

NOVEL ITERATIVE MULTIPLE DESCRIPTION CODING FOR CORRELATED SOURCES

Laurent Schmalen[†], Matthias Tschauner^{*}, Tobias Breddermann^{*}, and Peter Vary^{*}

^{*} Institute of Communication Systems and Data Processing, RWTH Aachen University, Aachen, Germany

[†] now with Alcatel-Lucent Bell Labs, Stuttgart, Germany

Email: {schmalen,vary,breddermann}@ind.rwth-aachen.de

Web: www.ind.rwth-aachen.de

ABSTRACT

In this paper, we study a novel multiple description coding approach which uses a convolutional code to generate the individual descriptions. Contrary to most conventional multiple description schemes, which attempt to partially reconstruct the signal in the presence of a packet loss, we are interested in the question, whether the signal can be completely reconstructed at the receiver if one description is missing by exploiting the residual source correlation. Usually, audio-visual source encoders for digital mobile communications extract parameters that – due to delay and complexity constraints – exhibit some residual redundancy. This residual redundancy is exploited in a multiple description receiver by performing iterative source-channel decoding (ISCD). The source correlation required for near perfect reconstruction in case of a loss of one description is analyzed by means of EXIT charts and simulation results show the superior performance of the new approach.

1. INTRODUCTION

Multiple Description Coding (MDC) [1, 2] is a tool to generate two (or more) descriptions of a signal which are then independently transmitted over a network with possible packet losses. If all descriptions are correctly received, the signal can be reconstructed with the best possible quality. If one or more descriptions of the signal are missing due to packet losses, the signal can still be reconstructed, however, with degraded overall quality.

Multiple description coding can also be used for a more general kind of hierarchical coding: due to bottlenecks in the network, parts of the packets may be rejected, thus allowing a flexible rate adaptation. One example of a speech and audio codec employing MDC is the *FlexCode* source coder [3]. Multiple description codes are generally quantified by their *index assignment* [1] which maps a central code book index to two or more side code book indices. The set of side code book indices form the individual descriptions.

Residual redundancy of source codec parameters such as scale factors or predictor coefficients for speech, audio, and video signals, occurs due to imperfect source encoding resulting for instance from delay and complexity constraints. This redundancy can be utilized by a *soft decision source decoder* (SDSD) [4] at the receiver to improve the reconstruction quality. *Iterative source-channel decoding* (ISCD) [5, 6] is an extension of SDSD and exchanges in an iterative process so-called *extrinsic* reliabilities between an SDSD and a channel decoder.

Approaches to utilize soft information in the decoding of multiple descriptions can be found in, e.g., [7], [8]. The concept presented in [7] uses the inherent redundancy of multiple descriptions to improve the decoding performance. This approach is then also extended to realize a Turbo-like transmission scheme with iterative decoding in [7].

In this paper, we present a new MDC approach based on ISCD. In this case, the different descriptions are not generated by a *Multiple Description Index Assignment* (MDIA), as proposed in [1] but by a convolutional channel code. The resulting system is denoted *Multiple Descriptions by Channel Coding* (MDCC). Our investigations will disclose that this methodology permits to fully reconstruct

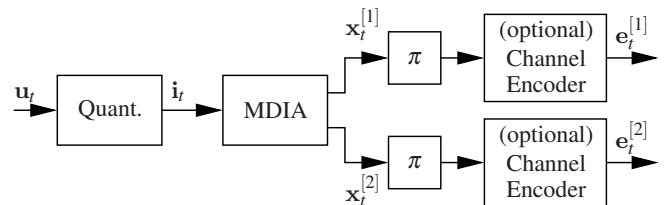


Figure 1: Transmitter of a conventional multiple description coding scheme according to [1] including multiple description index assignment (MDIA) and optional channel coding.

the signal if a description has been lost, as long as the source shows some minimum residual correlation. Simulation results reveal that this novel system is able to outperform the conventional multiple description techniques, even in the presence of additive channel noise.

2. CONVENTIONAL MULTIPLE DESCRIPTION CODING

Figure 1 depicts the base band transmitter block diagram of a conventional MDC scheme with two descriptions. It has been found that in the context of audio and speech transmission, two descriptions are generally sufficient in the most interesting range of conditions [9]. A frame \mathbf{u}_t consisting of N_t unquantized source codec parameters $u_{t,k}$ is quantized using a Q -level scalar quantizer $\mathcal{Q} : \mathbb{R} \rightarrow \mathbb{N}_1$ which maps the input parameter $u_{t,k}$ to a quantizer index $i_{t,k}$, denoting the selected entry of the quantizer code book $\mathbb{V} = \{\bar{v}^{(1)}, \dots, \bar{v}^{(Q)}\} \subset \mathbb{R}$. The set \mathbb{V} denotes the central quantizer code book. All quantizer indices of a frame are grouped to the vector $\mathbf{i}_t = (i_{t,1}, \dots, i_{t,N_t})$.

The *multiple description index assignment* (MDIA) generates two descriptions of the quantizer index $i_{t,k}$ according to the method proposed in [1]. The resulting indices are denoted by $i_{t,k}^{[v]} = \mathcal{D}_v(i_{t,k})$, with $v \in \{1; 2\}$ indicating the description. The indices $i_{t,k}^{[v]} \in \mathbb{I}^{[v]} \subset \mathbb{N}_1, v \in \{1, 2\}$, can be considered as indices of so-called *side quantizers* (utilizing potentially smaller code books) and are commonly denoted *side indices*. The side quantizers can be considered as quantizers with $Q^{[v]}$ code book entries. To each $i_{t,k}^{[v]}$, a bit pattern $\mathbf{b}_{t,k}^{[v]} = \mathcal{B}^{[v]}(i_{t,k}^{[v]})$, consisting of $B^{[v]}$ bits, is assigned. The bit patterns are selected from a set $\mathbb{B}_k^{[v]} = \{\bar{\mathbf{b}}_k^{[v](1)}, \dots, \bar{\mathbf{b}}_k^{[v](Q^{[v]})}\}$. The bit patterns of the individual descriptions are grouped to a bit vector $\mathbf{x}_t^{[v]}$ and are optionally interleaved and channel encoded (if bit errors are expected on the transmission link). Each of the channel encoded vectors $\mathbf{e}_t^{[v]}$ contains $N_E^{[v]}$ entries.

The MDIA shall be illustrated by means of an introductory example using the MDIA illustrated in Fig. 2. The quantizer indices are arranged in a matrix according to a certain structure. If the quantizer selects for instance the index $i = 12$, the MDIA outputs $i^{[1]} = 4$

$\bar{\mathbf{b}}^{1}$	000	001	011	010	110	111	101	$\bar{\mathbf{b}}^{[2](8)}$	
	000	1	2						
	001	3	4	6					
	011		5	7	8				
	010			9	10	12			
	110				11	13	14		
	111					15	16	18	
	101						17	19	20
	100							21	22

Figure 2: Nested multiple description index assignment with 3 filled diagonals optimized for a packet loss rate of $\varepsilon = 0.05$ according to [10], $Q = 22$, $Q^{[1]} = Q^{[2]} = 8$, and non-redundant Gray bit mapping $\mathcal{B}_k^{[v]}$ (i.e., $B_k^{[v]} = 3$, $v \in \{1, 2\}$).

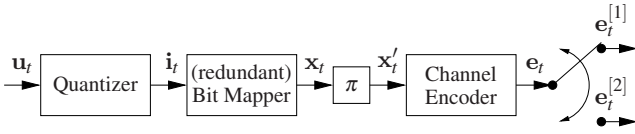


Figure 3: Base band model for the transmitter of the novel MDCC scheme.

(corresponding to the fourth row) and $i^{[2]} = 5$ (corresponding to the fifth column). This leads to the bit patterns $\bar{\mathbf{b}}^{[1](4)} = (010)$ for the first description and $\bar{\mathbf{b}}^{[2](5)} = (110)$ for the second description.

Each description is independently transmitted over a packet-erasure AWGN channel. The single bipolar values of the encoded vector $\mathbf{e}_t^{[v]}$ (each with symbol energy $E_s = 1$) may be subject to AWGN with power spectral density $\sigma_n^2 = N_0/2$. As the goal of this paper is to show design methodologies and guidelines that are independent of the transmission channel and thus applicable also to other channel models and modulation schemes, we consider non-fading AWGN channels (or block-fading, i.e., the fading remains constant for a complete frame) with BPSK mapping by way of example. The application of ISCD to other (fading) channel models, including higher order modulation and channel equalization, has been demonstrated in, e.g., [11, 12].

Additionally, each complete packet $\mathbf{e}_t^{[1]}$ may be erased with probability ε . This channel models, e.g., the packet transmission over a wireless link: due to network congestion or synchronization issues, the complete packet may be lost. If the packet is received, the receiver noise is modeled by AWGN. At the receiver, channel decoding (if necessary) is performed using a *soft-input, soft-output* (SISO) channel decoder. Note that if a description is lost, no channel-related information for the description is available. Several optimized MDC decoding algorithms taking into account the redundancy introduced by the MDIA as well as the residual source redundancy exist. These include cross decoding [7], MMSE or MAP parameter estimation [13, 14, 15].

3. NOVEL MDC BY CHANNEL CODING

In this section, an alternative approach to MDC is studied. This approach, denoted *Multiple Descriptions by Channel Coding* (MDCC), shall mainly be designed for error-free channel conditions (packet losses only), but its performance will also be studied in an AWGN environment. The transmitter of the proposed approach is depicted in Fig. 3. The main difference compared to the transmitter given in Fig. 1 is that no MDIA exists. As in conventional single descriptions transmission systems, a bit mapper \mathcal{B} assigns a unique

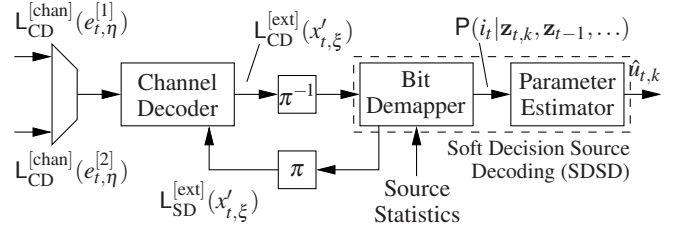


Figure 4: Baseband model of the MDCC receiver based on ISCD.

bit pattern $\mathbf{b}_{t,k} \in \mathbb{B} = \{\bar{\mathbf{b}}^{(1)}, \dots, \bar{\mathbf{b}}^{(Q)}\} \subseteq \mathbb{F}_2^B$ of B bits to each quantizer index $i_{t,k}$. The single bits of the bit pattern $\mathbf{b}_{t,k}$ are denoted by $b_{t,k,\mu} \in \mathbb{F}_2$, with $\mu \in \{1, \dots, B\} \subset \mathbb{N}_1$ denoting the μ th entry of $\mathbf{b}_{t,k}$. If $B > \log_2 Q$, the bit mapping is called redundant, as it introduces artificial redundancy: more bits than actually necessary are spent to represent a quantizer index.

After the bit mapping, the N_I individual bit patterns $\mathbf{b}_{t,k}$ are grouped to a bit vector $\mathbf{x}_t \doteq (\mathbf{b}_{t,1}, \dots, \mathbf{b}_{t,N_I}) = (x_{t,1}, \dots, x_{t,\xi}, \dots, x_{t,N_X})$. The size of the bit vector is $N_X \doteq N_I B$. As the bit mapping is considered to be a code, the rate of the bit mapping is defined by

$$r_{\text{BM}} = \frac{N_I \cdot \text{ld} Q}{N_X} = \frac{\text{ld} Q}{B}. \quad (1)$$

Following the bit mapping, the bit vector \mathbf{x}_t is permuted by a bijective interleaver function π which maps the bit vector \mathbf{x}_t of length N_X to an (interleaved) bit vector \mathbf{x}'_t of the same length. After interleaving, a convolutional channel encoder of rate $r_{\text{CC}} = N_X/N_E$ encodes \mathbf{x}'_t to $\mathbf{e}_t = (e_{t,1}, \dots, e_{t,\eta}, \dots, e_{t,N_E})$ consisting of N_E bipolar bits $e_{t,\eta} \in \{\pm 1\}$. In Turbo-like systems designed for iterative decoding, the rate of the (inner) channel code can be of $r_{\text{CC}} = 1$ (e.g., [16]) or even $r_{\text{CC}} > 1$ with $N_E < N_X$ (e.g., [16, 17]). We restrict our considerations to convolutional codes according to [16, 18], in which the inner component of a capacity-achieving serially concatenated system should be a recursive convolutional code of rate $r_{\text{CC}} \geq 1$.

The single descriptions are generated after channel coding by splitting the vector \mathbf{e}_t into two descriptions $\mathbf{e}_t^{[1]}$ and $\mathbf{e}_t^{[2]}$, which are independently transmitted. In this section, the descriptions are generated by alternately assigning the output bits to both descriptions, i.e.,

$$\mathbf{e}_t^{[1]} = (e_{t,1}, e_{t,3}, e_{t,5}, \dots) \quad (2)$$

$$\mathbf{e}_t^{[2]} = (e_{t,2}, e_{t,4}, e_{t,6}, \dots). \quad (3)$$

The MDCC approach follows the proposal of [19, 20], where the generation of multiple descriptions using a channel code has been attempted.

As the transmitter including the interleaver resembles (with the exception of the demultiplexing of the bit stream) a classical single description transmitter, a receiver incorporating *Iterative Source-Channel Decoding* (ISCD) as proposed in [5, 6, 21] can be used. Near-capacity system configurations of such setups can be realized using the *EXtrinsic Information Transfer* (EXIT) chart technique [22, 23]. After a brief description of the receiver, configuration rules for the transmitter depicted in Fig. 1 are given.

After transmitting the channel encoded bits $\mathbf{e}_t^{[v]}$ over the channel, noisy values $\mathbf{z}_t^v = (z_{t,1}^v, \dots, z_{t,N_E}^v)$ are received. Using the channel statistics, L-values $L_{\text{CD}}^{[\text{chan}]}(e_{t,\eta}^{[v]})$ are computed according to [24]. The aim of ISCD is to jointly exploit the channel-related L-values, the artificial channel coding redundancy, the artificial redundancy possibly introduced by a redundant bit mapping as well as the natural residual source redundancy for approximating the *a posteriori* probabilities $P(i_{t,k} | \mathbf{z}_t, \mathbf{z}_{t-1}, \dots)$. For this aim, a channel decoder and a *Soft Decision Source Decoder* (SDSD) iteratively exchange

extrinsic information in a Turbo-like process. Only a brief description of the ISCD receiver is given here, a detailed description can be found in the literature.

The channel decoder utilized within the ISCD receiver accepts two different inputs. First, the channel related L-values $L_{CD}^{[chan]}(e_{t,\eta}) = [L_{CD}^{[chan]}(e_{t,\eta}^{[1]}) L_{CD}^{[chan]}(e_{t,\eta}^{[2]})]$ are received once per frame. As the ISCD receiver operates on a frame-by-frame basis, the L-values of all transmitted symbols are grouped within a vector which is available to the channel decoder. The utilized channel decoder has to be a *Soft-Input/Soft-Output* (SISO) version of a channel decoder. Besides the channel-related input, the channel decoder requires the (interleaved) extrinsic output of the second component $L_{SD}^{[ext]}(x'_{t,\xi})$ as additional input. The channel decoder computes extrinsic information $L_{CD}^{[ext]}(x'_{t,\xi})$ for the single bits of a frame, which are grouped to a vector, deinterleaved, and fed to the *Soft Decision Source Decoder* (SDSD). The extrinsic output of the channel decoder may contain parts of the channel-related L-values $L_{CD}^{[chan]}(e_{t,\eta})$ if a systematic channel code is employed [23].

The SDSD module consists of two main parts: The *Bit Demapper* and the *Parameter Estimator*. The task of the bit demapper is to generate extrinsic information $L_{SD}^{[ext]}(x_t)$, which is interleaved and fed back to the channel decoder for use in the subsequent iteration. The bit demapper therefore makes use of the (possibly redundant) bit mapping and the inherent residual or natural source redundancy of the quantizer indices $i_{t,k}$. The residual source redundancy needs to be known at the receiver in order to be exploitable. It can either be stored in fixed tables, transmitted over a side channel, modeled [25], or estimated.

Throughout this paper, we distinguish two types of SDSD algorithms, differing in the type of redundancy that is exploited:

- The AK1-INTER algorithm, which exploits the correlation of indices between consecutive frames, i.e., $P(i_{t,k}|i_{t-1,k})$.
- The AK1-INTRA algorithm, which exploits the correlation between indices within a single frame, i.e., $P(i_{t,k}|i_{t,k-1})$.

Equations for both SDSD algorithms can be found in the literature, e.g., [23].

After a fixed number Ω of receiver iterations, the bit demapper computes a set of estimates of the *a posteriori* probabilities $P\{I_{t,k} = q | \mathbf{z}_t, \mathbf{z}_{t-1}, \dots\}$. Using these probabilities, the quantizer reproduction vectors $\hat{\mathbf{v}}_{t,k}$ are reconstructed using MMSE estimation by considering all quantizer code book entries

$$\hat{\mathbf{v}}_{t,k} = \sum_{q=1}^Q \hat{\mathbf{v}}^{(q)} \cdot P(I_{t,k} = q | \mathbf{z}_t, \mathbf{z}_{t-1}, \dots). \quad (4)$$

Finally, the estimated source parameter vector $\hat{\mathbf{u}}_t$ is obtained by concatenating all the estimated values $\hat{\mathbf{v}}_{t,k}$, i.e., $\hat{\mathbf{u}}_t = (\hat{\mathbf{v}}_{t,1} \cdots \hat{\mathbf{v}}_{t,N_t}) = (\hat{u}_{t,1} \cdots \hat{u}_{t,k} \cdots \hat{u}_{t,N_t})$. Using $\hat{\mathbf{u}}_t$, the signal synthesis stage of the source decoder can reconstruct the audio-visual source signal.

The receiver of Fig. 4 resembles a classical ISCD receiver, with several exceptions. At the receiver, first both descriptions are multiplexed and the ISCD receiver given in Fig. 4 can be used. If description v is lost, the received L-values $L_{CD}^{[chan]}(e_{t,\eta})$ of the channel decoder input corresponding to the bits of description v are set to zero (no available information), i.e., $L_{CD}^{[chan]}(e_{t,\eta}^{[v]}) = 0$. If both descriptions are lost, channel decoding is useless and the SDSD has to perform the estimation without input reliabilities. In the case of inter-frame correlation, information from previous frames can be exploited, however, in the case of intra-frame correlation, the SDSD has to output the mean of the central quantizer code book which is optimal in the MMSE sense [1, 26].

MDCC Conditions and Transmitter Setup

The goal of the proposed MDCC approach is to fully reconstruct the original data even if one description is lost. This is, however, only

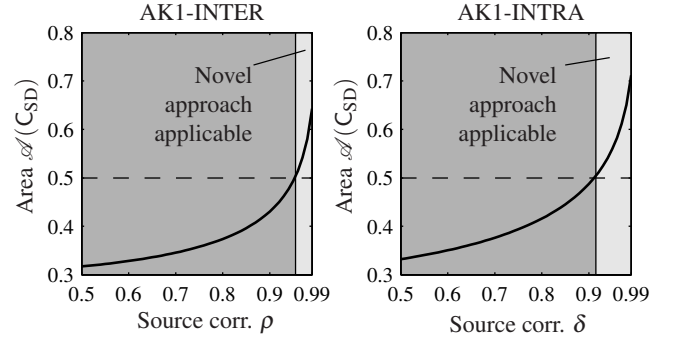


Figure 5: Residual source correlation required for near perfect reconstruction in the case that one description is lost. The MDCC approach (2 descriptions) for $Q = 22$ scalar LMQ and single parity check bit mapping with $B_k = 6$ bits is applied.

possible if the system setup fulfills certain conditions. The analysis of these conditions is subject of the forthcoming paragraphs.

If a description is lost, this means that the output of the channel encoder is punctured such that its rate is doubled. If, for instance, a rate $r_{CC} = 1$ channel code is used, the effective rate of the channel code thus becomes $r_{CC,eff} = 2$ in the case of a packet loss. From [16], it is known that the area underneath the channel decoder EXIT characteristic C_{CD} amounts to $\mathcal{A}(C_{CD}) = \nu_{\phi} / r_{CC,eff}$ if $r_{CC,eff} \geq 1$. In the given example, the channel capacity equals $\nu_{\phi} = 1$ as no AWGN noise is present for the received description and $r_{CC,eff} = 2$, leading to $\mathcal{A}(C_{CD}) = 1/2$. Thus, a necessary (but not sufficient) condition for reconstructing the frame if description v is erased is $1 - \mathcal{A}(C_{SD}) < \mathcal{A}(C_{CD}) = 0.5$, leading to $\mathcal{A}(C_{SD}) > 0.5$.

In order to compare the novel MDCC approach with the original MDC scheme of Fig. 1 utilizing the MDIA of Fig. 2, the following setup is utilized: A $Q = 22$ scalar *Lloyd-Max Quantizer* (LMQ) is followed by a redundant single parity check bit mapping. The bit mapping is realized using a single parity check bit in order to guarantee $d_{min} = 2$. We employ a rate $r_{CC} = 1$ inner convolutional code according to the guidelines in [16]. This setup leads to the same dimensioning as with the MDIA of Fig. 2, resulting in $B_k = 6$ bit per (central) quantizer index $i_{t,k}$. Figure 5 shows the required residual source correlation for guaranteeing $\mathcal{A}(C_{SD}) > 0.5$ for AK1-INTER and AK1-INTRA decoding, indicated by the light-gray regions. It can be seen that quite a high amount of correlation is necessary for guaranteeing decoding if a single description is lost.

Simulation Examples

The capabilities of the proposed approach shall be demonstrated by means of a simulation example. Two different source setups are studied in this example. The first source emits N_I samples generated by a unit-variance, zero-mean Gauss-Markov process of first order with intra-frame correlation coefficient $\delta = 0.98$. No inter-frame correlation occurs for this setup. The second setup uses a source emitting N_I i.i.d. Gauss-Markov samples with inter-frame correlation $\rho = 0.98$ (and no intra-frame correlation, i.e., $\delta = 0$). Such high correlation values can be observed for instance for the gain factors of the *FlexCode* source codec [3, 27]. A $Q = 22$ scalar LMQ is used with a single parity check bit mapping leading to $B_k = 6$, $\forall k \in \{1, \dots, N_I\}$. According to Fig. 5, the MDCC approach is applicable.

The rate-1 recursive non-systematic convolutional channel code with constraint length $J = 3$, generator polynomial $\mathbb{G}^{[CC]} = \{\frac{10}{17}\}_8$ is doped [28] with a doping ratio of 1 : 25 and employed for generating the descriptions. Doping is necessary to trigger the decoding process for the given code [28] in the case of a packet loss, i.e., if only half of the bits are received ($r_{CC,eff} = 2$). In this case, $L_{CD}^{[ext]}(L_{CD}^{[apr]} = 0) = 0$ for the given mother code. Doping leads to

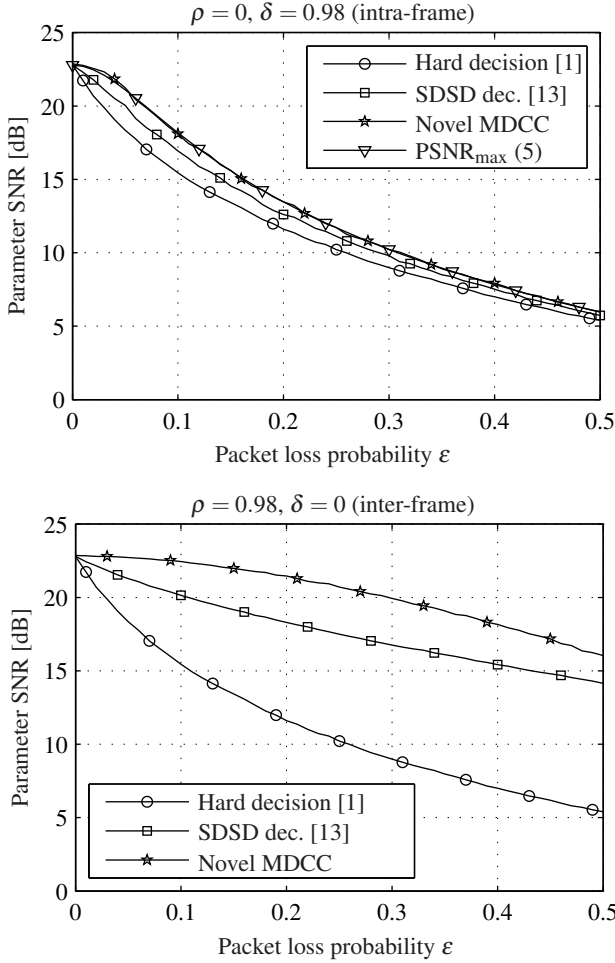


Figure 6: Achievable parameter SNR for the novel MDCC setup and conventional MDC of Fig. 1 [1] with the MDIA of Fig. 2 (Hard decision and AK1 SDSD with MMSE estimation). $N_I = 250$, $Q = 22$ scalar LMQ. MDCC with single parity check bit mapping, doped $r_{CC} = 1$ channel code with $J = 3$, $\mathbb{G}^{[CC]} = \{\frac{10}{17}\}_8$, doping ratio 1 : 25, $\Omega = 25$. Identical number of transmitted bits $N_E^{[1]} = N_E^{[2]} = 750$ in all cases. Two source setups: $\rho = 0$, $\delta = 0.98$ (intra-frame correlation) and $\rho = 0.98$, $\delta = 0$ (inter-frame correlation). No channel noise, i.e., $E_s/N_0 \rightarrow \infty$ (packet losses only).

$l_{CD}^{[ext]}(l_{CD}^{[apr]} = 0) > 0$. Note that the doping ratio needs to be selected such that the doped positions (i.e., the positions where the output bit is replaced by a systematic bit) are equally assigned to $e_i^{[1]}$ and $e_i^{[2]}$. For the demultiplexing strategy given by (2) and (3), this means that the doping ratio has to be based on an odd number. The selection of this convolutional code with the given demultiplexing strategy has been confirmed by an EXIT chart analysis.

Figure 6 shows the behavior of the proposed approach for $N_I = 250$, $Q = 22$ scalar LMQ, and varying packet loss probabilities ϵ if no channel noise is present ($E_s/N_0 \rightarrow \infty$). The parameter SNR between the unquantized source codec parameters $u_{t,k}$ and the reconstructed parameters $\hat{u}_{t,k}$ is used to assess the performance of the system. If both packets are received, the frame can immediately be reconstructed by performing only a single iteration (as for $E_s/N_0 \rightarrow \infty$, the utilized $r_{CC} = 1$ code delivers perfectly reliable extrinsic information, regardless of $l_{CD}^{[apr]}$). In the case of a packet loss, $\Omega = 25$ iterations have been carried out. As a reference serves the non-iterative system setup without channel coding according to Sec. 2, employing either conventional hard decision decoding [1] or

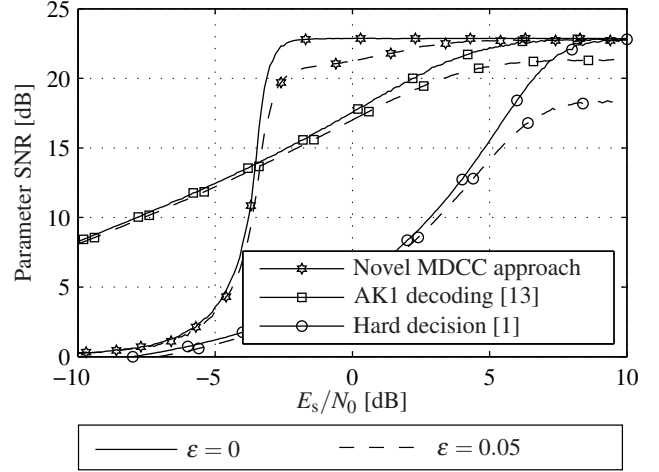


Figure 7: Performance of the novel MDCC scheme in the presence of AWGN channel noise compared to conventional MDC according to Fig. 1 with the MDIA of Fig. 2. Inter-frame source correlation of $\rho = 0.98$, $N_I = 250$, $Q = 22$ scalar LMQ, MDCC with single parity check bit mapping, doped $r_{CC} = 1$ channel code with $J = 3$, $\mathbb{G}^{[CC]} = \{\frac{10}{17}\}_8$, doping ratio 1 : 25, $\Omega = 25$. $N_E^{[1]} = N_E^{[2]} = 750$ in all cases.

AK1-INTRA/AK1-INTER decoding with MMSE estimation [13].

It can be seen that the proposed scheme outperforms the reference setups in all cases. In the case of the source with intra-frame correlation, the following upper limit can be computed, if we assume that the packet can be reconstructed completely as long as one description is available, and that $\hat{u}_{t,k} = 0$ if both descriptions are lost (as in this case the parameter SNR is maximized [26, 1] if the code book is symmetric around zero). This latter case occurs with probability ϵ^2 while the former case occurs with probability $2\epsilon(1 - \epsilon)$. The maximum achievable parameter SNR for the intra-frame correlation case can then be written as

$$\begin{aligned} \text{PSNR}_{\max}(\epsilon) &= \frac{S}{(1 - \epsilon)^2 N_{\text{LMQ}} + 2\epsilon(1 - \epsilon) N_{\text{LMQ}} + \epsilon^2 S} \\ &= \frac{\frac{S}{N_{\text{LMQ}}}}{1 - \epsilon^2 + \epsilon^2 \frac{S}{N_{\text{LMQ}}}}, \end{aligned} \quad (5)$$

with $S = E\{U^2\}$ denoting the power of the source symbols and N_{LMQ} the quantization noise power due to LMQ. If both descriptions are lost, the noise is assumed to be S as $\hat{u}_{t,k} = 0$ and thus $u_{t,k} - \hat{u}_{t,k} = u_{t,k}$. It can be seen in the upper part of Fig. 6 that the proposed approach is able to closely reach this limit.

On the other hand, if inter-frame correlation can be exploited by the AK1-INTER SDSD (bottom sub-plot of Fig. 6), the gains compared to the hard decision case are significantly higher. This is due to the high inter-frame correlation of $\rho = 0.98$ which is used to extrapolate the quantizer indices, even if both descriptions are lost. Also in this case, the MDCC approach leads to significant gains compared to the AK1 decoder, which is due to its ability to perfectly reconstruct the quantizer indices even if one description is missing.

Finally, Fig. 7 shows the behavior of the MDCC approach if AWGN channel noise is present. Results are only presented for the case of inter-frame correlation. Results for intra-frame correlation are similar and can be found in [27]. The simulation results are given for $\epsilon = 0$ (solid lines —) and $\epsilon = 0.05$ (dashed lines ---). Again, the hard decision approach is significantly outperformed by the AK1-INTER decoder. As already observed in Fig. 6, the novel MDCC approach outperforms the conventional

MDC scheme according to Fig. 1 with the MDIA of Fig. 2 (no channel coding) in good channel conditions. Figure 7 confirms that the MDCC approach also outperforms the conventional MDC system for $E_s/N_0 \gtrsim -4$ dB. For $\varepsilon = 0.05$, almost the same reconstruction quality as for $\varepsilon = 0$ is observed for $E_s/N_0 \gtrsim 5$ dB.

Note that no additional data rate is spent explicitly for dedicated channel coding, i.e., the number of transmitted bits per description amounts to $N_E^{[v]} = N_I B^{[v]} = 750$ (plus some terminating bits) in all cases. If the system design allows for channel coding, more redundancy bits can be used for bit mapping (i.e., $B^{[v]}$ can be increased). In this case, the system is still capacity-achieving as $r_{CC} \geq 1$ [16]. If channel coding can be allowed, the prerequisites for the source correlation are not as stringent, as the area under the channel decoder EXIT curve $\mathcal{A}(C_{CD})$ is increased by the additional parity bits.

4. CONCLUSIONS

In this contribution, we have presented a novel multiple description coding concept, where the single descriptions are generated by a convolutional code prior to symbol mapping. This novel alternative approach, denoted *Multiple Descriptions by Channel Coding* (MDCC), has been developed with the goal to exploit the residual source correlation such that the signal can be completely reconstructed if only one description is available at the receiver. The necessary source correlation for near perfect reconstruction in the case of a loss of one description has been determined using an EXIT chart analysis. The resulting system shows a superior reconstruction quality over a wide range of packet loss conditions, compared to a conventional index assignment based multiple description coding scheme. Even in the presence of bit errors due to additive channel noise, an excellent reconstruction quality can be guaranteed over a wide range of channel conditions.

REFERENCES

- [1] V. A. Vaishampayan, "Design of Multiple Description Scalar Quantizers," *IEEE Trans. Inf. Theory*, vol. 39, no. 3, pp. 821–834, May 1993.
- [2] V. K. Goyal, "Multiple Description Coding: Compression Meets the Network," *IEEE Signal Process. Mag.*, vol. 18, no. 5, pp. 74–93, Sept. 2001.
- [3] J. Klejsa, M. Li, and W. B. Kleijn, "FlexCode – Flexible Audio Coding," in *IEEE ICASSP*, Dallas, TX, USA, Mar. 2010.
- [4] T. Fingscheidt and P. Vary, "Softbit Speech Decoding: A New Approach to Error Concealment," *IEEE Trans. Speech Audio Process.*, vol. 9, no. 3, pp. 240–251, Mar. 2001.
- [5] M. Adrat, P. Vary, and J. Spittka, "Iterative Source-Channel Decoder Using Extrinsic Information from Softbit-Source Decoding," in *IEEE ICASSP*, Salt Lake City, UT, USA, May 2001.
- [6] N. Görtz, "On the Iterative Approximation of Optimal Joint Source-Channel Decoding," *IEEE J. Sel. Areas Commun.*, vol. 19, no. 9, pp. 1662–1670, Sept. 2001.
- [7] J. Barros, J. Hagenauer, and N. Görtz, "Turbo Cross Decoding of Multiple Descriptions," in *IEEE ICC*, Apr. 2002, pp. 1398–1402.
- [8] T. Guionnet, C. Guillemot, and E. Fabre, "Soft Decoding of Multiple Descriptions," in *IEEE ICME*, Lausanne, CH, Aug. 2002.
- [9] J. Klejsa and W. B. Kleijn, "Rate Distribution Between Model and Signal for Multiple Descriptions," in *IEEE ICASSP*, Taipei, Taiwan, Apr. 2009.
- [10] J. Klejsa, M. Kuropatwinski, and W. B. Kleijn, "Adaptive Resolution-Constrained Scalar Multiple-Description Coding," in *IEEE ICASSP*, Las Vegas, NV, USA, Mar. 2008.
- [11] Thorsten Clevorn, Johannes Brauers, Marc Adrat, and Peter Vary, "Turbo DeCodulation: Iterative Combined Demodulation and Source-Channel Decoding," *IEEE Commun. Lett.*, vol. 9, no. 9, pp. 820–822, Sept. 2005.
- [12] L. Schmalen, T. Clevorn, and P. Vary, "Iterative Source-Coded Equalization: Turbo Error Concealment for ISI Channels," in *IEEE SPAWC*, Helsinki, Finland, June 2007.
- [13] L. Schmalen and P. Vary, "Reconstruction of Multiple Descriptions by MMSE Estimation," in *ITG Conference on Speech Communication*, Bochum, Germany, Oct. 2010.
- [14] X. Wu and X. Wang, "Joint Source-Channel Decoding of Multiple Description Quantized Markov Sequences," in *Data Compression Conference (DCC)*, Snowbird, UT, USA, Mar. 2006.
- [15] X. Wu, X. Wang, and Z. Wang, "Resource-Scalable Joint Source-Channel MAP and MMSE Estimation of Multiple Descriptions," *IEEE Trans. Signal Process.*, vol. 57, no. 1, pp. 279–288, Jan. 2009.
- [16] A. Ashikhmin, G. Kramer, and S. ten Brink, "Extrinsic Information Transfer Functions: Model and Erasure Channel Properties," *IEEE Trans. Inf. Theory*, vol. 50, no. 11, pp. 2657–2673, Nov. 2004.
- [17] R. Thobaben, "A New Transmitter Concept for Iteratively-Decoded Source-Channel Coding Schemes," in *IEEE SPAWC*, Helsinki, Finland, June 2007, pp. 1–5.
- [18] J. Kliewer, A. Huebner, and D. J. Costello, Jr., "On the Achievable Extrinsic Information of Inner Decoders in Serial Concatenation," in *IEEE ISIT*, July 2006, pp. 2680–2684.
- [19] R. Puri and K. Ramchandran, "Multiple Description Source Coding Using Forward Error Correction Codes," in *Asilomar Conference on Signals, Systems and Computers*, Pacific Grove, CA, USA, Oct. 1999.
- [20] M. R. Stoufs, A. Munteanu, J. Barbarien, J. Cornelis, and P. Schelkens, "Optimized Scalable Multiple-Description Coding and FEC-based Joint Source-Channel Coding: A Performance Comparison," in *Workshop on Image Analysis and Multimedia Interactive Services (WIAMIS)*, London, United Kingdom, May 2009.
- [21] N. Görtz, "Iterative Source-Channel Decoding Using Soft-In/Soft-Out Decoders," in *IEEE ISIT*, Sorrento, Italy, June 2000, p. 173.
- [22] S. ten Brink, "Convergence Behavior of Iteratively Decoded Parallel Concatenated Codes," *IEEE Trans. Commun.*, vol. 49, no. 10, pp. 1727–1737, Oct. 2001.
- [23] Marc Adrat and Peter Vary, "Iterative Source-Channel Decoding: Improved System Design Using EXIT Charts," *EURASIP Journal on Appl. Sig. Proc.*, vol. 2005, pp. 928–941, May 2005.
- [24] J. Hagenauer, E. Offer, and L. Papke, "Iterative Decoding of Binary Block and Convolutional Codes," *IEEE Trans. Inf. Theory*, vol. 42, no. 2, pp. 429–445, Mar. 1996.
- [25] J. Kliewer, N. Görtz, and A. Mertins, "Iterative Source-Channel Decoding With Markov Random Field Source Models," *IEEE Trans. Signal Process.*, vol. 54, no. 10, pp. 3688–3701, Oct. 2006.
- [26] N. S. Jayant and P. Noll, *Digital Coding of Waveforms: Principles and Applications to Speech and Audio*, Prentice-Hall, 1984.
- [27] L. Schmalen, *Iterative Source-Channel Decoding: Design and Optimization for Heterogeneous Networks*, Ph.D. thesis, RWTH Aachen University, 2011.
- [28] S. ten Brink, "Code Doping for Triggering Iterative Decoding Convergence," in *IEEE ISIT*, June 2001, p. 235.