Robust Transmission of Multimedia Data over Power-Lines

R. Bernardini*, M. Durigon*, R. Rinaldo*, A. Tonello*, A. Vitali†

*Università degli Studi di Udine - DIEGM
Via delle Scienze 208, Udine, Italy
e-mail: {bernardini,rinaldo,tonello}@uniud.it

†ST Microelectronics
Via C. Olivetti n. 2, 20041, Agrate Brianza - Italy
email: andrea.vitali@st.com

***Abstract*— In this paper we jointly consider the source and channel coding problem with transmission over power line channels. We propose frame based Multiple Description (MD) video coding with an efficient reconstruction algorithm. Further, transmission deploys wide-band impulse modulation with bit interleaved convolutional channel coding. The MD video coder uses a redundant filterbank to generate multiple descriptions of the original video stream, and allows for exact signal reconstruction even in the presence of packet losses. In the presence of packet losses, we adopt a "restoring stage" before the synthesis filterbank whose purpose is to recover the lost coefficients. This solution has limited computational complexity and does not introduce excessive delay in interactive applications. We compare the robustness of the proposed solution with that of another MD solution based on polyphase spatial subsampling assuming wideband impulsive transmission over a power-line link with rate of 1.6 Mbps.**

I. INTRODUCTION

Multimedia transmission over power lines is an interesting low-cost solution for those applications that need to transmit audio/video data. For example, a real time video surveillance system which transmits video over the power line would not require the expensive masonry needed to install video cables.

Unfortunately, video transmission over power lines is not an easy task because of the characteristics of the power-line channel which typically exhibits high error probability and relatively low transmission rates. Moreover, the low-delay requirements of multimedia signals can make unpractical the usual mechanism of acknowledge and packet retransmission.

Multiple Description (MD) coding is a recently proposed solution for increasing the resilience of multimedia transmission to such problems. The idea is to add to the compressed data some redundancy which allows for signal reconstruction at the receiver side even when part of this information is lost. MD coding comprises a very wide range of techniques such as Forward Error Correction (FEC) coding [3], correlating transforms [2], multiple description quantization [1] and redundant basis (frames) [4], [6]. The latter technique analyzes the input signal with a redundant filterbank, producing a set of linearly dependent coefficients. By exploiting the introduced redundancy one can recover the coded signal even in presence of packet loss.

In a sense, frame expansion is similar to MD Forward Error Correction applied "across packets" [3]. The possible advantage in using frames is that added information can be perceptively consistent, so that every description contains information that can help to reconstruct or approximate the original sequence. On the other hand, an error correcting code is helpful only if a certain minimum amount of information is correctly received, and if no exact recovery is possible, "parity check" packets are not useful.

In this paper we analyze by means of simulations the performance of a simple frame-based scheme for video transmission over power-line channels. In the proposed scheme the input signal is separated into its spatial polyphase components and an additional description, which acts like a visually consistent "parity check" sequence, is obtained by low-pass filtering and subsampling the original signal. The three descriptions are coded using independent H264 / AVC video coders [8].

The MD coded streams are transmitted over a power line (PL) link using an impulse modulated based transmission scheme. In particular, packets of bits generated by the MD coder are first convolutionally encoded and interleaved. Then, the coded bits are sent using an impulse modulator combined with direct sequence data spreading (DS-CDMA) [12]. At the receiver, first detection is accomplished, then after de-interleaving Viterbi decoding is performed to reconstruct the packets generated by the MD source coder, i.e. packets of MD coefficients.

When the channel decoded packets are received without errors, the source coded signal is reconstructed by means of the *dual* synthesis filterbank. In case of packet loss the signal is reconstructed by means of the algorithm of [7]. If too many errors happen and exact reconstruction is not possible, error concealment based on bilinear interpolation is used. After error recovery or concealment, frames are copied onto the decoders' frame buffers at the receiver, in order to mitigate the effect of error propagation due to differential coding.

We show experimental results for the proposed frame based coding and transmission scheme for relatively low transmission rates. In particular we evaluate the performance of the proposed scheme comparing it with another MD scheme and with standard Single Description H264 coding, assuming a transmission rate over the PL link of up to 1.6 Mbps.

II. MD CODING WITH FRAMES

In order to make this paper self-contained, in this section we briefly summarize the algorithm of [7].

A family of signals $\Phi = \{\phi_k \in \ell^2(Z)\}_{k\in Z}$ constitutes a frame if for any signal $x \in \ell^2(Z)$ there exist two constants

Proceedings of the 9th International Symposium on Power-Line Communications and its Applications (*ISPLC 2005*)
April 6-8, 2005, Vancouver, Canada
ISBN 0-7803-8844-5 © IEEE
pp. 295-299

$A > 0$ and $B < \infty$ such that

$$A\|x\|^2 \leq \sum_{k=1}^{N} |\langle x, \phi_k\rangle|^2 \leq B\|x\|^2, \quad (1)$$

where $\langle f, g\rangle = \sum_n f[n]g^*[n]$ is the scalar product between f and g. In particular, the left-hand inequality guaranties that it is possible to reconstruct (in a robust way) the original signal x from the scalar products $y_k = \langle x, \phi_k\rangle$ and that it is possible to compute a set of *dual* signals $\tilde{\phi}_k$ such that

$$x = \sum_k \langle x, \phi_k\rangle \tilde{\phi}_k = \sum_k \langle x, \tilde{\phi}_k\rangle \phi_k. \quad (2)$$

In the framework of oversampled filterbanks, one computes a vector of output coefficients in each channel $c = 1, ..., N$, via convolution, i.e.,

$$y_c[n] = \sum_{m\in Z} x(m) h_c[Mn - m]. \quad (3)$$

The right-hand side of Eq. (3) can be interpreted as the scalar product between the input and the analysis vector $\phi_k \triangleq h_c[Mn - \cdot]$. By appropriate filter design, the ϕ_k constitute a frame. Oversampling, i.e., choosing $N > M$, implies that there is redundancy in the coefficients $y_c[n]$ which can be exploited to reconstruct x even if some coefficients are lost.

For finite dimensional signals and FIR analysis filters, one can collect the input and the filter output coefficients in vectors $\mathbf{x}$ and $\mathbf{y}$, respectively, and express the filtering operation via a matrix product $\mathbf{y} = \mathbf{F}\mathbf{x}$. One can recognize that the rows of matrix $\mathbf{F}$ correspond to ϕ_k and that they are the time-reversed and translated impulse responses of the analysis filters. Note that, because of oversampling, matrix $\mathbf{F}$ is a rectangular matrix with more rows than columns.

One can reconstruct $\mathbf{x}$ from $\mathbf{y}$ by means of the pseudo-inverse $\mathbf{F}^\dagger$ of matrix $\mathbf{F}$, i.e., $\mathbf{x} = \mathbf{F}^\dagger\mathbf{y}$. Considering (2), it turns out that the columns of $\mathbf{F}^\dagger$ correspond to the dual frame elements $\tilde{\phi}_k$. It is important to note that reconstruction via the dual frame is optimal even if $\mathbf{y}$ does not belong to Im($\mathbf{F}$). This is usually the case in applications where the coefficients in $\mathbf{y}$ are quantized into $\hat{\mathbf{y}}$ before coding and transmission. In such a case, the vector $\hat{\mathbf{x}} = \mathbf{F}^\dagger\hat{\mathbf{y}}$ is the minimum norm vector minimizing the squared error $||\mathbf{F}\hat{\mathbf{x}} - \hat{\mathbf{y}}||^2$, i.e., it is the one best describing the received coefficients.

In case of coefficient loss, one can pretend that the input was analyzed with a subset $\Phi_I = \{\phi_k\}_{k\in I}$ of the analysis functions. If the corresponding matrix $\mathbf{F}_I$ is still full rank, set Φ_I is indeed a frame and the input $\mathbf{x}$ can be recovered from the set of coefficients $\mathbf{y}_I$, namely $\mathbf{x} = \mathbf{F}_I^\dagger\mathbf{y}_I$. At the receiver, therefore, one needs to compute the pseudo-inverse of the original analysis matrix after some row cancellation, corresponding to the coefficient loss pattern. It turns out, unfortunately, that the pseudo-inverse operator does not have a filterbank structure anymore. Moreover, direct computation can be computationally demanding, due to the large dimensions of the involved matrix. In [7], an algorithm for the evaluation of the pseudo-inverse $\mathbf{F}_I^\dagger$ is presented. It puts in front of the original synthesis filterbank a "restoring" stage which recovers the lost coefficients from the received ones. For the sake of reference, we now briefly recall the results of [7].

If we denote with I^c the set of lost coefficients, one can write from (2)

$$\phi_k = \sum_{n\in Z} \phi_n\langle\phi_k, \tilde{\phi}_n\rangle = \sum_{n\notin I^c} \phi_n\langle\phi_k, \tilde{\phi}_n\rangle + \sum_{m\in I^c} \phi_m\langle\phi_k, \tilde{\phi}_m\rangle.$$

Taking the scalar product with x, we have

$$y_k = \langle x, \phi_k\rangle = \sum_{n\notin I^c} y_n\langle\tilde{\phi}_n, \phi_k\rangle + \sum_{m\in I^c} y_m\langle\tilde{\phi}_m, \phi_k\rangle.$$

In matrix form,

$$\boldsymbol{y}_m = \boldsymbol{M}\boldsymbol{y}_m + \boldsymbol{M}'\boldsymbol{y}_r$$

where $\boldsymbol{y}_m$ is the vector of *lost* coefficients and $\boldsymbol{y}_r$ the set of *received* coefficients. The lost coefficients can therefore be recovered by $\boldsymbol{y}_m = (\boldsymbol{I} - \boldsymbol{M})^{-1}\boldsymbol{M}'\boldsymbol{y}_r$. It is possible to show that using the restored coefficients as the input to the original dual synthesis filterbank is equivalent to the application of the pseudo-inverse $\mathbf{F}_I^\dagger$. Moreover, if the original analysis and synthesis filterbanks are FIR, matrix $(\boldsymbol{I} - \boldsymbol{M})$ has a block structure, and each block inversion can be computed as soon as the block is available, thus reducing the delay and the complexity of the procedure [7]. In the presence of excessive errors, it may happen that the resulting set Φ_I becomes incomplete. In this case, one could use the MSE optimal approximation $\boldsymbol{y}_m = (\boldsymbol{I} - \boldsymbol{M})^\dagger\boldsymbol{M}'\boldsymbol{y}_r$ [7]. In any case, the algorithm in [7] allows to recognize which parts of the input x could not be recovered.

III. Application of MD video coding over PL channels

We consider the application of the proposed MD coding approach for transmission of video streams over a power line (PL) link. In Section III-A we report the details about the source coder. In Section III-B we report the details of the transmission scheme that has been used to assess the overall performance over a power line channel.

A. MD coding for transmission of video streams

The results outlined in Section II are used for the design of a simplified frame based MD video coder. The descriptions are generated using a one-dimensional filterbank applied to columns of every sequence frame. The filter outputs are subsampled by a factor of 2. The analysis filters separate even and odd rows, i.e., the filter impulse responses are $h_0(n) = \delta_n$, $h_1(n) = \delta_{n+1}$. An additional low-pass filter $h_2(n)$ generates the third description. Thus, for an $N_r \times N_c$ input frame, the scheme originates 3 descriptions with dimension $N_r/2 \times N_c$ pixels. The third filter should be FIR, and orthogonal to its factor 2 translations. This condition grants that the synthesis filterbank is FIR too. In choosing $h_2(n)$ it is also important to evaluate the effects of the synthesis filterbank on the quantization error introduced by video coders [10]. To minimize the effects of quantization error on reconstructed video, $h_2(n)$

TABLE I

ANALYSIS FILTERBANKS FOR THE SYMMETRIC 4-TAP REDUNDANT CHANNEL FILTER.

	Symmetric 4 tap		
n	$h_0(n)$	$h_1(n)$	$h_2(n)$
-2	0	0	$-1.04 \cdot 10^{-1}$
-1	0	1	$+5.77 \cdot 10^{-1}$
0	1	0	$+5.77 \cdot 10^{-1}$
1	0	0	$-1.04 \cdot 10^{-1}$

can be chosen such that the synthesis filterbank is not FIR. However, if the obtained IIR synthesis filter impulse responses rapidly decay toward zero, they can be well approximated with FIR filters. The proposed $h_2(n)$ is a 4-tap linear-phase filter (Table I), whose corresponding synthesis filterbank is approximated with FIR filters (Table II). The use of a linear phase filter can be useful for video coding purposes, mainly because it allows for symmetric rather than periodic signal extension in the analysis stage.

TABLE II

SYNTHESIS FILTERBANKS FOR THE SYMMETRIC 4-TAP REDUNDANT CHANNEL FILTER.

	Symmetric 4 tap		
n	$\tilde{h}_0(n)$	$\tilde{h}_1(n)$	$\tilde{h}_2(n)$
-6	$+1.09 \cdot 10^{-4}$	0	$+1.24 \cdot 10^{-4}$
-5	$-1.68 \cdot 10^{-4}$	$+1.09 \cdot 10^{-4}$	$-1.91 \cdot 10^{-4}$
-4	$+1.52 \cdot 10^{-3}$	$-9.89 \cdot 10^{-4}$	$+1.74 \cdot 10^{-3}$
-3	$-2.35 \cdot 10^{-2}$	$+1.52 \cdot 10^{-3}$	$-2.68 \cdot 10^{-3}$
-2	$+2.14 \cdot 10^{-2}$	$-1.39 \cdot 10^{-2}$	$+2.43 \cdot 10^{-2}$
-1	$+5.70 \cdot 10^{-2}$	$+2.14 \cdot 10^{-2}$	$-3.75 \cdot 10^{-2}$
0	$+7.99 \cdot 10^{-1}$	$-1.94 \cdot 10^{-1}$	$+3.41 \cdot 10^{-1}$
1	$-1.94 \cdot 10^{-1}$	$+7.99 \cdot 10^{-1}$	$+3.41 \cdot 10^{-1}$
2	$+2.14 \cdot 10^{-2}$	$+5.70 \cdot 10^{-2}$	$-3.75 \cdot 10^{-2}$
3	$-1.391 \cdot 10^{-2}$	$+2.14 \cdot 10^{-2}$	$+2.43 \cdot 10^{-2}$
4	$+1.52 \cdot 10^{-3}$	$-2.35 \cdot 10^{-3}$	$-2.68 \cdot 10^{-3}$
5	$-9.89 \cdot 10^{-4}$	$+1.52 \cdot 10^{-3}$	$+1.74 \cdot 10^{-3}$
6	$+1.09 \cdot 10^{-4}$	$-1.68 \cdot 10^{-4}$	$-1.92 \cdot 10^{-4}$
7	0	$+1.09 \cdot 10^{-4}$	$+1.24 \cdot 10^{-4}$

The descriptions are coded using three independent H264 / AVC standard coders, as shown in Fig. 1. The H264 / AVC coder divides the input frame into slices made from macroblocks (MB) organized into rows. Each slice is then sent over the network in a single packet. The reason for subsampling along the columns is that the loss of one slice results in a limited number of contiguous lost coefficients in each column, hopefully permitting error recovery.

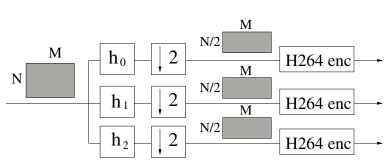


Fig. 1. Block diagram of the proposed frame based MD coder.

On the receiver side (Figure 2), the video streams are independently processed by H264 / AVC synchronized decoders. These decoders are connected to a restoring block **R**, which recovers channel errors by implementing the algorithm outlined above. We remark that this solution needs only local information and does not introduce any relevant delay in the decoding process. In the case of unrecoverable errors, corresponding to an incomplete Φ_I, missing regions in lost descriptions are reconstructed using bilinear interpolation from the received ones. This still gives acceptable results, due to the high spatial correlation among descriptions in the proposed scheme. We found that this is indeed preferable to the use of the MSE optimal approximation $\boldsymbol{y}_m = (\boldsymbol{I} - \boldsymbol{M})^\dagger \boldsymbol{M}' \boldsymbol{y}_r$.

The result is a set of three recovered subframes with dimension $N_r/2 \times N_c$. These subframes are then fed into the synthesis filterbank, whose output is the original full size sequence. Recovered subframes for each description are copied into the corresponding decoder frame buffer, in order to prevent error propagation from reference frames due to interframe coding.

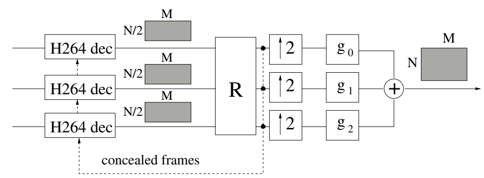


Fig. 2. Block diagram of the proposed decoder.

B. PL impulse modulated transmission scheme

In order to test the MD coding scheme over a power line (PL) link we consider a transmission scheme based on impulse modulation as described in what follows. Details can be found in [12] while a block diagram of the transmission scheme is shown in Figure 3.

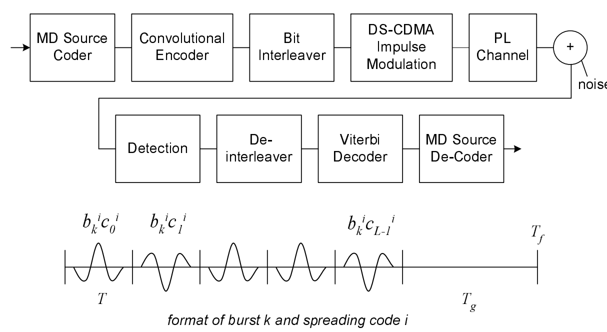


Fig. 3. Block diagram of the whole transmission scheme.

The stream of bits at the output of the video coder are packetized into packets of length 1918 bits. Each packet is coded with a tail terminated convolutional code of rate 1/2 and memory 4. The coded packet, of length 3844 bits, is interleaved with a matrix interleaver. Then, the interleaved bits are transmitted using an impulse modulation scheme combined with direct sequence code division multiple access (DS-CDMA) that uses Walsh Hadamard codes $\{c_m^i\}$ $i, m = 0, ., L-1$ of length $L = 32$. That is, a block of L bits ($L-1$ coded bits plus 1 training/signaling bit) are orthogonally modulated using Walsh-Hadamard spreading of size L to obtain a block of L symbols $\{b_k^i\}$, $i = 0, ., L-1$ that have alphabet ± 1. The block of symbols is pulse shaped with a short duration pulse. The transmitted signal (burst) that is associated to the block of L bits has duration LT where T is the chip period (Figure 3). We add a guard time T_g to cope with the channel time dispersion such that the raw transmission rate equals L/T_f with $T_f = LT + T_g$. In particular in the simulations we have assumed $T = 151$ ns and $T_f = 9.67$ μs which results in a raw date rate of about 3.3 Mbps. With convolutional coding the net transmission rate equals 1.6 Mbps.

It should be noted that adaptive rate transmission is possible by lowering the number of Walsh codes (bits) that are simultaneously used in each burst transmission. Further higher transmission rates can be achieved by choosing different parameters, e. g. longer spreading codes and shorter monocycles.

In order to evaluate performance over an ensemble of network topologies we have assumed a statistical channel model that can be considered representative of typical indoor PL channels (details can be found in [12]). The impulse responses are assumed to be shorter than 4 μs. The background noise is assumed white Gaussian. The receiver algorithm uses a frequency domain approach with successive interference cancellation. Successive interference cancellation is used to cancel the inter-code interference that is generated by the highly dispersive channel [12]. Channel decoding uses a soft Viterbi algorithm. The stream of bits at the output of the Viterbi decoder are mapped back into packets of source coder coefficients.

IV. EXPERIMENTAL RESULTS

We evaluate the performance of the proposed video coding scheme presented in Section III (referred to as MD3 scheme), where the redundant analysis filterbank introduced in the previous section is used.

It is compared with another MD system based on spatial subsampling of original video pictures (MD4 scheme). This system originates four descriptions by spatially subsampling by a factor of 2 both along rows and along columns each video picture. Each description, whose dimension is 1/4 of that of the original frame, is compressed independently, packetized and sent over an error prone network. Lost information must be recovered via interpolation, using the high spatial correlation between descriptions. Different interpolation techniques can be adopted to restore lost information [11]: both usual linear interpolators and non-linear interpolators (CALIC like) which try to replicate natural edges inside video pictures.

We compare the MD schemes with a standard Single Description H264 / AVC codec which includes enhanced error concealment capabilities with respect to the standard decoder described in [9]. These improvements are mainly due to the application of interframe concealment techniques on intra coded information, and on detecting scene changes. The codec is derived from the H264 / AVC test model software version *JM6.0a*.

Coding options are the same for the SD and MD coders. In particular the GOP structure is $IBBPBBPBBPBBPBBI$, and slices have a fixed 240 bytes dimension. Each slice is sent as a packet, and each packet is transmitted over the power line channel with the transmission and channel coding scheme described above. Similarly to the MD3 scheme, MD4 corrected subpictures are copied into the corresponding decoder frame buffers. In case all the descriptions are lost, H264 / AVC error concealment capabilities are applied in all the MD decoders.

We consider 240 frames (8 seconds) of two CIF video sequences at 30 frames per second. The *Foreman* sequence, which is characterized by high motion content, and the *News* sequence which has low motion content and a static background with a lot of details. Every simulation point is the average of 50 independent transmission trials.

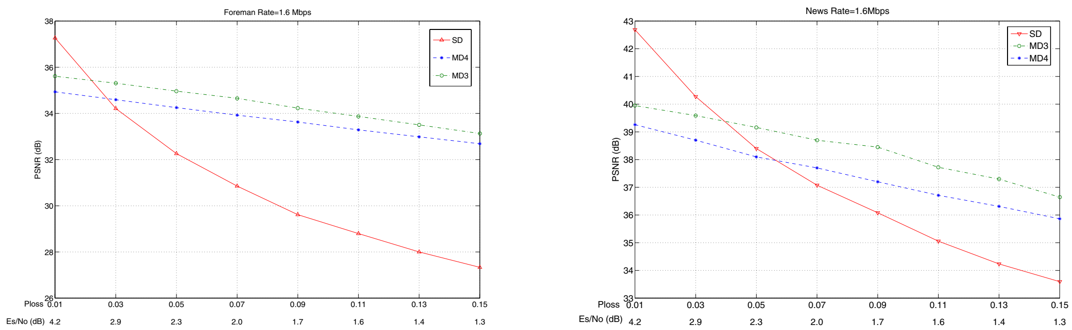
In Figure 4 we report PSNR as a function of both packet loss (computed over packets of 1918 information bits at the output of the Viterbi decoder) and as a function of the channel signal-to-noise ratio (E_s/N_0). The PSNR represents the peak SNR computed over the reconstructed video stream. E_s/N_0 is the channel signal-to-noise ratio that allows to achieve the desired packet error rate by the impulsive transmission scheme.

Sequences are compressed at 1.6 Mbps, in order to exactly match the transmission rate that is supported by the physical layer.

The SD coder can exploit spatial redundancy more efficiently, it has the best performance in terms of mean PSNR at low P_{loss} (for $P_{loss} = 0.01$ it gains 2.5-3.5 dB with respect to the MD4 scheme). Nonetheless, the performance of the SD codec drops very rapidly for increasing P_{loss}. We expect that MD codecs present good robustness to errors at the expense of some coding inefficiency. The proposed frame based codec (MD3) adds 1.5 redundancy to the video stream, and has therefore a low coding efficiency but possibly good robustness to errors both from a subjective and objective quality point of view. The MD4 scheme codes four subsampled video sequences and its coding inefficiency is bigger than that of the MD3 proposed scheme.

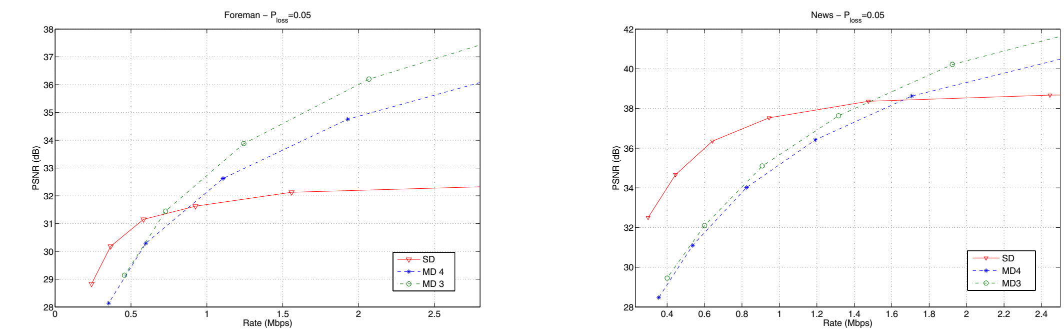
From Figure 4 we can see that SD performance rapidly drops for increasing P_{loss}. In terms of mean PSNR the MD3 scheme becomes better than the SD for $P_{loss} > 0.02$ and $P_{loss} > 0.04$, for the Foreman sequence and the News sequence respectively. In both cases the MD3 scheme gains about 1 dB with respect to the MD4 scheme, and this gain decreases for increasing P_{loss}. In fact, increasing the number of lost packets, the set of received coefficients is no longer a frame, and the perfect reconstruction property of the MD3 scheme cannot be exploited. For the News sequence the SD scheme has acceptable performance for relative high P_{loss}. This is due to the static background of this sequence, that makes the H264 / AVC enhanced concealment strategy very effective.

Figure 5 shows the reconstructed video quality (in terms of mean PSNR) for different compression rates and for a relative small packet loss probability $P_{loss} = 0.05$. Note that herein we assume transmission rates beyond 1.6 Mbps. As we said before, such rates can be achieved by choosing a different set of parameters for the physical layer. For the high motion Foreman sequence, the SD PSNR rapidly saturates due to the difficulties in error concealment. For the News



(a) (b)

Fig. 4. Mean PSNR for (a) the Foreman sequence and (b) the News sequence, for different P_{loss}. Sequences are coded at 1.6 Mbps.



(a) (b)

Fig. 5. Rate-distortion comparison of SD and MD schemes for different values of P_{loss}.

sequence, the SD scheme has good performance in terms of mean PSNR. However, besides mean PSNR the visual quality of reconstructed video is higher and more uniform for MD schemes than that for the SD scheme. In particular the SD reconstructed sequences present annoying artifacts in correspondence to motion objects that belong to spatial regions affected by channel erasures. The MD3 scheme has higher PSNR than MD4 scheme, and this gain increases at high bit rates where the perfect reconstruction property of MD3 allows for significant quality improvements over the approximate interpolation methods for error concealment used by MD4.

V. Conclusions

In this paper we have jointly considered the source and channel coding problem with transmission over power line channels. We have proposed frame based MD coding with efficient reconstruction algorithm. Further, transmission deploys wideband impulse modulation with bit interleaved convolutional channel coding that supports rates up to 1.6 Mbps.

References

[1] S. D. Servetto, V. A. Vaishampayan, and N. J. A. Sloane, "Multiple description lattice vector quantization," in *Proc. IEEE Data Compression Conference*, (Snowbird, UT), pp. 13–22, IEEE, Mar. 1999.

[2] Y. Wang, M. T. Orchard, and A. R. Reibman, "Multiple description image coding for noisy channels by pairing transform coefficients," in *Proc. IEEE Workshop on Multimedia Signal Processing*, (Princenton, NJ), pp. 419–424, June 1997.

[3] A. E. Mohr, E. A. Riskin, and R. E. Ladner, "Unequal loss protection: Graceful degradation of image quality over packet erasure channel through forward error correction," *IEEE Journal on Selected Areas in Communications*, vol. 18, pp. 819–828, June 2000.

[4] V. K. Goyal, M. Vetterli, and N. T. Thao, "Quantized overcomplete expansions in R^N: Analysis, synthesis and algorithms," *IEEE Transactions on Information Theory*, vol. 44, pp. 16–31, Jan. 1998.

[5] X. Yang and K. Ramchandran, "Optimal subband filter banks for multiple description coding," *IEEE Transactions on Information Theory*, vol. 46, pp. 2477–2490, Nov. 2000.

[6] P. Dragotti, J. Kovačević, and V. K. Goyal, "Quantized oversampled filter banks with erasures," in *Proc. IEEE Data Compression Conference*, (Snowbird, UT), pp. 173–182, Mar. 2001.

[7] R. Bernardini, M. Durigon, and R. Rinaldo, "Low-delay reconstruction of punctured frame-coded streams," in *Asilomar Conference on Signals, Systems, and Computers*, (Pacific Grove, CA), Nov. 2003.

[8] Joint Video Team of ITU-T and ISO/IEC JTC 1, "Draft ITU-T Recommendation and Final Draft International Standard of Joint Video Specification (ITU-T Rec. H.264 — ISO/IEC 14496-10 AVC)." Joint Video Team (JVT) of ISO/IEC MPEG and ITU-T VCEG, JVT-G050, Mar. 2003.

[9] T. Stockhammer, M. M. Hannuksela, and T. Wiegand, "H.264/AVC in wireless environments," *IEEE Transactions on Circuits and Systems for Video Technology*, vol. 13, no. 7, 2003.

[10] R. Bernardini, M. Durigon, R. Rinaldo, A. Vitali, P. Zontone, "Extended orthogonal filterbanks for robust video transmission," *submitted for publication.*

[11] R. Bernardini, M. Durigon, R. Rinaldo, L. Celetto, P. Zontone, "Polyphase spatial subsampling multiple description coding of video streams with H264," *IEEE International Conference on Image Processing*, Oct. 2004.

[12] A. Tonello, "An Impulse Modulation Based PLC System with Frequency Domain Receiver Processing", *Proc. of IEEE ISPLC 2005*, Vancouver, Canada, April 2005.