame has substituted for it. Therefore, in Fig. 2 the bitstream two data from the start to the second PSP frame is never transmitted.

Of course, multiple switching sequences could take place in practice, switching back and forth between several streams. For simplicity Fig. 2 shows only two streams. To achieve switching in both directions switching frames for both switching directions need to be generated beforehand, unless dynamic switching is implemented. The SSP-frame shown in Fig. 2a is the secondary representation of the corresponding PSP-frame in bitstream two. The corresponding PSP-frame is the one that would have been generated at the same time in bitstream 2. However, the SSP-frame is predicted from a bitstream one P-frame rather than a bitstream two P-frame. It is also generated as part of the same encoding process for the corresponding PSP-frame in bitstream one. Similarly, the SI-frame in Fig. 2b is also generated in this way (as part of the encoding cycle) but is not predictively-coded from a prior P-frame, with the result that error propagation does not occur.



(a)



(b)

Figure 2. Switching between streams using (a) SSP-frames and (b) SI-frames, showing transmission order and predictive dependencies between successive frames

3.3 Encoding switching frames

To create a suitable encoder a structure is used that will not only output either P-frames or I-frames but will also output the required switching frames. That is in switching frame mode it will output a PSP-frame and either a SSP-frame or an SI-frame. Therefore, another gain from stream switching is that a switching frame encoder is essentially a traditional encoder with an additional coding stage. This is achieved by including a P-frame encoder and an I-frame encoder within the same coding loop. For P-frames, only the MCP residual is encoded at the same or a coarser QP than for a PSP-frame. For an I-frame, the residual is not encoded but the frame itself is spatially encoded. For, a PSP-frame the residual is encoded at a finer QP than for a P-frame. Call the finer QP, QPSP. Additionally, the PSP-frame is reconstructed and intra-coded at a different QP for an SI- or SSP-frame. Call this secondary QP QPSP2. It is this second quantization during intra-coding that makes the switching frame compatible with the P-frame in the stream to which the switch is made.

Direct entropy coding of secondary quantized coefficients produces SI-frames as a by-product of the PSP encoder. For constructing SSP-frames, the motion compensated prediction residual after quantization is entropy coded [22]. In both cases variable length coding (VLC) is applied, a lossless process. However, an SSP-frame is dependent on the arrival of a prior P-frame, whereas an SI-frame is not. Similarly, if a switching frame fails to arrive then drift error will occur in the new sequence and it becomes worthwhile to retransmit the switching frame, at a cost in delay.

The original encoder design in [2] included a stage in which the predicted P-frame was further quantized and dequantized using QPSP2. This was found to normally introduce unnecessary degradation as a result of a further processing stage [23]. Therefore, the encoder loop was adjusted [23] and used in [18, 22] and this work. The adjustment is simply to allow the extra quantization and requantization to be turned off if the prior frame is not a PSP-frame.

In Fig. 3 for an H.264/AVC encoder, a PSP-frame is generated in the normal way for predictively-encoded frames by quantizing the residual. As a P-frame normally precedes a PSP-frame, the residual is taken from the bottom route in Fig. 3 without quantization and requantization by QPSP2. The upper route exists when a P-frame is formed if that P-frame is preceded by a PSP-frame (or the upper route can be neglected altogether as the distortion introduced in negligible [22]). The lower part of Fig. 3 is essentially a decoder (excluding motion estimation and with the addition of variable length decoding). SI-frames are directly transmitted when necessary but missing from Fig. 3 is a further stage of SSP-processing. This consists simply in forming the residual from the previous stored P-frame that resides in the frame buffer.

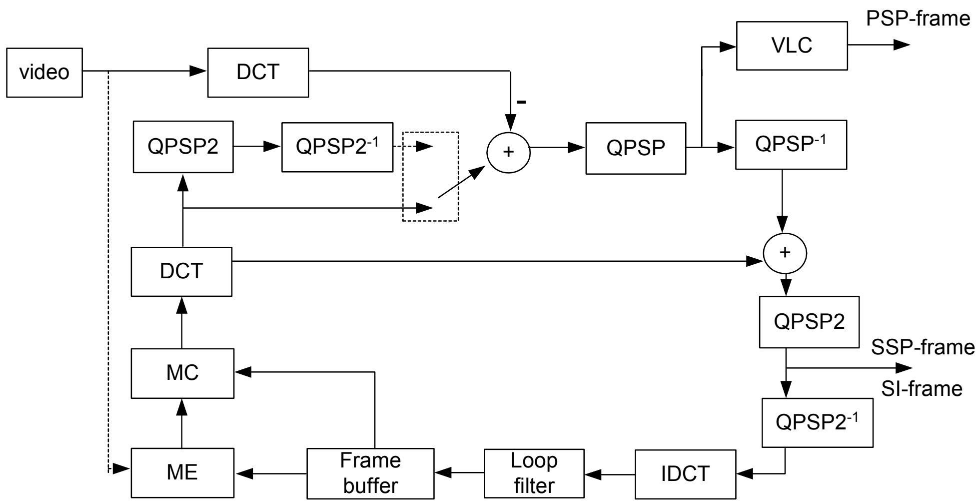


Figure 3. H.264/AVC bitstream switching encoder, ME = motion estimation, MC = motion compensation, (I)DCT = (Inverse) Discrete Cosine Transform

From Fig. 3, there is no difference between the way the residual is coded to produce a P-frame's bitstream and the way the residual is coded to produce a PSP-frame's bitstream. However, because the second quantization introduces further quantization errors it is necessary to reduce the distortion that would arise as subsequent P-frames were reconstructed. Therefore, QPSP should be set to the value of the QP for I- and P-frames or be set to lower than that QP. QPSP2 should be lower still to also compensate for the second quantization as well as the change in bitrates between the original and target stream. Selection of suitable values for QPSP and QPSP2 in bitstream switching is addressed in [24]. Tests were conducted in [24] with a variety of video sequences at various bit rates, to obtain rate versus distortion curves. If QP is the variable bit-rate quantization parameter for P-frames then to obtain PSP- and SSP-frames and to ensure reduced distortion relative settings for QPSP and QPSP2 were found empirically to be:

$$QP - 5 < QPSP < QP \quad (1)$$

$$QP - 10 < QPSP2 < QP - 8 \quad (2)$$

Recall that setting QPSP or QPSP2 to a value less than QP results in a higher quality result (for a single quantization step).

Furthermore, in [21] quantization parameters for SP- and SI-frames were chosen so that the rate-distortion tradeoff was optimized. The selection was based on the probability, x, of sending switching frames. If x is the probability of using switching frames, then optimum values for QPSP and QPSP2 can be defined as follows, as reported in Table 1. When $x < 0.1$, the selections of QPSP and QPSP2 lead to high peak data rates. However, these rates do not exceed those of sending I-frames.

x	< 0.1	≥ 0.1 and ≤ 0.19	> 0.19
QPSP	*QP-1*	*QP-2*	*QP-3*
QPSP2	*QP-10*	*QP-5*	*QP*

Table 1. QP values for PSP- and SSP-frames, where x is the probability of using switching frames, from results obtained in [21].

4. WiMAX broadband wireless

4.1 WiMAX standard

WiMAX [1] is standardized under IEEE 802.16. IEEE 802.16 comprises protocol specifications for physical layer and Medium Access Control (MAC). Over time, a number of improvements and changes have been attached to the standard. IEEE 802.16d-2004 (known as fixed WiMAX) supports fixed and portable devices for line-of-sight and non-line-of-sight conditions. IEEE 802.16e-2005 [3] supports mobile devices, handoffs, roaming and multiple users rather than a single broadcast channel. In this paper, IEEE 802.16d-2004 is used for simulations but the results broadly apply to the mobile version.

WiMAX channel just as every wireless channel suffers from channel impairments such as path loss, shadowing, multipath and fading. Path loss and shadowing are large-scale attenuation effects, which are due respectively to absorption and scattering over the transmission path and diffraction in the presence of obstructions. Fading is caused by multipath effects i.e. multiple versions of transmitted signal at the receiver, each of which has differences in phase and amplitude leading to constructive or destructive interference of the signal at the receiver. At the receiver, slow fading due to changes in the channel environment in particular manifests itself in error bursts and for that reason a Gilbert-Elliott [25, 26] burst error model is employed in this paper.

The WiMAX physical layer supports both TDD and Frequency Division Duplex (FDD) modes. However because TDD allows more flexible sharing of bandwidth between uplink and downlink, and has a reciprocal channel, it is used in this paper. In TDD mode, a frame is divided into two sub-frames: uplink and downlink. The ratio between downlink to uplink sub-frames is configurable between 3:1 to 1:1. In a physical-layer frame, multiple users and packets may be multiplexed in a single uplink or downlink frame. The frame size is variable between 2 ms to 20 ms. In addition each burst can contain multiple fixed or variable-sized packets from higher layers.

The MAC sub-layer of IEEE 802.16 provides an interface between physical layer and higher transport layers. The MAC layer receives the packets from higher layers (a packet received from higher layers is called a MAC Service Data Unit (MSDU)) then organizes them into MAC Protocol Data Unit (MPDU) to transmit them over the air. In particular, the MAC supports variable-sized MPDUs and it allows concatenation or fragmentation of MSDUs into a single or number of MPDUs.

4.2 Feedback mechanism

Switching between different streams is possible at every PSP-frame, if the server becomes aware of adverse channel conditions. In the proposed feedback mechanism, the channel condition is specified by measuring the packet loss and a notification is sent by the receiver to the server if the number of losses passes a threshold. In the current implementation the packet loss threshold between switching frames is the loss of just one packet between PSP-frames. As ACK packets are sent every several packets, the overhead is limited.

Assuming that the initial stream is of high quality, the server will change to a low quality stream if it receives a packet loss announcement. On the contrary, when a low-quality stream is being sent and no packet loss is experienced for the period between switching frames, the streaming server waits for the next PSP-frame and changes back to high-quality video. Using this mechanism, low-quality packets experience more time in bad channel conditions than high-quality packets. Therefore, an improvement in video quality is expected.

If a frame loss occurs, errors will propagate, unless there is a decoder reset, as all the frames are encoded using inter-prediction. However, the severity of the effect is enhanced when errors propagate through the loss of switching frames of any kind. In this case, the entire video stream of the other encoded version will be affected, which will naturally cause degradation in received video quality. Error propagation through the loss of video frames can be prevented through the use of intra-coded SI-frames. By reducing the relative size of SI-frames, the probability if losing them can also be reduced.

4.1.1 ACK protection of switching frames

To reduce the effect of loss of SSP-frames, a mode with additional ACKs as well as adaptive stream switching was implemented. A streaming server waits for a round trip time (RTT) to receive an ACK of a switching frame packet (SSP- or PSP-frame). Each successfully received packet provides a sample of the duration between sending and receiving a packet. The average RTT value for each packet, $n = 1, 2, 3, \ldots$, is updated as a moving average as in (1).

$$RTT_{avg(n)} = 0.9 \times RTT_{avg(n-1)} + 0.1 \times RTT_{n-1} \tag{1}$$

When an RTT interval expires without an ACK, the switching frame is re-sent. The number of retransmission must be kept to minimum value to avoid extra delay that is harmful to a multimedia application. In the implementation evaluated, the number of retransmissions was limited to two retry limits. This value was chosen so that a compromise between delay and packet loss requirements of the system was met.

4.3 'Bursty' channel model

During periods of time in which the signal amplitude is below a certain level, the state of the channel causes higher bit error rates compared to other times and when the fade duration is long, burst errors occur. To model the burstiness error property of a wireless channel, a two-state Markovian chain was implemented. A two-state Markov or Gilbert-Elliott error model comprises a good and bad state. This model was first used in [25] for modeling bit errors. In this model, Gilbert assigned a bit error probability of zero for a good state and a binary symmetric channel for the bad state. In [26] Elliott improved the Gilbert model by assuming a binary symmetric channel for the good state as well.

In the model, Fig. 4, p_{gb} is probability of transition from the good state to the bad state and likewise p_{bg} is the probability of transition from the bad state to the good state. Then p_{gg} and p_{bb} are the probabilities of staying in the good state and the bad state respectively, according to (2).

$$p_{gg} = 1 - p_{gb}, p_{bb} = 1 - p_{bg} \tag{2}$$

Each state is assigned a specific bit error rate (BER). Thus, e_g, e_b are the BERs in the good and bad states respectively, where $e_g << e_b$. Steady-state probabilities for staying in the good and bad state are defined as π_G and π_B and found from (3).

$$\pi_G = \frac{p_{bg}}{p_{bg} + p_{gb}}, \pi_B = \frac{p_{gb}}{p_{bg} + p_{gb}} \tag{3}$$

As a result, average BER is defined in (4).

$$BER_{avg} = e_g.\pi_G + e_b.\pi_B \tag{4}$$

According to [27] finding the average burst length (or average burst time parameter) is a method to find the Gilbert-Elliott model parameters. Average burst length is defined in (5).

$$T_B = \frac{1}{p_{bg}} \tag{5}$$

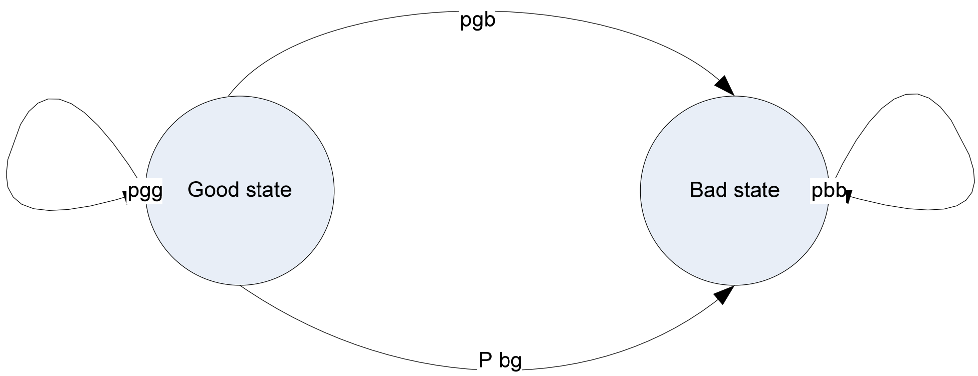


Figure 4. Gilbert-Elliott two-state model showing transition probabilities

In [5], average, good state, and bad state BERs are defined for a specific WiMAX channel, as given in Table 2. With BER_{avg}, e_g, e_b, defined, then by means substituting (3) in (4) it is possible to relate p_{gb} to p_{bg}. Specifically, for the figures in Table 2,

$$p_{gb} = \frac{1}{8} p_{bg} \quad . \tag{6}$$

Parameter	Value
BER in good state	10^{-4}
BER in bad state	10^{-3}
Average BER	2×10^{-4}

Table 2. BERs from [5] leading to the relationship given in (6).

From predicted BERs, the packet error rate can be found [27]. A packet is considered correct when all of its bits are received correctly, leading to (7), with *p(.)* denoting a probability.

$$p(packet\_error) = 1 - (1 - p(bit\_error))^{packet\_length} \tag{7}$$

This relationship can be expressed in terms of the two-state model [29]. Denote the status of sent and received bits by X, Y, respectively and set Z as the packet status and l as the average packet length. The event status of $X = \{g, b\}$, where g is being in the good state and b is being in the bad state. Further, let A denote that a packet or bit is received correctly and B that a packet or bit is erroneous. Assuming the initial channel condition of good, the probability of a packet being error free is then found from the law of total probability as:

$$p(z = A \mid X_0 = g) = \sum_{j=1}^{j=l-1} p(Y_j = A \mid Y_0 = A) \prod_{i=1}^{i=j-1} p(Y_i = A \mid X_i) p(X_i \mid X_{i-1}), \tag{8}$$

Therefore, the probability of packet being erroneous is:

$$p(Z = B \mid X_0 = g) = 1 - p(Z = A \mid X_0 = g) \tag{9}$$

5. Evaluation

5.1 Methodology

For encoding of raw YUV video files, a modified version of JM10.2[2] was used. However, as this codec does not support switching functionality with PSP/SSP- and SI-frames, Eric Setton's amendment[3] to the codec was used. Therefore, this simulation employs the simplified design as accepted by the H.264/AVC development team, JVT, which slightly differs [22] from that presented in [2]. To generate simulation results the network simulator ns2 v. 31 was used[4] with the NIST WiMAX module[5]. Each data point obtained is the average of ten runs. As the Evalvid tool [30] for assessing video quality after packet losses is only able to cope with the basic facilities of H.264/AVC, AWK scripts [31] were constructed to remove lost data from the compressed bitstream based on the ns output file. The resulting PSNR was found by comparison with the YUV video file. VideoMeter [32] was employed to assist with subjective assessment.

The Foreman sequence with QCIF resolution (176×100 pixel/frame) was encoded at 30 Hz. The well-known Foreman sequence contains a close-up head view followed by a pan to another view. Therefore, its coding complexity is moderate to high. A Group of Pictures (GoP) size of eight consisting of an initial PSP-frame and seven P-frames was adopted, with one I-frame to start the sequence of 399 frames. Other GoP sizes are considered in Section 5.3.) Recall that when I-frames are used a GoP size of 15 is normal for a 30Hz stream. For clarity of interpretation, two streams are switched. The implications of testing with two streams easily extend to more than two streams. For comparison with

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[2] H.264 software coordination, Software version JM10.2, available from `http://iphome.hhi.de/suehring/tml/`

[3] Setton, E.. Encoder for switching with SP and SI frames. Stanford University, Available from *`http://ivms.stanford.edu/~esetton/H264_2.htm`*

[4] The network simulator NS-2 version 31, available from `http://www.isi.edu/nsnam/ns/`

[5] The Network Simulator NS-2 NIST add-on IEEE 802.16 model (MAC+PHY), `http://www.antd.nist.gov/seamlessandsecure/download.html`

others' results and practical implementations, only previous frame replacement was used by way of error concealment.

The higher quality QP was selected to be 20 and that of the lower QP to be 38. QPSP and QPSP2 are chosen based on the discussions in Section 3.3 on how rate distortion may be optimized by selecting particular ranges for the QPs. According to Section 3.3, different values are selected depending on whether SSP-frames or SI-frames are chosen. As different values are chosen for the QPs depending on whether there is an up or down transition, these values are summarized in Table 3.

Table 3 brings with it some important implications. As the QPSP values for streaming switching with SI-frames are lower than the corresponding values for switching with SSP-frames, PSP-frames in the SI-frame streams (both high- and low-quality) will be of higher quality. Consequently, the PSP-frame packets will be larger in size for the SI-frame switched stream. Conversely, the relative quality of the SI-frames will be lower than that for the SSP-frames, because of the QPSP2 values (which are higher for the SI-frames).

Though it is entirely possible to adopt other error resilience methods, for the sake of clarity of identifying the effects in tests, the only resilience method was through switching frames. PSP-frames were inserted every eight frames to give a refresh rate, as it were, of 266 ms, close to the 320 ms recommended in [22].

Switching type	QP	QPSP	QPSP2
SSP High	20	19	10
SSP Low	38	37	28
SI High	20	17	20
SI Low	38	35	38

Table 3. Quantization parameter settings for higher rate bitstream and high-to-low transitions, lower rate bitstream for low-to-high transitions, when using either SSP or SI switching frames

The basic MAC and physical layer features can be seen in Table 2. The Fast Fourier Transform (FFT) size relates to the modulation method, Orthogonal Frequency Division Multiplexing (OFDM), as does the cyclic prefix, which provides a kind of guard interval between symbols [1]. The settings in Table 4 are intended to be indicative for testing purposes and do not necessarily correspond to implemented values. The two-ray ground propagation model is suitable for modeling line-of-sight communication, as in practice just two paths tend to dominate [33] the received signal strength. The fragmentation capability of the MAC was enabled to take advantage of the reduced error probability for smaller sized packets. In this work, average size of the fragmented block was set to 20 B. Therefore, the average block error rate is about 3% from equation (7).

Parameter	Value
Channel Bandwidth	6 MHz
FFT size	256
Coverage radius	0.5 km
Frame duration	4 ms
DL/UP sub-frame ratio	0.3
Cyclic prefix	0.25
Propagation model	Two-ray ground

Table 4. Basic configuration for WiMAX

5.2 Packet loss and delay

Figure 5 presents packet loss statistics. As expected packet loss increases with increasingly duration of bad channel conditions. This is most marked when sending the high quality stream without switching. The packet loss percentage for switching with SSP- and SI-frames is close. However, two points must be considered for these two cases:

1. For most of the time, the packet loss percentage of switching with the bitstreams using SI-frames is a little larger than that the streams switching with SSP-frames. The reason for this is that in switching with SI-frames, the PSP-frames in the high- and low-quality bitstreams are larger than the equivalent PSP-frames in stream-switching with SSP-frames. Therefore, there is an increased probability that more of the PSP-frames will be lost from the SI-frame switched bitstreams. However, because the SI-frames can reset the decoder, the effect of the loss of PSP-frames is mitigated.
2. The variation of packet loss when switching is used is not as significant as in the no-switching scenario. The standard deviation values shown in Table 5 confirm this. The reason for this lies in the main objective of using switching frames: switching frames are used adaptively based on channel condition. Therefore, they tend to keep packet loss constant.

The last streaming mode tested was a combination of adaptive ARQ with stream switching with SSP-frames. It is obvious that retransmissions will reduce the total packet loss percentage as it can be seen from Figure 5. The adaptive ARQ scheme is more effective when the bad state duration is more than 50%. The aim of using adaptive ARQ scheme is to avoid the loss of PSP- and SSP-frames and increase the received video quality (PSNR), which it achieves when the bad state durations increase.

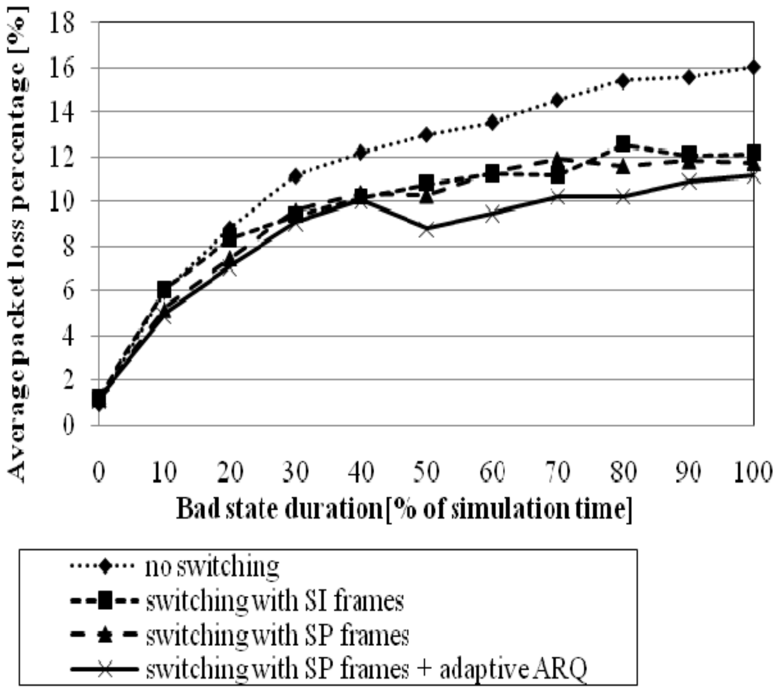

Figure 5. Mean packet loss rate with increasing bad state durations

Streaming mode	No switching	With SI-frames	With SSP-frames	With SSP-frames + adaptive ARQ
Standard deviation	4.69%	3.36%	3.43%	3.04%

Table 5. Variation in packet loss over the simulation runs

Figure 6 shows the average (arithmetic mean) end-to-end delay for the same system modes of Figure 5. The delay is almost constant when no-switching is used. The reason is that when the packet loss increases, the application continues to send the packets as if no packet loss has occurred and the average delay remains constant. In the adaptive streaming modes, when the bad state time increases the delay decreases. This is because when a bad state occurs, if it is necessary, a switch is made to a lower sending rate. A lower bitrate introduces lower delay, while a higher bitrate introduces larger delay values. Increasing bad channel time, applies frequent switching to low quality and subsequently lower delay results. The PSP-frames within SI-frame switched bistreams consist of larger packets generating higher average bitrates. When switching with SSP-frames is combined with the adaptive ARQ scheme (protection for switching frames), the average delay is significantly higher than all of the other modes. The reason is obvious: retransmission increases the reception time. However, for the WiMAX link simulated the actual delays in all streaming modes are minimal, being mostly less than 10 ms.

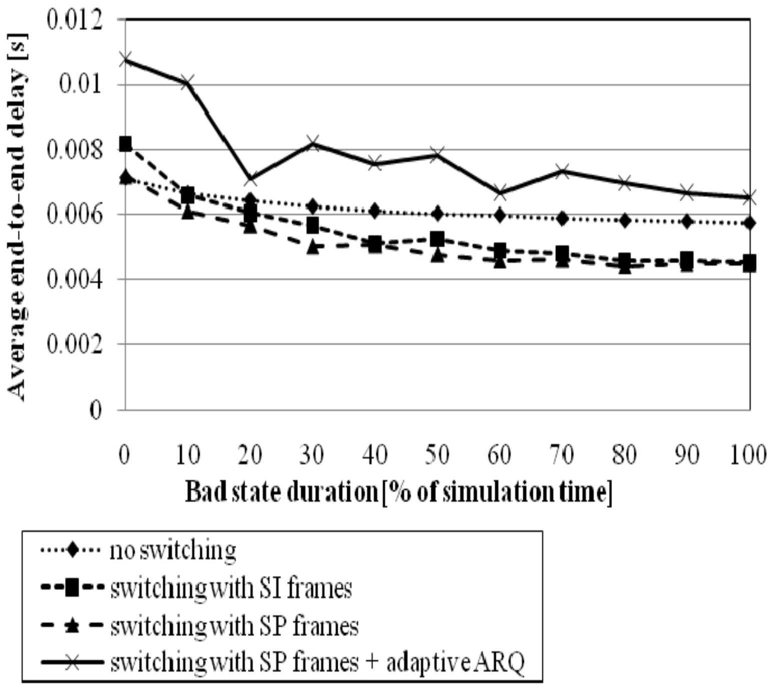

Figure 6. Per packet end-to-end delay over the WiMAX link with increasing bad state durations

The throughput or strictly goodput (i.e. successfully received data) is shown for the different streaming modes in Fig. 7. The stream with no switching results in a higher throughput as the higher quality compressed video continues to be transmitted despite packet losses. For the switched streaming schemes, the lower bitrate stream is increasingly chosen as the channel conditions deteriorate. The throughput of the stream employing SI-frames is greater than that of the stream employing SSP-frames, but not noticeably so. This indicates that the smaller sized SI-frames to some extent counter the larger PSP-frames in switching with SI-frames.

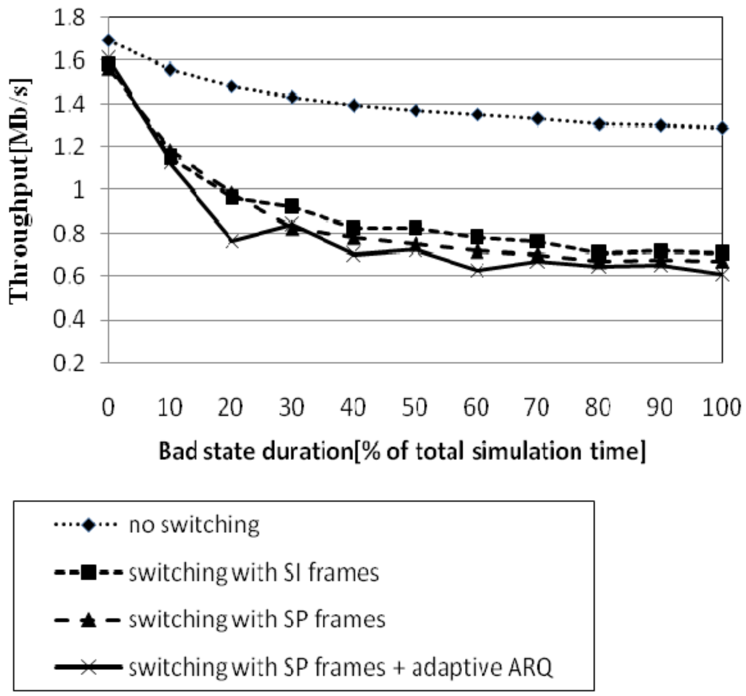

Figure 7. Throughput arising from different streaming modes according to bad state duration

5.3 Video quality

The high-quality video's PSNR on a frame-by-frame basis for an error-free WiMAX channel is shown as the top plot in Fig. 7. Introducing 10% duration in the bad state with SSP-frames and switching results in the lower plot in the Figure. This plot is one example from the set of simulated streaming sessions. From Fig. 5, 10% in the bad state approximates to 5–6% packet loss. The fluctuations in frame-by-frame quality in the lower-quality streaming session are due to packet loss.

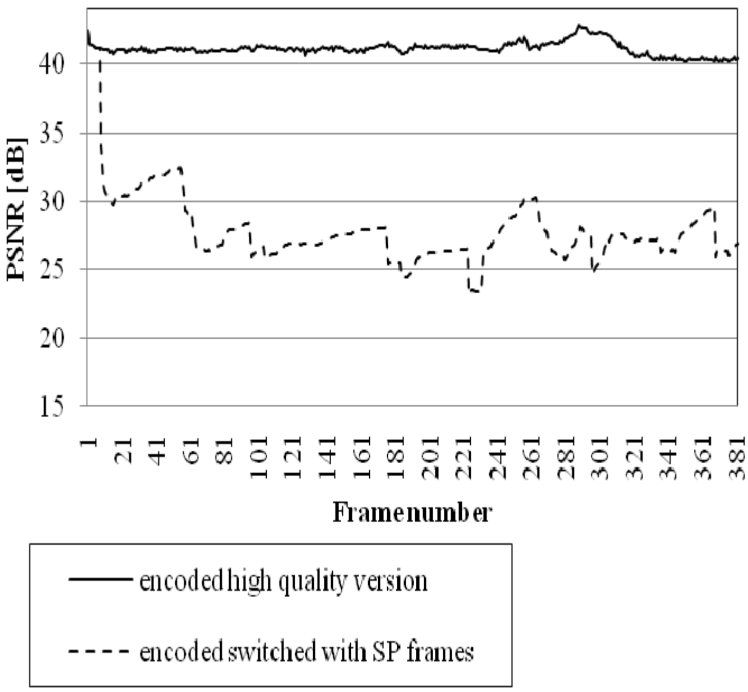

Figure 8. Sample streaming session showing the video quality after decode for the high quality version in an error-free channel and after encoding with SSP-frames with a 10% duration of bad states

Overall assessment of video quality for the Foreman sequence is shown in Fig. 9. The PSNR for switching with SI frames is significantly higher than all the other modes, even if its packet loss rate is sometimes more than the packet loss rate when switching with secondary SP-frames. The reason for this is the intra-coding employed for SI-frames, which prevents error propagation. The other two switched schemes improve upon 'no switching'. However, in the case of switching with SSP-frames, much of this improvement must be attributed to the presence of the improved quality PSP-frames, as will be evident from the PSNR values at zero loss. Employing the adaptive scheme to retransmit switching frames is an improvement but the limited number of retries does not improve the quality enough to compete with the use of SI-frames. Referring back to Fig. 6, there is about a 2 ms impact from retransmission, which would obviously increase if more retransmissions were permitted.

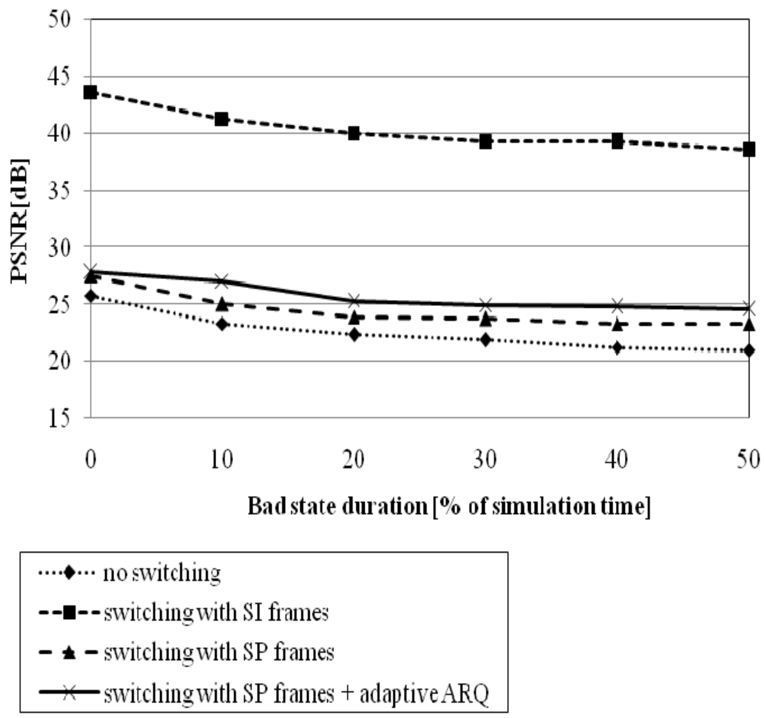

Figure 9. Average video quality with increasing bad state durations with the different streaming modes

The interval between PSP-frames was changed to examine the GoP size dependency. From Fig. 10 (for an average duration in the bad state of 16.6%, equivalently about 7–8% packet loss rate) it is apparent that shortening the GoP size can indeed improve the performance of the SSP-frame switching schemes. Furthermore, this Figure shows that when switching is combined with the adaptive ARQ scheme, a relative average of 1.74 dB improvement in PSNR occurs for all GoP sizes. The switching flexibility also increases, with switching available about every 7 s. However, obviously bitstream size and storage overhead both increase, as discussed in the next Section.

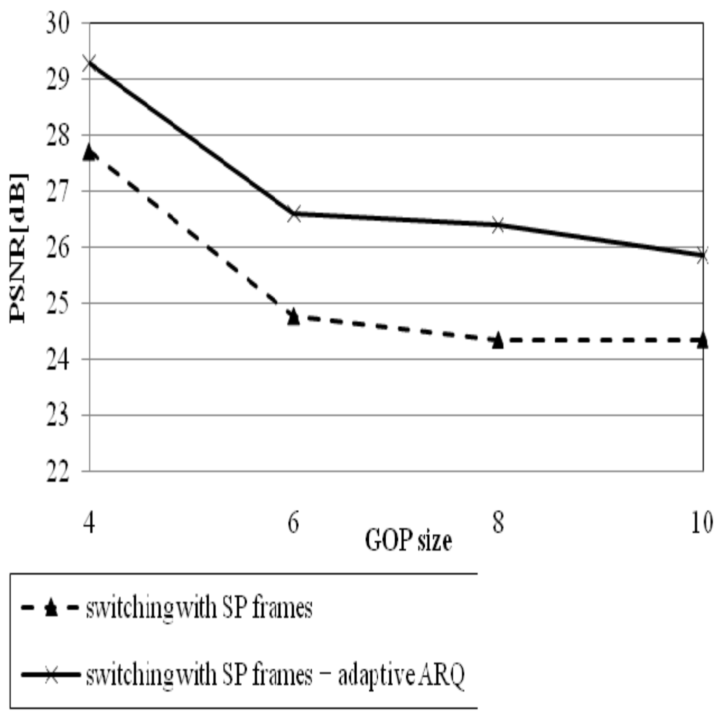

Figure 10. Video quality with change of PSP interval (GoP size) with average bad state duration of 16.6%, with comparison to the high-quality decoded version

5.4 Switching overhead

The extra overhead incurred by sending switching frames in presented in Table 6. In this Table, the total contribution to the bitstream in size of the switching frames (SSP- or SI-frames) is expressed as a percentage of the total bitstream size. Notice that it is still necessary to switch within the good state, even at the lower BER. As a consequence of setting a high QPSP2 in the SSP-frame experiments (see Table 3 and the discussion in Section 5.1) the contribution of SSP-frames is larger than that for the SI-frames. The actual percentage of switching frames transmitted depends on the number of stream switches occurring during the simulations. However, it is clear that the trend is slowly rising as the duration in the bad states increases. Therefore, depending on switching frame type and quantization parameter selected there is likely to be up to about 25% overhead in transmitting switching frames. However, the switching frame overhead is not the whole story as the sizes of the low- and high-quality bitstreams vary because of the relative sizes of the encoded PSP-frames.

Time in bad state[%]	SSP percentage of total transmitted bitstream	SI percentage of total transmitted bitstream
0	5.69	4.48
10	20.21	17.23
20	17.58	21.79
30	22.82	20.31
40	28.97	20.12
50	23.30	20.52
60	23.30	20.86
70	25.71	21.81
80	25.65	22.08
90	22.81	21.31
100	24.35	21.46

Table 6. Size of SSP/SI frame bitstream as a percentage of transmitted bitstream according to the duration of time spent in the bad state.

Table 7 compares the storage overhead from the need to store all stream switching frames according to GoP size. Two sets of frames are required to make upwards and downwards transitions. As mentioned, because of the choice of a fine quantization value for QPSP2, the SSP-frames occupy more storage space than SI-frames. In fact, as the GoP size is reduced the relative advantage in storage terms of SI-frames increases, though obviously this is dependent on choice of quantization parameter values. Clearly, a storage increase of over 60% for a small switching interval of four frames is a factor to consider when storing large numbers of lengthy videos. However, in general storage should not need to be above 40% of the whole for two streams. From Table 2, the impact of increasing the quality of SI-frame switching stream PSP-frames can be judged. For example, for a GoP size of eight, as in the majority of the tests, the higher-quality bitstream is about 64% larger when using SI-frame switching for the chosen QP values, which is compensated for in the total stored bitstreams by the lower quality SI-frames.

High-to-low switching frame bitstream size [byte]	Low-to-high switching frame bitstream size [byte]	High-quality bitstream size [byte]	Low-quality bitstream size [byte]	Total stored data [byte]	% of switching frames in the total bitstream	GoP size/ switching frame type
780389	792322	921164	61775	2555650	61.54	4/SSP
446716	459941	1411179	193286	2511122	36.11	4/SI
437776	451285	907105	61180	1857346	47.87	6/SSP
290738	429308	1230806	148433	2099285	34.30	6/SI
542172	133434	900658	60556	1636820	41.28	8/SSP
511895	69132	1222036	125402	1928465	30.13	8/SI
259605	267735	896666	60330	1484336	35.53	10/SSP
171177	252366	1091842	112024	1627409	26.03	10/SI

Table 7. Sizes of stored SSP/SI frame bitstreams, showing the percentage of transmitted bitstream according to the GoP size and switching frame type. Sizes are to the nearest byte.

6. Conclusion

In this paper, adaptive stream switching (with SSP- and SI-frames) a feature of H.264/AVC is used to provide more protection for video data against bad channel conditions. When the channel condition goes into bad state a server switches to lower-quality video and *vice versa*. This dynamic switching mechanism was enabled by sending side information about packet losses. A further extension provides repeat transmission of switching frames. The results obtained from the simulations verified the expectation that with increasing bad state duration or equivalently longer burst lengths stream switching improves video quality and link delay. However, switching with SI-frames should normally be given preference over secondary SP-frames. The increased throughput from using SI-frames is moderate and should not inconvenience traffic from other sources using the WiMAX link. This is because it is possible to send lower quality SI-frames but still halt error propagation, whereas errors continue to propagate when SSP-frames are sent. In compensation, if the quality of SI-frames is reduced then the quality of primary SP-frames should be increased to optimize rate-distortion tradeoffs. Storage overhead can be considerable if a high (every four frames) switching frequency is provided for. Using SI-frames, at a switching frequency of eight, results in a 30% overhead from switching frames. The transmission overhead from sending SI-frames in the tests was around 20%. However, all this is relative to the switched bitstreams and their transmission frequencies. By varying the QP values it becomes possible to tune the overheads in a way that accounts for delivered video quality and throughput.

In this work, only the performance of switching techniques was investigated and no error resiliency features of the H.264/AVC were used during the encoding process. Using the error resiliency tools of H.264 encoder should significantly improve stream switching performance and some of these tools such as gradual decoder refresh do not result in significant overhead even at lower bitrates.

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