Windowed Iterative Source-Channel Decoding with Delay Constraints

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***Abstract*—Iterative source-channel decoding (ISCD) exploits the residual redundancy of source codec parameters by using the Turbo principle. In most practical implementations of ISCD, a delay constraint prohibits the utilization of future frames required for optimal soft decision source decoding if correlation between successive frames is exploited. In this paper, we propose a novel windowed receiver concept that attempts to partly overcome the receiver sub-optimality by refining the extrinsic information of past frames in order to improve the estimation of the parameters of the current frame. Simulation results show consistent performance improvements, which can be visualized by an information flow analysis.**

I. Introduction

With the discovery of Turbo codes, channel decoding close to the Shannon limit has become possible with moderate computational complexity. The Turbo principle is not limited to channel coding but has been extended to other receiver components. One of these extensions is *Iterative Source-Channel Decoding* (ISCD) [1], [2] which allows to exploit the residual redundancy in source codec parameters such as scale factors or predictor coefficients for speech, audio, and video signals in a Turbo process. This residual redundancy occurs due to imperfect source encoding resulting, e.g., from delay constraints. The *a priori* knowledge on the residual redundancy, e.g., non-uniform probability distribution or auto-correlation, can be utilized by a derivative of a *Soft Decision Source Decoder* (SDSD) [3] which exchanges extrinsic reliabilities with a channel decoder.

Two types of correlation may occur: intra-frame correlation, where the source codec parameters within a frame are correlated, and/or inter-frame correlation, where the single parameters in a frame are independent, but there exist a correlation between parameters of consecutive frames. This paper deals with the latter case. Many source codecs aim at removing most of the intra-frame correlation but typically leave most of the inter-frame correlation in order not to introduce additional dependencies between consecutive frames. Because of error propagation, such dependencies can be disadvantageous. If a delay constraint, which requires that a frame has to be decoded immediately upon reception, is imposed, the SDSD can only rely on received values from previous frames. Such a delay constraint occurs for instance in real-time speech and audio communication or video conferencing. Values from future frames are not available, however, they could improve source decoding, as the optimum MAP decoder for Markov sources benefits from information from future frames [4]. In this paper we propose a novel windowed receiver that improves the ISCD performance without violating the delay constraint.

II. System Model

At time instant t, a source encoder generates a frame of N_I i.i.d. source codec parameters $\mathbf{u}_t = (u_{t,1}, \ldots, u_{t,N_I})$, which show inter-frame correlation with correlation coefficient $\rho = \mathsf{E}\{U_{t,k}U_{t-1,k}\}/\,\mathsf{E}\{U_{t,k}^2\}$. Note that we use upper-case letters to denote random variables, e.g., $U_{t,k}$ denotes the random variable describing the source codec parameter $u_{t,k}$.

The parameters of a frame $\mathbf{u}_t$ are quantized using a Q-level scalar quantizer which maps the input parameter $u_{t,k}$ to a quantizer index $i_{t,k}$ denoting the selected entry of the quantizer code book $\mathbb{V} = \{\bar{v}^{(1)}, \ldots, \bar{v}^{(Q)}\} \subset \mathbb{R}$. Note that we assume scalar quantization w.l.o.g., however, all algorithms can be generalized easily to include vector quantization. All quantization indices within a frame are grouped to a vector $\mathbf{i}_t = (i_{t,1}, \ldots, i_{t,N_I})$. To each quantizer index $i_{t,k}$ selected at time instant t and position k, a unique bit pattern $\mathbf{b}_{t,k} \in \mathbb{B} \subseteq \mathbb{F}_2^B$ of B bits is assigned according to the bit mapping function ($\mathbb{F}_2 \doteq \{0,1\}$). The individual bits of the bit pattern $\mathbf{b}_{t,k}$ are denoted by $b_{t,k,\mu} \in \mathbb{F}_2$, with $\mu \in \{1, \ldots, B\} \subset \mathbb{N}_1$ denoting the μth entry of $\mathbf{b}_{t,k}$. The bits of the allowed bit patterns $\bar{\mathbf{b}}^{(q)} = (\bar{b}_1^{(q)}, \ldots, \bar{b}_\mu^{(q)}, \ldots, \bar{b}_B^{(q)})$ are denoted by $\bar{b}_\mu^{(q)}$. For non-redundant bit patterns, $B = \lceil \log_2 Q \rceil$ holds. If $B > \log_2 Q$, the bit mapping is called redundant as more bits than actually necessary are spent to represent a quantizer index. In [5], [6], [7], the benefits of redundant bit mappings are shown. In this paper, we restrict ourselves in most examples to the more "conventional" case of non-redundant bit mappings, however, the proposed receiver can directly be applied to the setup with redundant bit mappings.

After the bit mapping, the N_I individual bit patterns $\mathbf{b}_{t,k}$ are grouped to a bit vector $\mathbf{x}_t \doteq (\mathbf{b}_{t,1}, \ldots, \mathbf{b}_{t,N_I}) = (x_{t,1}, \ldots, x_{t,N_X})$ of size $N_X \doteq N_I B$.

After bit mapping, the bit vector $\mathbf{x}_t$ is permuted by the interleaver function π which maps the bit vector $\mathbf{x}_t$ of length N_X to a bit vector $\mathbf{x}'_t$ of the same length. The interleaving can also be performed for a sequence of Λ consecutive frames $(\mathbf{x}_{t-\Lambda+1}, \ldots, \mathbf{x}_t)$ as shown in [8], resulting in an additional

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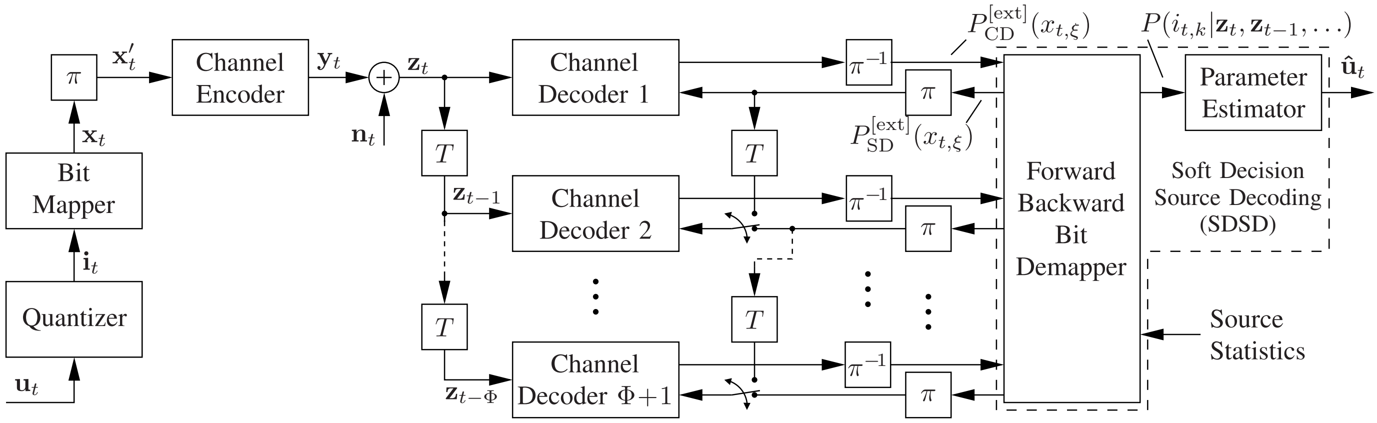

Fig. 1. Transmitter and proposed receiver (baseband model) for improved inter-frame based ISCD utilizing a forward-backward algorithm.

delay of $\Lambda - 1$ time instants. However, this paper deals with the case where this delay cannot be afforded. Therefore, we limit the interleaving to the present frame $\mathbf{x}_t$ only.

After interleaving, a terminated convolutional channel encoder of rate $r_{\mathrm{CC}} = N_X/N_E$ encodes a frame $\mathbf{x}'_t$ to $\mathbf{y}_t = (y_{t,1}, \ldots, y_{t,\eta}, \ldots, y_{t,N_E})$ consisting of N_E bipolar bits $y_{t,\eta} \in \{\pm 1\}$. Note that convolutional encoding is performed on a frame-by-frame basis and the decoder is terminated after each frame. In Turbo-like systems designed for iterative decoding, the rate of the (inner) channel code can be $r_{\mathrm{CC}} = 1$ (e.g., [9]) or even $r_{\mathrm{CC}} > 1$ with $N_E < N_X$ (e.g., [9], [10]). An inner code of rate $r_{\mathrm{CC}} \geq 1$ is in fact a necessary condition to realize capacity-achieving systems. It has furthermore been shown in [11] that the inner code has to be recursive in a system setup designed for iterative decoding.

On the channel, the bipolar symbols of $\mathbf{y}_t$ (with symbol energy $E_{\mathrm{s}} = 1$) are subject to additive white Gaussian noise (AWGN) with known variance $\sigma_n^2 = N_0/2$. After transmitting the bipolar values over the channel, a vector of noisy values $\mathbf{z}_t = (z_{t,1}, \ldots, z_{t,N_E}) = \mathbf{y}_t + \mathbf{n}_t$ is received. Note that we only consider bipolar BPSK modulation in this paper in order to demonstrate the concept, which can easily be extended to higher order modulation schemes.

The aim of ISCD is to jointly exploit the channel-related knowledge, the artificial channel coding redundancy, the artificial redundancy possibly introduced by a redundant bit mapping as well as the natural residual source redundancy for computing the approximated *a posteriori* probabilities $P(i_{t,k}|\mathbf{z}_t, \mathbf{z}_{t-1}, \ldots)$ that are used to estimate the source codec parameters. For the attainment of this aim, a channel decoder and a *Soft Decision Source Decoder* (SDSD) iteratively exchange extrinsic information in a Turbo-like process [1], [2]. The proposed, improved ISCD receiver presented in Sec. III is an extension of the conventional ISCD receiver, which is contained as a special case.

III. Windowed Inter-frame Receiver

A. Receiver Description

The delay-constrained ISCD receiver which exploits inter-frame correlation can only utilize information from all preceding frames. The future frames are not yet available at the receiver during decoding of a frame at time t. This results in a forward-only algorithm in conventional ISCD [1], [2], i.e., the backward recursion of the underlying BCJR algorithm [4] cannot be carried out. In order to improve the inter-frame based decoding, we propose the iterative receiver architecture depicted in Fig. 1.

The idea of the proposed decoder is to refine the extrinsic information of the SDSD for the past frames $t-1$, $t-2$, ..., $t-\Phi$ by exploiting the inter-frame dependencies between the current frame t and the Φ past frames, i.e., by considering a decoding *window* consisting of $\Phi+1$ frames. Using this refined information, the channel decoders of the past frames can generate improved extrinsic information which, in the subsequent source decoder executions, can improve the extrapolation used to estimate the codec parameters of the current frame. The key component of the novel windowed receiver is the use of refined channel decoder output information from past frames.

The novel receiver performs individual channel decoding of $\Phi + 1$ frames $\mathbf{z}_t, \mathbf{z}_{t-1}, \ldots, \mathbf{z}_{t-\Phi}$ received at the (past) time instants $t, t-1, \ldots, t-\Phi$. To ease notation, the past time instants which are jointly considered at the decoder to estimate $\hat{\mathbf{u}}_t$ are grouped into the (time-dependent) set $\mathbb{T}_t \doteq \{t, t-1, \ldots, t-\Phi\}$. The $\Phi + 1$ Soft-Input/Soft-Output (SISO) channel decoders compute extrinsic reliabilities $P_{\mathrm{CD}}^{[\mathrm{ext}]}(x_{t',\xi} = 0)$ and $P_{\mathrm{CD}}^{[\mathrm{ext}]}(x_{t',\xi} = 1)$ for the bits of the past bit vectors $\mathbf{x}_{t'}$, with $t' \in \mathbb{T}_t$.

After deinterleaving, a forward-backward SDSD [4], [8] can then perform the complete inter-frame forward-backward algorithm jointly on the $\Phi + 1$ considered frames and generate extrinsic information $P_{\mathrm{SD}}^{[\mathrm{ext}]}(x_{t',\xi} = \chi)$, $\chi \in \mathbb{F}_2$, $t' \in \mathbb{T}_t$, for each of the $\Phi + 1$ ($= |\mathbb{T}_t|$) channel decoders. Furthermore, an estimate $\hat{\mathbf{u}}_t$ of the present frame is computed.

The first step of the bit demapper component of the SDSD consists in computing the channel-related reliabilities for all $\Phi + 1$ considered frames, with $t' \in \mathbb{T}_t$

$$\gamma_{t',k}(q) = \prod_{\mu=1}^{B} P_{\mathrm{CD}}^{[\mathrm{ext}]}\left(b_{t',k,\mu} = \bar{b}_{k,\mu}^{(q)}\right),$$

where $P_{\mathrm{CD}}^{[\mathrm{ext}]}(b_{t',k,\mu} = \chi) = P_{\mathrm{CD}}^{[\mathrm{ext}]}(x_{t',(k-1)B+\mu} = \chi)$, $\chi \in \mathbb{F}_2$, is the extrinsic reliability of the μth bit of the bit pattern corresponding to the kth source codec parameter. This probability is computed by the channel decoder.

The inter-frame dependencies are exploited within the so-called forward-backward equations [4], [2]. The forward equation, which is executed for all $\Phi+1$ distinct $t \in \mathbb{T}_t$, is given by ($\forall k \in \{1,\ldots,N_I\}$, $\forall q \in \mathbb{I}$)

$$\alpha_{t',k}(q) = \frac{\gamma_{t',k}(q)}{\mathrm{K}_1} \sum_{\tilde{q}=1}^{Q} P(I_{t',k}{=}q | I_{t'-1,k}{=}\tilde{q}) \alpha_{t'-1,k}(\tilde{q})$$

with the source inter-frame *a priori* knowledge $P(I_{t',k}|I_{t'-1,k})$ (obtained via training or using a model) and the normalization constant K_1. At the beginning of the overall transmission, i.e., for $t=1$, the forward reliabilities are initialized with the probabilities of occurrence of the single parameters according to ($\forall q \in \mathbb{I}, k \in \{1,\ldots,N_I\}$)

$$\alpha_{0,k}(q) = \alpha_{-1,k}(q) = \cdots = \alpha_{-\Phi,k}(q) = P(I_{0,k}=q)\,.$$

Note the internal memory for the variables $\alpha_{t',k}(q)$. For $t' = t-\Phi$, the values $\alpha_{t-\Phi,k}$ are computed using $\alpha_{t-\Phi-1,k}$, which has been saved after the execution of the iterative receiver at time $t-1$ (with $\mathbb{T}_{t-1} = \{t-\Phi-1,\ldots,t-1\}$).

The backward recursion is only computed for the past Φ frames $t \in \mathbb{T}_t \setminus \{t\}$ and is given by ($\forall q \in \mathbb{I}, k \in \{1,\ldots,N_I\}$)

$$\beta_{t'-1,k}(q) = \frac{1}{\mathrm{K}_2} \sum_{\tilde{q}=1}^{Q} \gamma_{t',k}(\tilde{q}) P(I_{t',k}{=}\tilde{q} | I_{t'-1,k}{=}q) \beta_{t',k}(\tilde{q})$$

with the normalization constant K_2. At the beginning of the decoding of each frame ($t'=t$), the factors $\beta_{t,k}$ are initialized with $\beta_{t,k}(q)=1$, $\forall q \in \mathbb{I}, k \in \{1,\ldots,N_I\}$.

Using the outcome of the forward and backward recursion, the extrinsic information to be fed back to the channel decoders is obtained for the bits of all indices of all $\Phi+1$ frames ($t' \in \mathbb{T}_t$) by [1], [2] (with $\chi \in \mathbb{F}_2$)

$$P_{\mathrm{SD}}^{[\mathrm{ext}]}(b_{t',k,\mu}=\chi) = \frac{1}{\mathrm{K}_3} \sum_{\substack{q=1 \\ \bar{b}_{k,\mu}^{(q)}=\chi}}^{Q} \frac{\alpha_{t',k}(q)\cdot\beta_{t',k}(q)}{P_{\mathrm{CD}}^{[\mathrm{ext}]}(b_{t',k,\mu}=\chi)}.$$

a) Immediate Estimation: If no delay can be tolerated at the receiver, we compute an immediate estimate of the current frame t. After a fixed number Ω of iterations, the elements of the current parameter vector $\hat{\mathbf{u}}_t$ are MMSE estimated

$$\hat{u}_{t,k} = \frac{1}{\mathrm{K}_4} \sum_{q=1}^{Q} \bar{v}^{(q)} \cdot \alpha_{t,k}(q)\,.$$

Note that for the final estimation, only the factors $\alpha_{t,k}$ are utilized, as only the parameters of the current frame t are estimated, with $\beta_{t,k}(q)=1$ by initialization. Note again that no re-estimation of the previous frames is performed due to delay constraints. After Ω iterations, the extrinsic information of the SDSD can be saved and used as initial *a priori* information for the first execution of the channel decoding stages in the subsequent frames for speeding up convergence. The switches in Fig. 1 are therefore moved into the upper position at the beginning of the first iteration and moved back into the lower position afterwards.

b) Delayed Estimation: If an additional system delay of Δ (with $0 \le \Delta \le \Phi$) frames can be tolerated, the estimate can be improved by taking into account information from future (with respect to the frame to be estimated) frames. In this case, the parameter estimator block in Fig. 1 computes the vector $\hat{\mathbf{u}}_{t-\Delta}$ using

$$\hat{u}_{t-\Delta,k} = \frac{1}{\mathrm{K}_5} \sum_{q=1}^{Q} \bar{v}^{(q)} \cdot \alpha_{t-\Delta,k}(q)\beta_{t-\Delta,k}(q)\,.$$

Finally note that $\Phi=0$ with $\mathbb{T}_t=\{t\}$ yields the original inter-frame ISCD receiver. Although we have given the SDSD equations in the probability domain, an actual realization would be performed in the log-domain for numerical reasons.

B. Bound on Decoding Performance

If we ignore all delay and complexity constraints in the system, we can grow $\Phi \to \infty$ and $\Delta \to \infty$, corresponding to

a) buffering of all received frames
b) considering the complete history of received frames to perform iterations between the channel decoders of all frames and a joint SDSD operating on the full vuffer
c) reconstructing the complete history of received frames.

In a practical simulation, this corresponds to storing all the frames for a single E_b/N_0 step and then performing the processing of all frames at once. Such a buffer-and-decode-all setup gives an upper bound (denoted *buffering bound* in the following) on the achievable system performance.

Note that in a scenario where the source samples do not show any inter-frame correlation, i.e., $\rho=0$, this buffering setup used for computing the bound will lead to the same performance than the "traditional" receiver operating on a frame-by-frame basis. Due to the inter-frame dependencies, however, the buffering bound effectively considers a system with infinite block length, which is one prerequisite to achieve capacity. Note that the finite length analysis using the sphere packing bound which we have conducted in [12] doesn't apply in this case. Finally note the intricate connection between the ISCD system exploiting inter-frame redundancies and spatially coupled codes, e.g., [13], [14]. The inter-frame correlation between neighboring frames corresponds to an implicit spatial coupling.

IV. Simulation Example

A. Strict Delay Constraints ($\Delta=0$)

We show the abilities of the novel receiver by means of a simulation example. A frame consists of $N_I=250$ i.i.d. source parameters generated by independent Gauss-Markov sources with inter-frame correlation $\rho=0.9$. This auto-correlation value can be observed in typical speech and audio codecs, e.g., for the scale factors in CELP codecs or MP3 [3]. Quantization is performed using $Q=16$ level Lloyd-Max quantization and the EXIT-optimized bit mapping found in [8] is used, resulting in $B=4$ bit per index. After S-random interleaving (interleaver size $N_X=1000$), a rate $r_{\mathrm{CC}}=\frac{1}{2}$ recursive, non-systematic convolutional code of

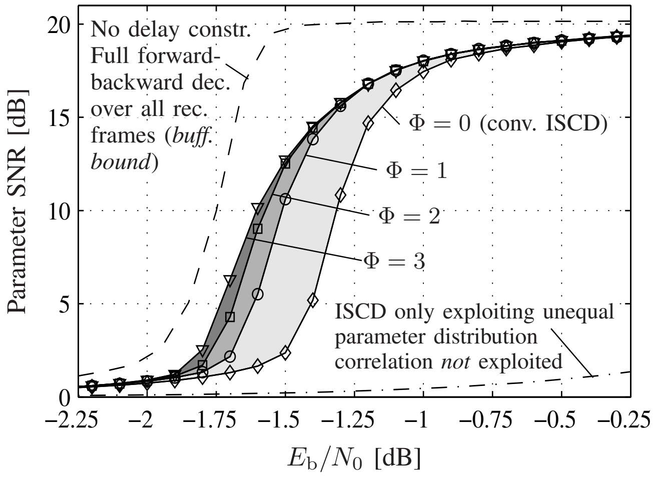

Fig. 2. Parameter SNR performance of ISCD with improved inter-frame decoding for a source with $\rho = 0.9$, scalar LMQ ($Q = 16$), $N_I = 250$, $r_{\text{BM}} = 1$ EXIT-optimized bit mapping, 8-state conv. code ($\mathbb{G}^{[\text{CC}]} = \{\frac{15}{17}, \frac{13}{17}\}_8$).

constraint length $J+1 = 4$ with (octal) generator polynomial $\mathbb{G}^{[\text{CC}]}\{\frac{15}{17}, \frac{13}{17}\}_8$ is used. In the first simulation example we assume immediate estimation ($\Delta = 0$) and no additional receiver delay is tolerated.

Simulation results are depicted in Fig. 2 for $\Phi \in \{1,2,3\}$ and $\Phi = 0$ (conventional ISCD) and for $\Omega = 20$ ISCD iterations. The simulation results show the parameter SNR between the original ($\mathbf{u}_t$) and the estimated source codec parameters ($\hat{\mathbf{u}}_t$) over E_b/N_0. As a reference, the results of ISCD exploiting only unequal parameter distribution (no correlation) are given. The novel receiver outperforms the conventional ISCD receiver ($\Phi = 0$) especially in bad channel conditions. For $E_b/N_0 = -1.5\,\text{dB}$ and $\Omega = 25$, a parameter SNR gain of $\approx 7\,\text{dB}$ is observed. Note that the parameter correlation of $\rho = 0.9$ corresponds to an additional code of rate $r_{\text{SC}} \approx 0.66$ [15], which explains why decoding is possible for $E_b/N_0 < 0\,\text{dB}$ (E_b/N_0 computed w.r.t. r_{CC} only).

Already for $\Phi = 1$, a significant improvement is obtained. Increasing Φ, leads to additional improvements, which are, however, smaller compared with the improvement obtained for $\Phi = 1$. It is noteworthy to mention that the overall decoding complexity linearly scales with Φ. Thus, $\Phi = 1$ might be a best practice selection given a certain performance/complexity trade-off. If an estimate of the channel quality E_b/N_0 is available at the receiver, $\Phi = 0$ can be used in good channel qualities and as soon as E_b/N_0 drops below a certain threshold (e.g., $-1\,\text{dB}$), Φ can be increased to $\Phi > 0$.

The estimates of previous frames cannot be updated due to $\Delta = 0$ and the estimate of the current frame can still not take into account information from future frames. The proposed receiver mainly updates the extrinsic information, which supports the channel decoder of the previous frames to generate better extrinsic information for use within the SDSD. We additionally show the buffering bound (obtained by simulation) as described in Sec. III-B, which is an upper bound on the performance of the proposed receiver.

B. Information Flow Analysis

In order to give an explanation for the performance improvement of the proposed windowed decoder, we first visually show how the information which is exchanged in the $\Phi + 1$ parallel interleavers within the iterations. We follow the EXIT chart approach [16] and record the *mutual information* (MI) between the the data bits $x_{t'}$ and the corresponding extrinsic log-likelihood ratios (or probabilities). For ease of visualization, we restrict ourselves to $\Phi = 1$.

Figure 3 shows the channel decoder output MI $I_{\text{CD}}^{[\text{ext}],t'}$ of both channel decoders ($\Phi + 1 = 2$) for the ongoing simulation example at $E_b/N_0 = -1.5\,\text{dB}$. We can see that for the current frame t, the mutual information increase within the first iterations is dramatic and then saturates towards some value. As, after a fixed number of iterations, this information is shifted towards the next branch (after SDSD processing) for use in the subsequent frame, the value at the second branch $t' = t-1$ of the decoder already starts at a considerably higher value ($I_{\text{CD}}^{[\text{ext}],t-1} \approx 0.86$) and then further increases. Note that the intersection point of the conventional EXIT chart (which is not shown here) already occurs for relatively small values of $I_{\text{CD}}^{[\text{ext}],t'}$ (≈ 0.35).

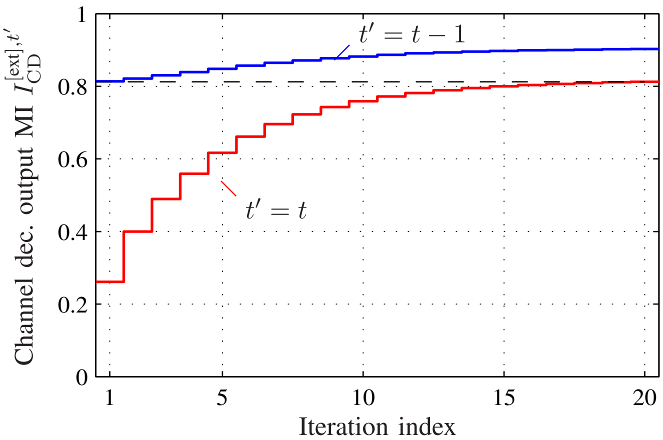

Fig. 3. Extrinsic mutual information at each of the $\Phi + 1 = 2$ channel decoder outputs for the example presented in Fig. 2 at $E_b/N_0 = -1.5\,\text{dB}$

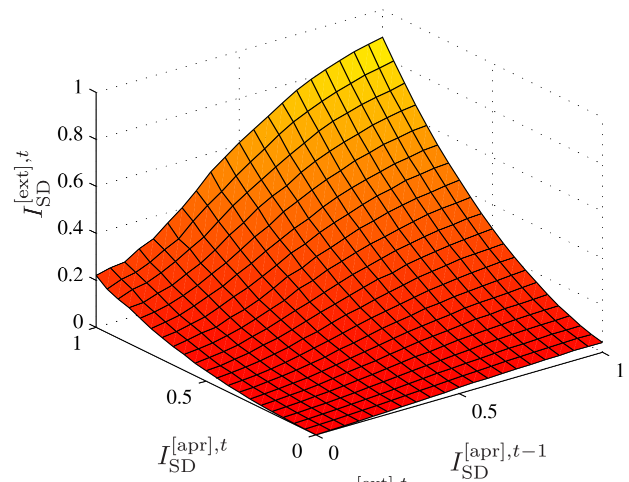

Fig. 4. Extrinsic mutual information $I_{\text{SD}}^{[\text{ext}],t}$ at the output of the current extrinsic SDSD output ($\Phi = 1$) for varying *a priori* inputs

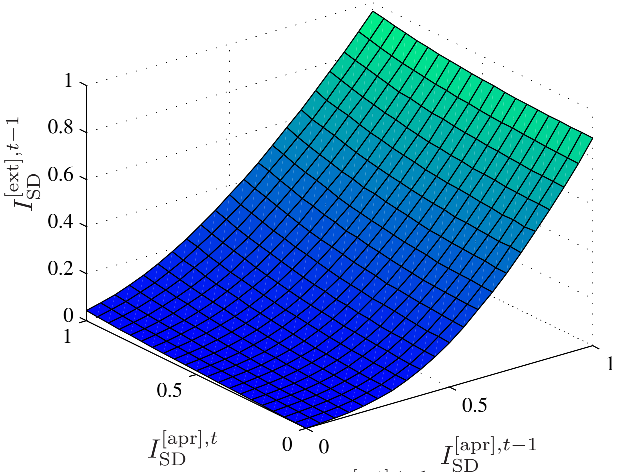

Fig. 5. Extrinsic mutual information $I_{\mathrm{SD}}^{[\mathrm{ext}],t-1}$ at the output of the past extrinsic SDSD output ($\Phi = 1$) for varying *a priori* inputs

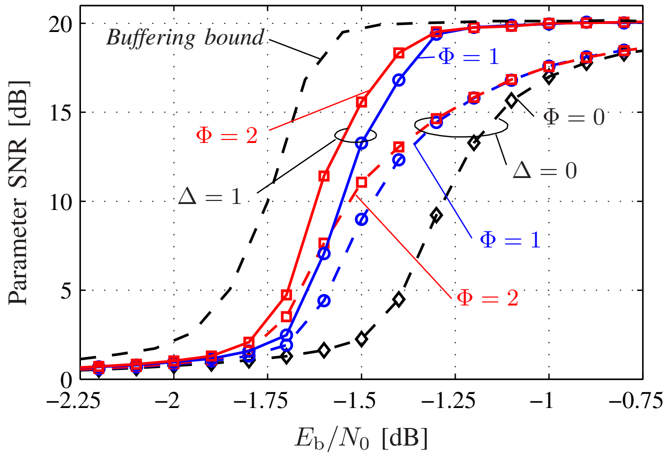

Fig. 6. Parameter SNR performance of ISCD with improved inter-frame decoding and relaxted delay constraints ($\Delta > 0$), same settings as in Fig. 2.

Furthermore, Figs. 4 and 5 show the extrinsic output of the SDSD for the current (t, Fig. 4) and the past frame ($t-1$, Fig. 5) for varying (modelled) *a priori* MI at both inputs. Note that for $I_{\mathrm{SD}}^{[\mathrm{apr}],t} = 0$ in Fig. 5, we get the original forward-only SDSD EXIT characteristic (no information from the future). The convergence analysis can be carried out by analyzing the flow of information on both branches of the decoder. The SDSD output MI can be taken from Figs. 4 and 5 while the channel decoder output MI is obtained from its conventional EXIT characteristic [16].

The above plots, especially Fig. 3 show the steady state performance. If during operation, Φ is increased from 0 to 1, then some kind of bootstrapping effect takes place. For $E_{\mathrm{b}}/N_0 = -1.5\,\mathrm{dB}$ in this example, it takes around 30 iterations (or 2 frames, if 20 iterations/frame are carred out), until the steady state is reached.

C. Relaxed Delay Constraints ($\Delta > 0$)

Finally, we visualize the effect of relaxing the delay constraints on the estimation performance. The simulation results are depicted in Fig. 6. The results of Fig. 2 (for $\Phi \in \{0, 1, 2\}$) are reproduced as dashed lines for comparison. Already with $\Delta = 1$, we observe large performance improvements over the whole range of considered E_{b}/N_0 values. Also with $\Delta = 1$, further increasing Φ leads to additional performance gains.

V. Conclusion

We have presented a novel improved receiver for delay-constrained ISCD exploiting inter-frame correlations. The new receiver employs a forward-backward SDSD which refines the extrinsic information of the current and past frames. This information is iteratively evaluated in a loop comprising channel decoding of several past frames. The proposed system generates improved estimates of the current frame's source codec parameters without introducing any additional delay. Simulation results show a relevant and consistent performance improvement, especially in bad channel conditions. The performance improvement has been explained by means of an information flow analysis. Furthermore, we have shown that already with a slight relaxation of the delay constraint, additional improvements can be obtained.

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