Adaptive Video Stream Switching for an IEEE 802.16 Channel

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Abstract— **The aim of this paper is to enhance video streaming over an IEEE 802.16d (WiMAX) broadband wireless link through the application of encoded bitstream switching. The 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 together with SI frames bring a significant gain in video quality compared to other switching schemes with secondary SP-frames. The other schemes in turn show an improvement to using 'no switching' when streaming takes place over a typical WiMAX channel with burst errors. Link delay is also reduced.**

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

I. Introduction

The main contribution of this work is a scheme for switching frames for broadband wireless, especially during 'bursty' channel error conditions. Specifically, it proposes adjustment of the bit-rate to reduce the impact of adverse channel conditions leading to packet loss. The paper investigates choice of stream-switching with Secondary SP-frames or SI-frames relative to the selection of quantization parameter (QP) values. To control the switching points at the WiMAX server, the proposed scheme applies a feedback mechanism that monitors packet loss. The paper also considers an adaptive Automatic Repeat reQuest (ARQ) scheme to protect switching frames against packet loss. Work on cellular wireless applications of stream switching [1] concentrated on stream switching as a way of varying the bitrate to reduce the risk of buffer overflow or exceeding the channel capacity, while this paper proposes that switching takes place in a different manner.

Broadband wireless in general is an alternative to other types of access networks such as cable and Digital Subscriber Line, while WiMAX itself can also support mobile users. IEEE 802.16d (known as fixed WiMAX) [2] allows rapid deployment of video services in rural areas in the world that are unlikely to benefit from extensions to Third Generation (3G) cellular systems. Notice that fixed WiMAX is more widely deployed than the mobile variety, IEEE 802.16e. In particular, IPTV is an attractive application of WiMAX [3]. Multicast is regarded as a baseline service by some IPTV providers, whereas unicast services such as video-on-demand for TV are commercially attractive, as they represent 'value-added' services. In the not-for-profit sector, the BBC iPlayer is an example of a public broadcasting company providing unicast streaming for time-shifted TV and 'start-over' live TV programs. In areas where a 3G system 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 WiMAX multimedia broadcast and multicast service (MBMS) is similar for WiMAX and LTE [4]. In fact, as IEEE 802.16m [5] approaches deployment, other than the form of uplink modulation, the performance of each will be similar.

An effective form of bitrate adaptation amongst Internet broadcasters is simulcast in which pre-encoded video streams at different bit-rates are switched between, depending on congestion conditions. The same approach can also be adopted for varying wireless channel conditions. Though switching can occur at spatially-encoded anchor pictures (I-pictures), these pictures result in sudden increases in bitrate when encoding at a variable bitrate for constant quality. Therefore, to provide smooth switching between streams at reduced bitrates we employ switching frames, namely an H.264/AVC (Advanced Video Coding) codec's SP/SI-frames [6], which form part of the Extended profile. Acknowledgment messages are sent to identify when switching is necessary to avoid deterioration in the video quality.

Because a feedback channel is required, the scheme is suited to unicast delivery and not to multicast services such as MBMS [4]. Delay should also be limited across the feedback channel. To that end, the Time Division Duplex (TDD) frame structure of IEEE 802.16 with send and receive sub-frames [2] conveniently provides a feedback channel from the streaming client to the server. The streaming client records packet losses and uses this information to decide if a stream switch is required. A request is then transmitted to the streaming server. If switching is performed, it can be both from high-to-low and from low-to-high quality video streams, depending on packet loss statistics. The paper subsequently investigates trade-offs in the choice of predictively-coded or spatially-coded secondary switching frames (SP- or SI- frames respectively).

The remainder of this paper is organized as follows. Section II introduces the background required for an understanding of this paper. Section III outlines our proposed switching scheme, whereas Section IV describes the simulation model in preparation for evaluation of the scheme in Section V. Section VI provides some concluding remarks.

II. CONTEXT

Stream switching presents a natural progression from commercial simulcast streaming systems, allowing these systems to be adapted for use over a wireless channel. Compared to the complexity of scalable coding, their computational complexity is small and 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. 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 [9], 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.

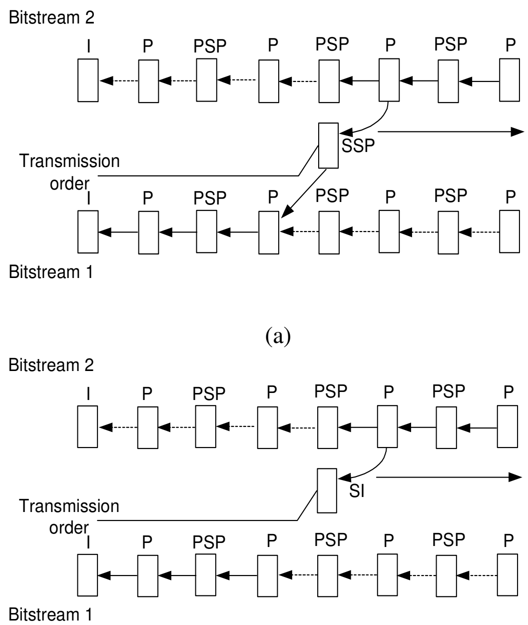
Using I-frames (key frames) has some drawbacks: the compression efficiency of I-frames is low, as their compressed size is approximately 5-10 times higher than that of predictively-coded P-frames. Therefore, inserting many switching points in a bit stream results in a significant increase in bandwidth. In fact, I-frames are normally placed every 30 s, which is a significant time to wait while visual artifacts caused by encoder-decoder drift occur. In the event 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. Instead, switching frames can be used for stream switching and, 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.

H.264/AVC SP- and SI-frames were originally proposed by Karczewiz and Kurceren [6]. The aim of the SP/SI frames [6] is to "enable reconstruction of identical frames using different reference frames". Thus, in stream switching between bitstreams 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. 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, whereas 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 temporal error propagation. If the quantization parameter of the SI-frame is appropriately set, 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. If SSP/SI-frames were to be created dynamically, this would cause delay which would prevent their application to interactive applications. In the event of one or more packet losses in a normal video stream without switching, the loss of synchronization normally results in temporal error propagation until the next synchronization point, which is normally the next intra-coded 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. 1, to enable drift-free switching, the streaming server stores the same sequences encoded at different datarates and, therefore, different qualities resulting from different QPs. As mentioned previously, these bitstreams are populated with PSP-frames at the locations where switching is allowed, as shown in Fig. 1. Notice, however, that Fig. 1 is only an illustrative example and that the periodicity of PSP frames can vary, just as the I-frames they replace can. As already mentioned, if switching becomes necessary, an SSP- or SI-frame is transmitted instead of the PSP frame. The arrowed line in Fig. 1 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. 1 the bitstream two data from the start to the second PSP frame are never transmitted.

Transmission overhead for SP-frames is dependent on channel conditions. In preliminary tests, we found that spending 100% of the time in adverse conditions resulted in about 24% and 22% transmission overhead for SP-frames and SI-frames respectively. However, when more favorable conditions were modeled, the overhead was about 6% and 4% respectively. Thus, the transmission overhead could be no more than 5%. Simulcast servers already store a set of



(a)

(b)

Figure 1. Switching between streams using (a) SSP-frames and (b) SI-frames, showing transmission order and predictive dependencies between successive frames

different versions of a video stream according to video quality and these versions include intra-coded I-frames. Because of inefficient spatial encoding, I-frame sizes can be about 50% larger in size [8] than predictively-coded P-frames, which in turn can be twice the size of bi-predictively coded B-frames (if used). Moreover in [9] an innovative scheme for creating switching frames in the frequency domain allowed storage space for the switching frames to be considerably reduced, though at a cost in computational complexity. That paper [9] demonstrated another application of stream switching to multi-view TV, in which the viewer can switch between different views of the same scene. This application also requires feedback to select the view being streamed from the user. However, the work in [9] was targeted at the wired Internet, whereas our work is aimed at wireless networks.

III. PROPOSED SCHEME

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 or counting packet losses and a notification is sent by the receiver to the server if the number of losses passes a threshold. Though the WiMAX standard does include a mandatory requirement for channel state estimation at a mobile station, packet sequence number monitoring is an obvious way at a receiver to detect packet losses. In the current implementation, the packet loss threshold between switching frames is the loss of just one packet between PSP-frames. As feedback packets are at the most 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 transmitted 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 poor channel conditions than high-quality packets. Therefore, an improvement in video quality is expected.

Unless there is a decoder reset, if a frame loss occurs errors will propagate over time, as all the frames were 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 result in a degraded video quality. Temporal error propagation can be arrested by means of intra-coded SI-frames. By reducing the relative size of SI-frames, the probability of losing them can also be reduced.

To reduce the effect of SSP-frame loss, a mode with SSP-frame ACKs was added to the basic adaptive stream switching scheme. A streaming server waits for an RTT to receive an ACK for 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, … , is updated as a moving average as in (1).

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

Whenever an RTT interval expires without an ACK, the switching frame is re-sent. The number of retransmission must be kept to a minimum to avoid extra delay, which can cause display or decode deadlines to be missed. In the evaluation, the number of retransmissions was limited to two retry limits. This value was chosen as a compromise between the delay and packet loss requirements of the system.

IV. SIMULATION MODEL

This Section details the wireless and video configuration prior to evaluation of the proposed scheme.

A. *WiMAX channel modeling*

The basic MAC and PHY layer features can be seen in Table I. Orthogonal Frequency Division Multiplexing (OFDM) was applied with the given cyclic prefix. The settings in Table I are intended to be indicative for testing purposes and do not necessarily correspond to implemented values. The two-ray ground path-loss model is suitable for line-of-sight communication, as in practice just two paths tend to dominate the received signal strength. The fragmentation capability of the MAC was enabled to take advantage of the reduced error probability for smaller sized packets.

TABLE I. BASIC CONFIGURATION FOR WIMAX

Parameter	Value
Channel Bandwidth	6 MHz
FFT size	256
Coverage radius	0.5 km
Frame duration	5 ms
DL/UP sub-frame ratio	3:1
Cyclic prefix	0.25
Path loss model	Two-ray ground

TABLE II. BERS FOR A WIMAX CHANNEL

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

Burst errors are a threat to compressed video [10] because of the predictive nature of source coding. Therefore, we applied the Gilbert-Elliott two-state channel model [11] to show burst errors as experienced by the user application, though the physical channel is not directly modeled. In [4], average, good state, and bad state Bit Error Rates (BER's) were defined for a typical WiMAX channel, as given in Table II.

From predicted BERs, the packet error rate can be found. A packet is considered correct when all of its bits are received correctly, leading to (2), with p(.) denoting a probability:

$$p(packet\_error) = 1 - (1 - p(bit\_error))^{packet\_length} \tag{2}$$

This relationship can be expressed in terms of the two-state model [11]. 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:

$$p(z = A \mid X_0 = g) = \sum_{j=1}^{j=l-1} p(Y_j = A \mid Y_0 = A) \prod_{i=1}^{i=j-1} p(Y_i = A \mid X_i) p(X_i \mid X_{i-1}), \tag{3}$$

Therefore, the probability of packet being erroneous is:

$$p(Z = B \mid X_0 = g) = 1 - p(Z = A \mid X_0 = g) \tag{4}$$

B. Video characteristics

For encoding of raw YUV video files the JM 10.2 1 H.264/AVC codec[1] was used. To generate simulation results the network simulator ns-2 v. 31 was used[2] with the NIST WiMAX module[3]. Each data point obtained is the average of ten runs. AWK scripts 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` from Arizona State University was employed to assist with subjective assessment.

The *Foreman* sequence with Quarter Common Intermediate Format (QCIF)) was encoded at 30 Hz. The well-known *Foreman* sequence contains a close-up head view followed by a rapid pan to another view. Consequently, 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 by default, with one I-frame to start the sequence of 399 frames. 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 others' results and practical implementations, 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. Secondary QPs necessary for generating switching frames, QPSP and QPSP2, were chosen based on how rate distortion may be optimized by selecting particular ranges for the QPs. Different values are selected depending on whether SSP-frames or SI-frames are chosen. Additionally, different values were chosen for the QPs depending on whether an up or down transition is selected. These values are summarized in Table III, as a full discussion of their choice is beyond the scope and space of this paper.

TABLE III. QUANTIZATION PARAMETER SETTINGS FOR HIGHER RATE BITSTREAM AND HIGH-TO-LOW TRANSITIONS, LOWER RATE BITSTREAM FOR LOW-TO-HIGH TRANSITIONS

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

V. EVALUATION

A. Packet loss and delay

Fig. 2 presents packet loss statistics. As expected, packet loss increases with increasingly duration of bad channel

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[1] H.264 software coordination, Software version JM 10.2, available from `http://iphome.hhi.de/suehring/tml/`

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

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

conditions. This is most marked when sending a 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 of stream 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 IV 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 Fig. 2. The adaptive ARQ scheme is more effective when the bad state duration is more than 50%. The purpose of using the 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.

End-to-end packet 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 the bad channel time, results in frequent switching to low quality and, consequently, 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 (providing 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.

The throughput, i.e. successfully received data, is shown for the different streaming modes in Fig. 3. 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

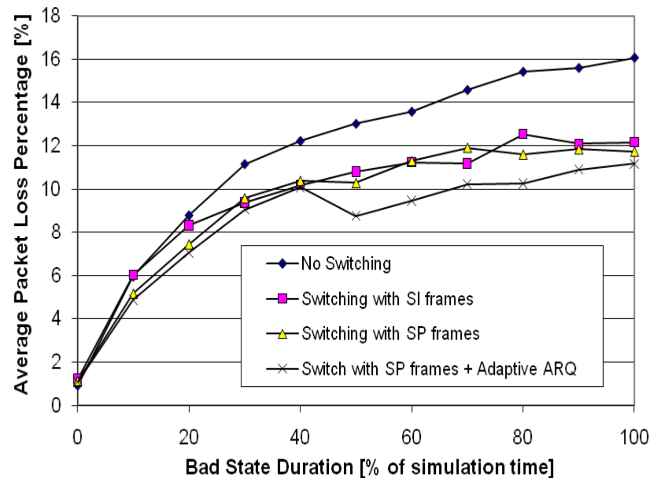

Figure 2. Mean packet loss rate with increasing bad state durations

TABLE IV. VARIATION IN PACKET LOSS OVER THE SIMULATION RUNS

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%

deteriorate. The throughput of the stream employing SI-frames is greater than that of the stream employing SSP-frames but the throughput is not noticeably so. This indicates that the smaller-sized SI-frames to some extent counter the larger PSP-frames in switching with SI-frames.

B. *Video quality*

An overall assessment of video quality for the Foreman sequence is shown in Fig. 4. The PSNR for switching with SI frames is significantly higher than for 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 switching 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. There is about a 2 ms impact resulting from retransmission, which would obviously increase if more retransmissions were permitted.

The interval between PSP-frames was changed to examine the GoP size dependency. From Fig. 5 (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

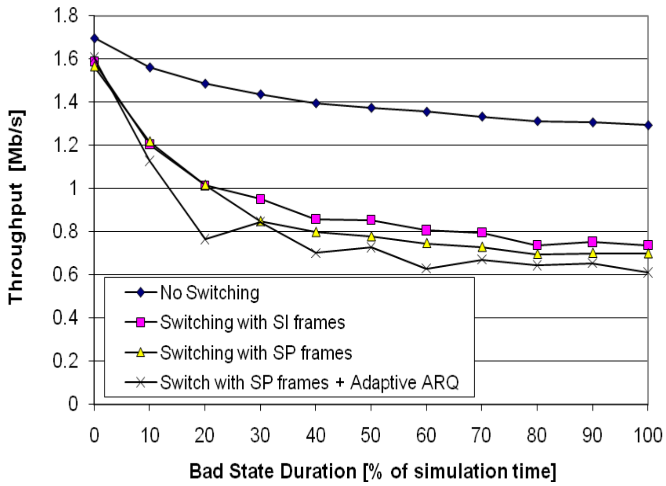

Figure 3. Throughput arising from different streaming modes according to bad state duration

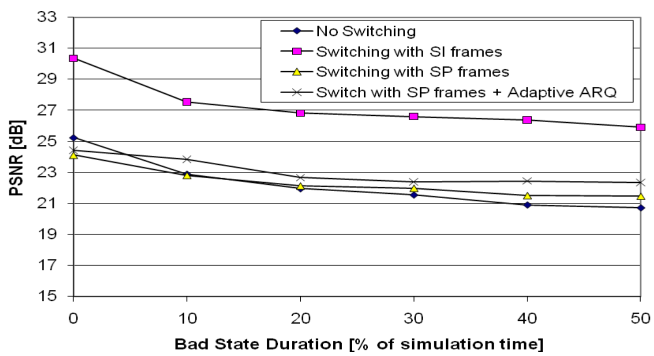

Figure 4. Mean video quality with increasing bad state durations with the different streaming modes

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.

VI. CONCLUSION

In this paper, adaptive stream switching (with SSP- and SI-frames) a feature of H.264/AVC was 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

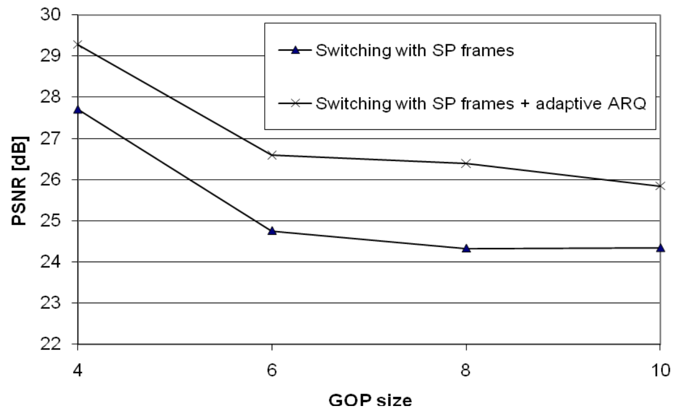

Figure 5. Mean video quality with change of PSP interval (GoP size) with average bad state duration of 16.6%, with comparison to the high-quality decoded version

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.

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