# Multiple Description Video Streaming Over Asymmetric Channels

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***Abstract*—This paper proposes an efficient Unbalanced Multiple Description Scalar Quantisation (U-MDSQ) method for video streaming over asymmetric channels. In order to control the asymmetric target rates for each coded descriptions, the U-MDSQ parameters are combined with the rate control method based on the existing linear relationship between rate and percentage of zeros in transform coefficients in MDSQ domain. The simulation results show that the proposed method exhibits high accuracy for a wide range of target bitrates and unbalanced rates between descriptions. Moreover, the obtained results show the effectiveness of the proposed method in order to improve the overall performance over channels with asymmetric rate and packet loss rate (PLR) conditions, when compared with Balanced MDSQ. This method finds application in video streaming with path diversity based on Multiple Description Coding (MDC) with dynamic channel conditions.**

## I. INTRODUCTION

Multiple Description Coding (MDC) has been used as an efficient approach to improve the video quality in lossy channels, particularly those that can provide multiple paths between the sender and the receiver. In MDC, several streams (i.e., descriptions) are produced by the video encoder and each one is an independent coded representation of the same source signal. The main characteristic of MDC is that any single description can be independently decoded while joint decoding of all descriptions yields higher signal quality than individual decoding of any single one [1].

Path diversity is a robust mechanism to overcome the effects of transmission errors and data loss in the quality of multimedia streaming applications. Therefore, MDC video streaming is particularly useful in highly dynamic networking environments. In MDC video streaming, path diversity networks are used to deliver several descriptions (i.e., compressed streams) from the source to the destination, through distinct transmission channels. For instance, the source can be either a streaming server or a network node connected to the user terminal through P2P networks, wireless links with path diversity, etc. [2].

In general, most of the existing MDC architectures use $N = 2$ descriptions with approximately the same rate, *i.e.* balanced. However, in order to cope with to non-stationarity characteristics of channel conditions in terms of available bandwidth and packet error rates, MDC encoders should dynamically adapt the transmission rate of each description in order to minimize the distortion at decoder. Such requirement implies that Multiple Description (MD) streaming schemes should be able to produce asymmetric bitrate for each description as necessary, *i.e.* unbalanced MDC (U-MDC), where each description may have different resolutions and/or qualities when only one description is decoded.

Unbalanced rates can be obtained by adapting the amount of data, including explicit redundancy that is sent in each description. For instance, in [3], an unbalanced multiple description coding framework is proposed for wavelet-based coders. The proposed method groups different wavelet coefficient trees in each sub-stream based on the available source coding bit rate for each sender, and then each group is independently encoded. The work shows that unbalanced MDC combined with rate allocation algorithm achieves higher performance than conventional balanced MDC. Unbalanced descriptions may also be obtained from scalable coding using scalar quantization with successive refinement, as proposed in [4]. However, in this particular work the unbalanced rates are obtained only when some higher layers are dropped, which means that the equivalent single description quality cannot be obtained from a set of unbalanced descriptions.

In [5] an asymmetric rate control scheme is proposed for H.264/AVC, where the MDC redundancy is adaptively allocated based on an end-to-end distortion model. The overall rate is controlled by jointly setting the central and side distortion. Although resultant rates of the encoded descriptions may vary the corresponding redundancy according to channel conditions, this method does not allow changes in the coded rate of each description without severe degradation in the central distortion. Overall, further research needs to be done in order to generate unbalanced descriptions without losing error resilience capabilities.

Using the concept of index assignment for symmetric descriptions as reference [6], this paper proposes a new unbalanced MDSQ method based on variable index assignment tables. Furthermore, the linear relation between coding rate and percentage of null coefficients evaluation in the MDSQ context, is maintained in each description after applying a cascade of quantisation and MDSQ to H.264/AVC transform coefficients. The proposed method is able to allocate and

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control different rates in each description according to an global bandwidth constraint and asymmetric distribution of the overall rate budget. The results show that the proposed framework is able to choose the rate of each description without losing the overall MDC rate-distortion performance. Although, rate control methods based on the linear model has been used in MDC context, to the best of our knowledge, its use in U-MDSQ applications has not been proposed in the literature.

The paper is organized as follows. Section 2 introduces unbalanced MDSQ (U-MDSQ) and section 3 describes the $\rho$ model for MDSQ. In section 4 is explained the proposed U-MDSQ rate control method. Section 5 presents the simulation results and finally section 6 concludes the paper.

## II. UNBALANCED MDSQ

MDSQ is designed to generate two independent descriptions from the same source signal. Considering only two different descriptions, MDSQ finds coarse scalar quantisers for each side encoder such that a finer central quantiser, producing lower distortion than each individual side decoder, is achieved after combining together each side description. MDSQ is based on two different functions: central quantisation and index assignment. The index assignment is used for mapping each quantisation index of the original transform coefficients $i_0$ (i.e., central indices) into a pair of side indices $(i_1, i_2)$. Considering the unbalanced MDSQ case, the main goal is to generate two descriptions $S1$ and $S2$, in which any rate-distortions pair $(R1, D1)$ and $(R2, D2)$ have $R1 \neq R2$ and $D1 \neq D2$. The central quantiser is characterized by a different rate-distortion pair $(R0, D0)$.

To obtain unbalanced descriptions, a new type of index assignment table is proposed to allow dynamic changes in the number of central coefficients that are indexed as $i_n = 0, n = 1, 2$, as shown in Table I. The proposed unbalanced index assignment method is defined by $k$, which in turn defines the number of diagonals $(2k+1)$ and the central index spread variation parameter $Z$, respectively. The total spread is expressed by $S = 2*(2k)+1+Z$. The spread $S$ is the number of central indices that are coded as zero in a given description $i_n = 0, n = 1, 2$. Different values of $S$ can be defined by varying $Z$ and consequently several index assignment tables can be used to dynamically unbalance each description. For example, Table I shows the index assignment function for $k = 1$ (3 diagonals) and $Z = 0; S = 5$ ($i_1 = 0$ for central indices $i_0 = -2, -1, 0, 1, 2$).

TABLE I
UNBALANCED MDSQ, $k$=1 AND $Z = 0$

| | | i2 (Description 2) | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|
| | | -4 | -3 | -2 | -1 | 0 | 1 | 2 | 3 | 4 | 5 |
| i1 (Description 1) | | -11 | | | | | | | | | |
| | -3 | -9 | -7 | | | | | | | | |
| | -2 | -8 | -6 | -5 | | | | | | | |
| | -1 | | -4 | -3 | | | | | | | |
| | 0 | | | -2 | -1 | 0 | 1 | 2 | | | |
| | 1 | | | | | | | 3 | 5 | | |
| | 2 | | | | | | | 4 | 6 | 7 | |
| | 3 | | | | | | | | 8 | 9 | 11 |
| | 4 | | | | | | | | | 10 | 12 |
| | 5 | | | | | | | | | ... | ... |

Considering the total channel rate as $R = R_{D1}+R_{D2}$ [bps], the unbalanced rate percentage for each description $(\pi_1, \pi_2)$ is defined as

$$\pi_1 = \frac{R_{D1}}{R} \times 100, \tag{1}$$

and

$$\pi_2 = 100 - \pi_1, \tag{2}$$

where $R_{Di}$ is the channel rate used for each description with $i = 1, 2$.

Using a linear model, $\pi_1$ and $\pi_2$ are related with $Z$ by,

$$\begin{cases} \pi_1(Z,t) = m_t Z + b_t, \ \ -1 \leq Z \leq 3, \ \ t = I, P, B \\ \pi_2(Z,t) = 100 - \pi_1(Z,t) \end{cases} \tag{3}$$

This model is used to define different index assignment matrices by computing an adequate value for $Z$, according to a given unbalanced rate percentage. In U-MDC, the model parameters $m_t$ and $b_t$ for the current frame can be computed using coded data from previous frames of the same type.

## III. THE $\rho$ MODEL FOR MDSQ

The $\rho$ model is based on the linear dependence between the coding rate and the percentage of null transform coefficients $\rho$, after quantization [7]. This model is very simple and can be expressed by the following equation, using a single model parameter $\phi$,

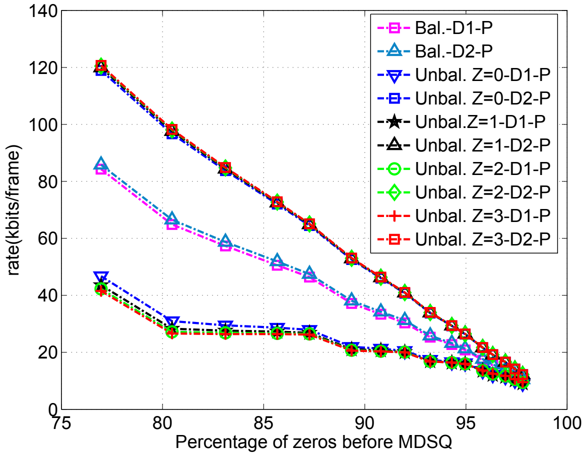
$$R(\rho) = \phi(1 - \rho), \tag{4}$$

where $R(\rho)$ is the rate obtained as a function of $\rho$, i.e. the percentage of null coefficients. The model parameter $\phi$ is obtained from perviously coded units. The R-Q function is indirectly found by relating the source statistics with quantization stepsize. Therefore, the source statistics need to be computed which can be achieved based on parametric models [8]. Other approach is to use previously coded units to find an operational source statistics allowing to find an adequate QP for a given $\rho$ value.

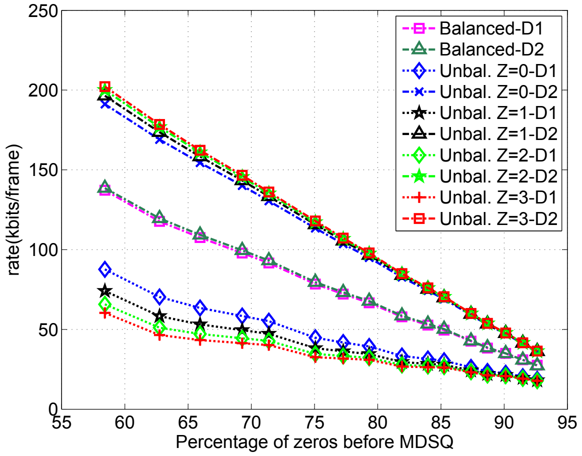
Fig. 1 shows the dependence between the coding rate of each description, resulting from MDSQ encoding of P frames, and the percentage of null coefficients before MDSQ. This example shows that unbalanced descriptions follow a linear relationship between rate and percentage of zeros before MDSQ with the interesting characteristic of keeping the proportionality fairly constant among different unbalanced rates. This is particularly evident in balanced MDSQ and for high rates in unbalanced MDSQ.

## IV. U-MDSQ RATE CONTROL

A new approach for U-MDSQ rate control is proposed in this section, based on the concept of index assignment for



(a) *foreman*



(b) *bus*

Fig. 1. Relation between percentage of zeros and rate.

asymmetric MDC. This approach allows dynamic rate allocation between descriptions by changing the index assignment tables of MDSQ. The proposed rate control method aims at producing two descriptions with different rates, based on the linear dependency between the percentage of zeros of transform coefficients and the rate obtained in each description after MDSQ. A joint encoder buffer for all descriptions is used. The overall rate produced by MDSQ is controlled by finding an adequate central quantisation parameter $QP_0$ and index assignment matrix, such that the predefined target is achieved. This method operates at two different levels: GOP and Frame. At the GOP level, the rate control algorithm finds the rate budget in order to keep an appropriate buffer occupancy under a given channel rate constraint. Also, the unbalanced rate percentage is set according to the channel rate allowed for each description. An extension of the model used in [9] is proposed to include an additional target rate according to the unbalanced rate percentage, in order to define the rate of each description, given the overall target.

At frame level, the rate control method is comprised of two stages. The first stage defines the corresponding overall target rate (i.e., for both descriptions) based on buffer occupancy and frame complexity. Additionally, it computes the target rate for the main description (the one with higher rate) taking into account the unbalanced rate percentage between descriptions. The second stage use the $\rho$ model applied to the main description to compute the target percentage of zeros of transform coefficients, $\rho$, followed by the computation of the stepsize $\delta$ that produce the target description rate. Finally, the central quantisation parameter $QP_0$ is defined based on $\delta$. Each description is encoded based on $QP_0$ and the index assignment matrix determined for a given unbalanced rate percentage. After encoding both descriptions, the $\rho$ model parameter and the index assignment matrix are updated for the next frame. Each step of the method is described in the next subsections.

### A. GOP Level rate control

The GOP level rate control defines the total number of bits allocated to each GOP and the initial quantization parameter of each GOP considering the individual target rates defined for each description. The total number of bits allocated for one GOP is determined as defined in the JVT-G012 rate control method [9]. The unbalance between descriptions is achieved by using unbalanced MDSQ method described in section II.

### B. Frame Level rate control

The rate control at frame level determines the overall target rate for each frame, taking into account the individual rates of each description. The I frame and the first P frame of each $GOP_i$ are coded using the average central quantisation parameter $QP_0$ used on previous GOP. The $QP_0$ for B frames are obtained from interpolation of neighboring anchor frames (I and P) and depends on its distance. Furthermore, the $QP_0$ values for P frames are obtained in two steps. The first step is to determine the target bit rate for each frame, which will be distributed among descriptions according to their corresponding unbalanced rates, obtained by applying the index assignment matrix. The last step is to find the $QP_0$ for achieving the target rate of the main description. The coding rate of the other description is implicitly defined by MDSQ and its unbalanced rate percentage.

*1) I frames:* The I frame of each $GOP_i$ is coded by using $QP_{0_i}(1)$, i.e., the central quantisation parameter of the $i$th GOP, which is defined based on the total number of $I$ and $P$ frames in the $i-1$ GOP, the number of B frames between P frames and the average quantisation parameter used in I and P frames in the $(i-1)th$ GOP.

*2) P frames:* The global target rate (# bits) for each P frame is defined by the sum of the individual description rates. This target rate is determined based on the buffer occupancy after coding each frame, the complexity weights of P and B frames and also the $QP_0$ values used in previous frames. In MDSQ, the complexity weights are updated after encoding each frame, taking into account the total number of bits generated in both descriptions. The target rate for each description is defined based on the target rate for each frame

$T_i(j)$ and the unbalanced rate percentage $(\pi_1, \pi_2)$. This is given by,

$$\begin{cases} T_i(j)_{D1} = T_i(j) \times \pi_1 \\ T_i(j)_{D2} = T_i(j) \times \pi_2 \end{cases} \tag{5}$$

The control algorithm seeks to achieve the unbalanced target rate for the current frame by adjusting the $Z$ parameter and consequently the index assignment matrix, based on the actual rate obtained in each description of the previous frame. This is done by using the linear model defined in (3).

Then, based on the findings presented in section III, a $\rho$ model was devised to determine the central quantisation parameter ($QP_0$) for a given target rate in each description. Note that the first P frame of each $GOP$ is coded with the same $QP_0$ as the I frame. Taking into account the results previously obtained, the model has higher linearity for the description with higher rate, named as main description in this context. Therefore, the model parameter for each frame is computed based on the main description by using the source statistics of the last frame of the same type. Thus, the model parameter $\phi_i$ for frame $i$ is computed as,

$$\phi_i = R_{i-1}/(1 - \rho_{i-1}). \tag{6}$$

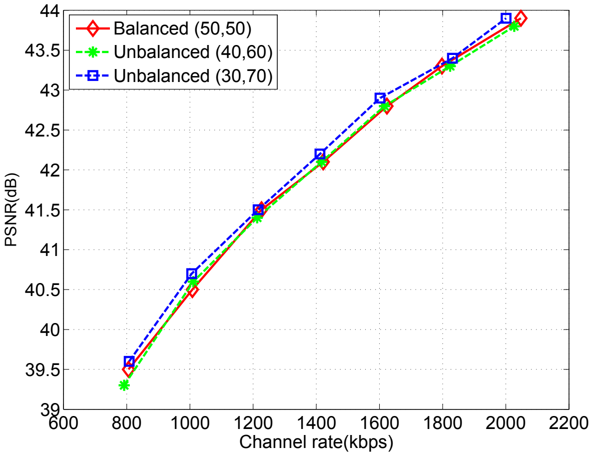
After computing $\phi_i$ for a given frame and setting the target rate for the main description, the corresponding $\rho$ value (i.e., percentage of zeros) is determined using (4). Using the histogram of the transform coefficients from the previous frame, the step size $\delta$ is determined according to the percentage of zeros $\rho$ required to produce the target rate. The relationship between $\delta$ and $\rho$ is established through a lookup table, filled with data from the previous frame. Then, the central quantisation parameter $QP_0$ is determined from the following expression [8]

$$QP_0 = 6 \times log_2(\frac{3\delta}{2}), 0 \le QP_0 \le 51. \tag{7}$$

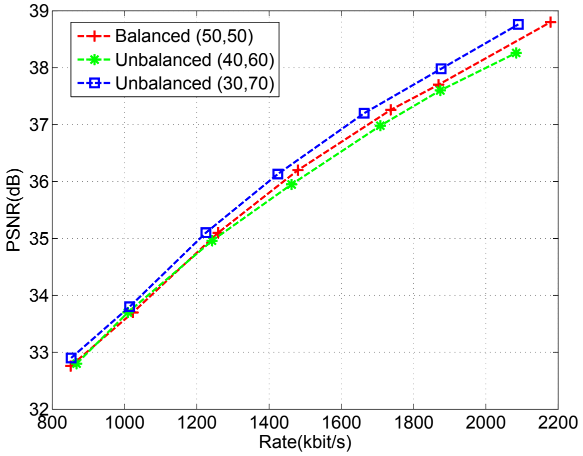
*3) B frames:* The central quantisation values ($QP_0$) for B frames are obtained from interpolation of neighboring anchor frames (I and P) taking into account the corresponding temporal distances. The $QP_0$ absolute difference between adjacent frames is limited to a maximum of 2, in order to guarantee uniform visual quality along the video sequence. The index assignment matrix used in MDSQ of B frames is the same as the one used in the closest anchor.

## V. PERFORMANCE EVALUATION

The performance evaluation of the proposed MDC rate control method is focused on three aspects: i) the R-D performance; ii) the rate control accuracy; iii) performance over lossy channels. The sequences *foreman* and *bus*, CIF resolution, frame rate= 15Hz, GOP=IPBBP... and 16 frames/GOP were used in the simulation. The MDC drift-free close-loop video architecture described in [10] was used to implement the proposed rate control method for unbalanced MDC. A buffer size $B = Channelrate * 1.5$ (in bits) with an initial delay $D = B * 0.8$ equivalent to 1.2 sec was used in these simulations. The coding efficiency of the proposed unbalanced MDC is compared with the balanced MDC and SDC. The overall target bitrate and the target unbalanced rate percentages ($\pi_1$,$\pi_2$) are defined as setting parameters.



(a) *foreman*



(b) *bus*.

Fig. 2. Rate control R-D performance.

### A. R-D Performance

Fig. 2 shows the PSNR obtained from both unbalanced descriptions encoded at different rates. The Figure shows that unbalanced MDSQ has similar overall RD performance in comparison with balanced MDSQ (50,50), showing that coding efficiency of unbalanced MDSQ is mostly equivalent to balanced MDSQ. It is noted that in higher unbalanced rates the rate-distortion performance is improved. In this case the coding efficiency of unbalanced MDSQ is better than balanced MDSQ, because at higher rates the number of zero coefficients in one description is increased while the other carries the full value of a higher number of non-zero ones, i.e., more coefficients are not split between the two descriptions. The same behavior for unbalanced MDC was found in [3], though in a different context.

### B. *Rate Control Accuracy*

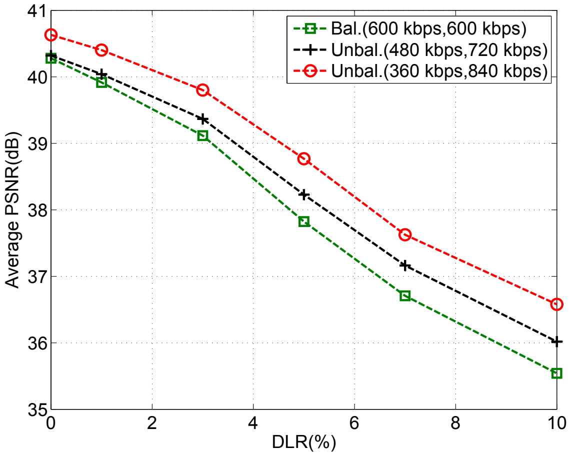
The accuracy of the proposed rate control method was evaluated in order to find the magnitude of deviations from the specified target. Table II shows the rate obtained for both descriptions, i.e., ($R_{D1}$) and ($R_{D2}$), and the unbalanced rate percentages actually achieved ($\pi_1$, $\pi_2$). The results show the achieved bitrate closely following the specified target as well as the unbalanced rate percentages defined for both descriptions. From the results in the table, the maximum target deviation lies between 0% and 6% in one of the two descriptions. Taking into account that the $\rho$ model is only applied to $I$ and $P$ frames, this is considered quite good performance.

TABLE II
RATE CONTROL ACCURACY

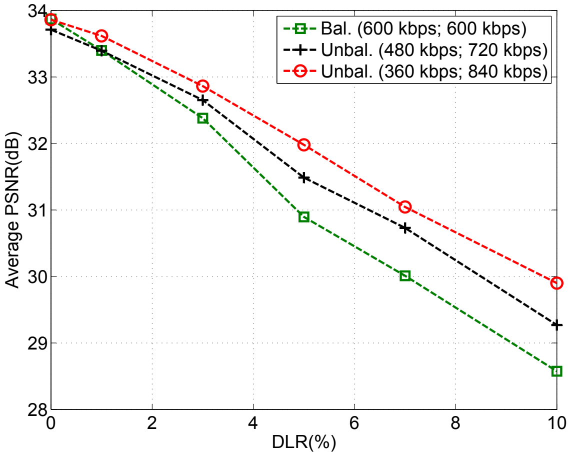
| Target (kbit/s) | Foreman | | | Bus | | |
|---|---|---|---|---|---|---|
| | Target $(\pi_1,\pi_2)$=(50,50) | | | Target $(\pi_1,\pi_2)$=(50,50) | | |
| | $R_{D1}$ | $R_{D2}$ | $(\pi_1,\pi_2)$ | $R_{D1}$ | $R_{D2}$ | $(\pi_1,\pi_2)$ |
| 800 | 399 | 407 | (49, 51) | 427 | 424 | (50,50) |
| 1000 | 501 | 507 | (50, 50) | 525 | 529 | (50,50) |
| 1200 | 608 | 619 | (50,50) | 637 | 642 | (50,50) |
| 1400 | 704 | 718 | (49, 51) | 754 | 762 | (50,50) |
| 1600 | 804 | 820 | (49, 51) | 833 | 842 | (50,50) |
| 1800 | 890 | 908 | (49,51) | 962 | 972 | (50,50) |
| 2000 | 1014 | 1034 | (49,51) | 1045 | 1056 | (50,50) |
| Target (kbit/s) | Target $(\pi_1,\pi_2)$=(40,60) | | | Target $(\pi_1,\pi_2)$=(40,60) | | |
| | $R_{D1}$ | $R_{D2}$ | $(\pi_1,\pi_2)$ | $R_{D1}$ | $R_{D2}$ | $(\pi_1,\pi_2)$ |
| 800 | 303 | 489 | (38, 62) | 326 | 541 | (38,62) |
| 1000 | 381 | 630 | (38, 62) | 391 | 657 | (37,63) |
| 1200 | 447 | 766 | (37, 63) | 470 | 800 | (37,63) |
| 1400 | 514 | 903 | (36, 64) | 558 | 953 | (37,63) |
| 1600 | 579 | 1037 | (36, 64) | 609 | 1049 | (37,63) |
| 1800 | 645 | 1179 | (35, 65) | 679 | 1171 | (37,63) |
| 2000 | 714 | 1313 | (35, 65) | 782 | 1334 | (37,63) |
| Target (kbit/s) | Target $(\pi_1,\pi_2)$=(30,70) | | | Target $(\pi_1,\pi_2)$=(30,70) | | |
| | $R_{D1}$ | $R_{D2}$ | $(\pi_1,\pi_2)$ | $R_{D1}$ | $R_{D2}$ | $(\pi_1,\pi_2)$ |
| 800 | 275 | 533 | (34,66) | 274 | 578 | (32,68) |
| 1000 | 328 | 678 | (33, 67) | 320 | 706 | (31,69) |
| 1200 | 379 | 837 | (31, 69) | 371 | 874 | (30,70) |
| 1400 | 435 | 977 | (31, 69) | 424 | 1043 | (29,71) |
| 1600 | 482 | 1120 | (30, 70) | 456 | 1162 | (28,72) |
| 1800 | 535 | 1296 | (29, 71) | 517 | 1365 | (27,73) |
| 2000 | 565 | 1436 | (29, 71) | 565 | 1476 | (28,72) |

### C. *Streaming over asymmetric channels*

The proposed rate control scheme was also evaluated in packet loss streaming scenarios with different available rates and asymmetric packet loss rates. In this case, for drift compensation the side information is coded using fixed quantisation parameters. The output rate was set to 1 Mbps, using a fixed number of 10 packets per frame with 20% of side redundancy distributed by each description, resulting in an overall rate of 1.2 Mbps. The side information is asymmetrically distributed over the two descriptions such that unbalanced rates are maintained as specified by the target rates of each description. Burst packet loss was simulated using a Gilbert-Elliott 2-state Markov model in order to generate different average packet loss rates (PLR) and mean burst duration [11]. In order to obtain statistically meaningful results, transmission of each sequence (*foreman* and *bus*) was simulated 50 times under the same network conditions. In these simulations, an average PLR between 0% and 10% and average burst length of 4 packets was used.



(a) *foreman* sequence, close-loop MDC.



(b) *bus* sequence, close-loop MDC.

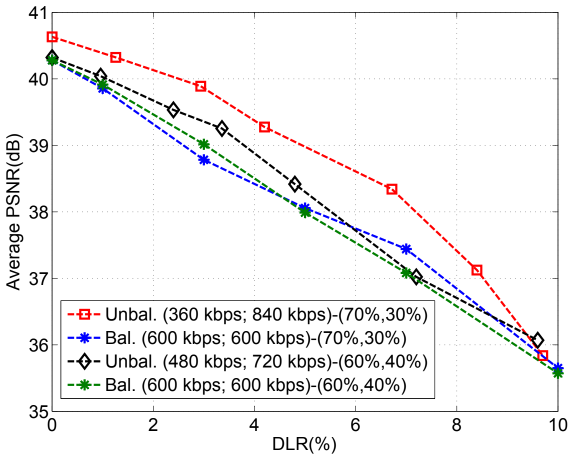
Fig. 3. Average PSNR using channels with the same packet loss ratios.

Two application scenarios are considered for comparison between balanced MDSQ and U-MDSQ: i) channels with the same PLR and ii) channels with different PLR. In both cases, the performance is evaluated by measuring the overall PSNR (i.e., obtained by decoding both descriptions) as a function of the total data loss percentage (DLR) in both descriptions, i.e.,

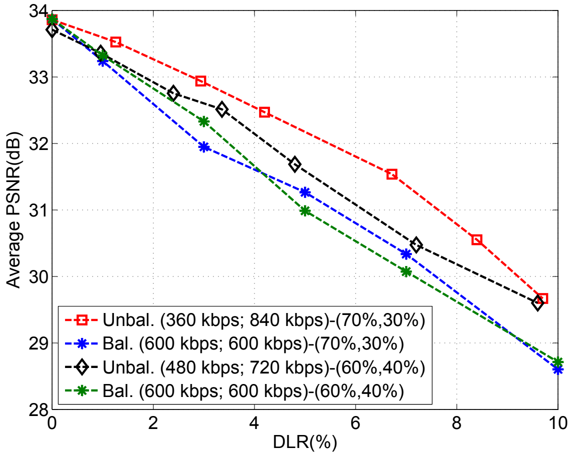
$$DLR(\%) = (1 - Rx\_rate/Tx\_rate) \times 100\%. \quad (8)$$

Note that in U-MDC, different values of PLR may produce the same DLR because the packet size is variable in both descriptions, though the number of packets per frame in each of them is the same.

*1) Channels with the same PLR:* Figs. 3(a) and 3(b) show the performance results using channels with the same PLR. The average PSNR is shown as a function of DLR. The comparison between balanced and U-MDSQ shows that the average PSNR obtained for U-MDSQ is better than balanced MDSQ for the same total rate, i.e., between 0.5dB and 1dB on average. This is explained because better quality in side



(a) *foreman* sequence,close-loop MDC.



(b) *bus* sequence,close-loop MDC.

Fig. 4. Average PSNR using channels with different packet loss ratios.

reconstruction is obtained in U-MDSQ rather than in the balanced case at the same rate. This results show that proposed U-MDSQ scheme is more efficient than the classic balanced one in path diversity applications using channels with different available bandwidths.

*2) Channels with different PLR:* This transmission scenario assumes that each channel has different PLR. To establish equivalent conditions for comparison, each corresponding description of the balanced and unbalanced cases suffer the same percentage of data loss. Note that, as previously pointed out, this is obtained by using different PLR. In unbalanced MDSQ, the higher rate description is transmitted over the channel with better conditions, i.e, lower PLR. Two unbalanced target rates were used: *i*) $R_{D1} = 480kbps; R_{D2} = 720kbps$; *ii*) $R_{D1} = 360kbps; R_{D2} = 840kbps$. The balanced MDSQ used for comparison has the same total rate of 1200 $kbps$, i.e., $R_{D1} = 600kbps; R_{D2} = 600kbps$.

Figs. 4(a) and 4(b) show the average PSNR versus DLR. The comparison is made between the results where the percentage of data loss is the same in each corresponding description, i.e., (70%, 30%) and (60%, 40%). These results show that U-MDSQ also performs better than balanced MDSQ for almost the whole DLR range. The average gains of U-MDSQ are about 1dB over balanced MDSQ. Thus, the results demonstrate that quality improvement is achieved by using the proposed method in path diversity transmission environments with multiple channels, different available bandwidths and PLR.

## VI. CONCLUSIONS

In this paper a new U-MDSQ method for asymmetric channel bandwidths was proposed. The proposed method was shown to be effective in unbalanced MDSQ output rates with high accuracy to a specified target rate for each description without losing R-D performance comparing with balanced case. Evaluation results show that proposed method improves the performance over lossy channels in scenarios with asymmetric bandwidths and packet loss rates, showing the effectiveness of the proposed method. Therefore, the research developed in this works finds application in MDSQ-based video encoders for MD streaming with path diversity based on channels with asymmetric bandwidth and loss constraints.

## ACKNOWLEDGMENT

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## REFERENCES

[1] V. Goyal, “Multiple description coding: Compression meets the network,” *IEEE Signal Processing Magazine*, vol. 18, no. 5, pp. 74–93, September 2001.

[2] J. Noh and B. Girod, “Robust mobile video streaming in a peer-to-peer system,” *Signal Processing: Image Communication*, vol. 27, no. 5, pp. 532 – 544, 2012.

[3] J. Kim, R. Mersereau, and Y. Altunbasak, “Distributed video streaming using multiple description coding and unequal error protection,” *IEEE Transactions on Image Processing*, vol. 14, no. 7, pp. 849 –861, july 2005.

[4] M. Majid and C. Abhayaratne, “Multiple description scalar quantization with successive refinement,” in *Proceedings 17th European Signal Processing Conference (EUSIPCO)*, August 2009.

[5] N. Kamnoonwatana, D. Agraotis, and N. Canagarajah, “Rate controlled redundancy-adaptive multiple description video coding,” *Signal Processing: Image Communication*, vol. 26, no. 45, pp. 205 – 219, 2011.

[6] P. Correia, P. Assuncao, and V. Silva, “Multiple description of coded video for path diversity streaming adaptation,” *IEEE Transactions on Multimedia*, vol. 16, no. 3, pp. 923–935, 6 2012.

[7] Z. He and S. Mitra, “A unified rate-distortion analysis framework for transform coding,” *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 11, no. 12, pp. 1221 –1236, dec 2001.

[8] S. Milani, L. Celetto, and G. A. Mian, “An accurate low-complexity rate control algorithm based on $\rho$-domain,” *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 18, no. 2, pp. 257–262, Feb. 2008.

[9] Z. Li, W. Gao, F. Pan, S. Ma, K. Lim, G. Feng, X. Lin, S. Rahardja, H. Lu, and Y. Lu, “Adaptive rate control for H.264,” *Journal of Visual Communication and Image Representation*, vol. 17, no. 2, pp. 376 – 406, 2006.

[10] P. Correia, P. Assuncao, and V. Silva, “Drift-free multiple description intra video coding,” in *Image Processing (ICIP), 2009 16th IEEE International Conference on*, 7-10 2009, pp. 625 –628.

[11] Z. Li, J. Chakareski, X. Niu, Y. Zhang, and W. Gu, “Modeling of distortion caused by markov-model burst packet losses in video transmission,” in *IEEE International Workshop on Multimedia Signal Processing (MMSP)*, Oct. 2009, pp. 1–6.