NOVEL ITERATIVE MULTIPLE DESCRIPTION CODING FOR CORRELATED SOURCES

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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 employ approaches to utilize soft information in the decoding of multiple descriptions. However, in the 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. A frame $u_{t}$ consisting of $N_{1}$ unquantized 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. 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.

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. 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 $u_{t}$ consisting of $N_{1}$ 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 $\mathbb{V} = \{v^{(1)}, ..., v^{(Q)}\} \subset \mathbb{R}$. The set $\mathbb{V}$ denotes the central quantizer code book. All quantizer indices of a frame are grouped to the vector $i_{t} = (i_{t,1}, ..., i_{t,N_{1}})$.

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 $i_{t,k}^{[v]} \in \mathbb{V}_{1,1} \subset \mathbb{N}_{1}, v \in \{1, 2\}$ 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 $b_{k}^{[v]} = \mathcal{B}_{v}(i_{t,k}^{[v]})$, consisting of $Q^{(v)}$ bits, is assigned. The bit patterns are selected from a set $\mathcal{B}_{v}^{(Q^{(v)})} = \{b_{k}^{(v)(1)}, ..., b_{k}^{(v)(Q^{(v)})}\}$.

When the channel encoder vectors $e_{k}^{[v]}$ contain $N_{v}^{(k)}$ 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 $[v] = 4$.

Figure 1: Transmitter of a conventional multiple description coding scheme according to [1] including multiple description index assignment (MDIA) and optional channel coding.
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 $\mathcal{B}^v_k$ (i.e., $B^v_k = 3$, $v \in \{1, 2\}$).

![Table with data](image)

**Figure 3: Baseband 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 $\mathcal{B}^4_k(4) = (010)$ for the first description and $\mathcal{B}^{2}(5) = (110)$ for the second description.

Each description is independently transmitted over a packet-erasured AWGN channel. The single bipolar values of the encoded vector $\mathbf{e}_k = (\text{each symbol energy } E_a = 1)$ may be subject to AWGN with power spectral density $\sigma_a^2 = N_0/2$. 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 $\mathbf{e}_k$ may be erased with probability $\varepsilon$. 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 MDAs 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 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 bit pattern $b_{k}\in \mathbb{B} = \{\mathcal{B}^1, \ldots, \mathcal{B}^v\} \subseteq \mathbb{B}^v$ of B bits to each quantizer index $i_{k}$. The single bits of the bit pattern $b_{k}$ are denoted by $b_{i_{k},\mu} \in \mathbb{F}_2$, with $\mu \in \{1, \ldots, B\}$. Denoting the $\mu$th entry of $b_{i_k}$.

After the bit mapping, the $N_f$ individual bit patterns $b_{i_k}$ are grouped to a bit vector $\mathbf{x}_i = (b_{i_1}, \ldots, b_{i_{N_f}})$ = $(x_{1}, \ldots, x_{N_f})$. The size of the bit vector is $N_f = N_f B$.

As the bit mapping is considered to be a code, the rate of the bit mapping is defined by

$$r_{BM} = N_f \cdot \frac{\text{ld} Q}{N_f} = \frac{\text{ld} Q}{B}.$$  \hspace{1cm} (1)

Following the bit mapping, the bit vector $x_i$ is permuted by a bijective interleaver function $\pi$ which maps the bit vector $x_i$ of length $N_f$ to an (interleaved) bit vector $x'_i$ of the same length. After interleaving, a convolutional encoder of rate $r_{CC} = N_f/2B$ encodes $x'_i$ to $e_i = (e_{i_1}, \ldots, e_{i_{N_f}})$ consisting of $N_f$ bipolar bits $e_{i,\mu}$ ($\mu \in \{1, \ldots, N_f\}$). In Turbo-like systems designed for iterative decoding, the rate of the (inner) channel code can be of $r_{CC} = 1$ (e.g., [16]) or even $r_{CC} > 1$ with $N_f < N_f$ (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 $r_{CC} \geq 1$.

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

$$e_i^{[1]} = (e_{i_1}, e_{i_3}, e_{i_5}, \ldots)$$  \hspace{1cm} (2)

$$e_i^{[2]} = (e_{i_2}, e_{i_4}, e_{i_6}, \ldots).$$  \hspace{1cm} (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 [8, 21, 22] 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 $e_i^{[v]}$ over the channel, noisy values $z_k^{[v]} = (z_{k_1}^{[v]}, \ldots, z_{N_f}^{[v]})$ are received. Using the channel statistics, L-values $L_{\text{chan}}^{[v]}(z_k^{[v]})$ 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 $P(b_{i_k}|z_k, z_{k+1}, \ldots)$.

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
\[ L_{\text{CD}}^\text{chan}(\hat{v}, \eta) = L_{\text{CD}}^\text{chan}(v_1^N, \eta) = \begin{bmatrix} L_{\text{CD}}^{\text{chan}}(v_1, \eta) \\ L_{\text{CD}}^{\text{chan}}(v_2, \eta) \end{bmatrix} \]
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
\[ L_{\text{SD}}^\text{ext}(\hat{v}') \]
as additional input. The channel decoder computes extrinsic information \( L_{\text{SD}}^\text{ext}(\hat{v}') \) 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 \( L_{\text{CD}}^\text{chan}(v, \eta) \) if a systematic channel code is employed.

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 \( L_{\text{SD}}^\text{ext}(x) \), 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 \( i_k \). 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 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., \( P(i_k|\hat{i}_{k-1}) \).
- The AK1-INTRA algorithm, which exploits the correlation between indices within a single frame, i.e., \( P(i_k|\hat{i}_{k-1}) \).

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

After a fixed number \( \Omega \) of receiver iterations, the bit demapper computes a set of estimates of the \( a \) posteriori probabilities \( P(i_k|\hat{i}_{k-1}) \) using these probabilities, the quantizer reproduction vectors \( \hat{v}_{\hat{i}_k} \), and the received and deinterleaved L-values of the channel decoder \( L_{\text{CD}}^\text{chan}(\hat{v}, \eta) \).

Finally, the estimated source parameter vector \( \hat{u}_k \) is obtained by concatenating all the estimated values \( \hat{v}_{\hat{i}_k} \), i.e., \( \hat{u}_k = (\hat{v}_{\hat{i}_1}, \ldots, \hat{v}_{\hat{i}_N}) = (\hat{u}_{1}, \ldots, \hat{u}_k, \ldots, \hat{u}_N) \). Using \( \hat{u}_k \), 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 \( L_{\text{CD}}^\text{chan}(\hat{v}, \eta) \) of the channel decoder input corresponding to the bits of description \( v \) are set to zero (no available information), i.e., \( L_{\text{CD}}^\text{chan}(\hat{v}, \eta) = 0 \). 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 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 decoder 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 \( \alpha(\beta_{CD}) = \log_2 r_{CC eff} \) if \( r_{CC eff} = 1 \). In the given example, the channel capacity equals \( \log_2 1 = 0 \) AWGN noise is present for the received description and \( r_{CC eff} = 2 \), leading to \( \alpha(\beta_{CD}) = 1/2 \). Thus, a necessary (but not sufficient) condition for reconstructing the frame if description \( v \) is erased is \( 1 - \alpha(\beta_{CD}) = 0.5 \), leading to \( \alpha(\beta_{CD}) > 0.5 \).

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 \( B_0 = 6 \) bit per (central) quantizer index \( i_k \). Figure 5 shows the required residual source correlation for guaranteeing \( \alpha(\beta_{CD}) > 0.5 \) 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 \( N_f \) samples generated by a unit-variance, zero-mean Gauss-Markov process of first order with inter-frame correlation coefficient \( \delta = 0.98 \). No inter-frame correlation occurs for this setup. The second setup uses a source emitting \( N_f \) 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 code [3, 27]. A \( Q = 22 \) scalar LMQ is used with a single parity check bit mapping leading to \( B_0 = 6 \), \( \forall k \in \{1, \ldots, N_f\} \). 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 \( g(\delta) = [1 \, 0 \, \delta^2] \) 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, \( l_{\text{CD}}^\text{ext}(\hat{v}') = 0 \) for the given mother code. Doping leads to
\[ \rho = 0, \delta = 0.98 \text{ (intra-frame)} \]

\[ \rho = 0.98, \delta = 0 \text{ (inter-frame)} \]

\[ \text{Parameter SNR} \quad \text{Parameter SNR} \quad \text{SNR} \quad \text{SNR} \]

\[ E_{\text{snr}} / N_0 \quad E_{\text{snr}} / N_0 \quad E_{\text{snr}} / N_0 \quad E_{\text{snr}} / N_0 \]

\[ 0 \quad 0 \quad 0 \quad 0 \]

\[ 10 \quad 10 \quad 10 \quad 10 \]

\[ 20 \quad 20 \quad 20 \quad 20 \]

\[ 25 \quad 25 \quad 25 \quad 25 \]

\[ \text{Hard decision [1]} \quad \text{SDSD dec. [13]} \quad \text{Novel MDCC} \quad \text{PSNR}_{\text{max}} (5) \]

\[ \text{Novel MDCC approach} \quad \text{AK1 decoding [13]} \quad \text{Hard decision [1]} \]

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, dropped \( r_{CC} = 1 \) channel code with \( J = 3, C_{\text{CC}} = \{10\} \), doping ratio \( 1 : 25, \Omega = 25 \). Identical number of transmitted bits \( N_0^0 = N_0^2 = 750 \) 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).

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, dropped \( r_{CC} = 1 \) channel code with \( J = 3, C_{\text{CC}} = \{10\} \), doping ratio \( 1 : 25, \Omega = 25 \). \( N_0^0 = N_0^2 = 750 \) 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 \( u_{t,k} = 0 \) if both descriptions are lost (as in this case the parameter SNR is maximized [26, 1] if the codebook is symmetric around zero). This latter case occurs with probability \( \epsilon^2 \) 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

\[ \text{PSNR}_{\text{max}}(\epsilon) = \frac{S (1 - \epsilon)^2 N_{\text{LMQ}} + 2\epsilon (1 - \epsilon) N_{\text{LMQ}} + \epsilon^2 S}{1 - \epsilon^2 + \epsilon^2 \frac{N_{\text{LMQ}}}{N_{\text{LMQ}}}} \]

with \( S = E \{ T^2 \} \) denoting the power of the source symbols and \( N_{\text{LMQ}} \) the quantization noise power due to LMQ. If both descriptions are lost, the noise is assumed to be \( S \) as \( u_{t,k} = 0 \) and thus \( u_{t,k} - u_{t,k} = u_{t,k} \). 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 $E_b/N_0 = -4 \, \text{dB}$. For $\epsilon = 0.05$, almost the same reconstruction quality as for $\epsilon = 0$ is observed for $E_b/N_0 = 5 \, \text{dB}$.

Note that no additional data rate is spent explicitly for dedicated channel coding, i.e., the number of transmitted bits per description amounts to $N_0 = N_B^{(1)} = 750$ (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., $B^{(1)}$ 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 $\phi(C_{TD})$ 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 concept, 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


