On the Application of Turbo Equalizers in GSM Compatible Receivers

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ABSTRACT

In this paper, we discuss the application of iterative, decoding algorithms (turbo-equalizer) to GSM data channels (TCH/F9.6). Although the turbo-equalizer under realistic time-variant channel scenarios has a high potential, we only get a minor performance gain when compared to conventional soft-decision algorithms. Certain constraints of the GSM coding scheme prevent better results. In this paper, we discuss the transition from a very simple data transmission model to a full GSM-compatible system and compare the resulting bit-error-rates (BER). We indroduce a modified turbo-equalizer structure which adapts to the pipelining interleaver.

I. INDRODUCTION

Recently, we published the application of iterative decoding techniques in conjunction with GSM *speech* channels [3]. There are certain contstraints that prevent better results:

- small interleaver with pipelinening structure
- coding scheme with unequal error protection
- 78 uncoded bits.

This paper adds simulations for GSM *data* channels. These channels are more attractive for iterative decoders, because a different channel coding scheme is used, the interleaver covers more bursts and additional time delay is not so critical. There are several publications analyzing turbo-codes in conjunction with GSM[7], but channel coding is applied, that is different from GSM standard. Our intention is to be fully compatible with the existing specification. For the readers convenience we will repeat the basics of GSM in section II and summerize the differences of speech and data channel coding in paragraph II.A (In general we will use TCH/FS for speech and TCH/F9.6 for data channels[5].)

II. BASICS OF GSM

In this section, we will summarize components of the GSM system, as far as they concern the turbo-equalizer. Since the GSM-standard specifies only minimum performance criteria and does not define an exact model for the receiver, it is possible to apply new algorithms to increase the receiver's performance. The only constraint is that the receiver must be compatible with the specified transmitting scheme.

Figure 1 shows the data transmission scheme of a full rate data channel (traffic channel / full rate data [TCH/F9.6]) with 12kbit/s radio interface. Each frame contains 240 bits. In this transmission mode a ½-rate convolutional code is used for error protection. The coded data is modulated and transmitted. A channel model summarizes the effects of the physical channel. However, in the channel model, the RF components in the transmitter and receiver are omitted. The receiver demodulates the received Gaussian Minimum Shift Keying (GMSK) signal. Multipath effects are reduced with an equalizer in conjunction with channel estimation. The ECC-decoder exploits the additional information to reduce the BER. Although not specified, both equalizer and ECC-decoder are usually Viterbi-algorithms.

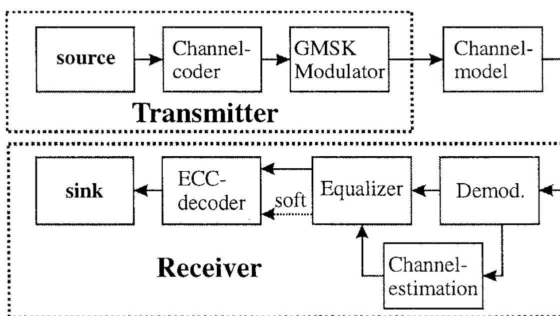

Figure 1: GSM transmission scheme

GSM uses bursts containing 148 bits to transmit data over the radio interface (see Figure 2). Two blocks with 57 bits each are used for transmission. A special training sequence (midamble) supports channel estimation.

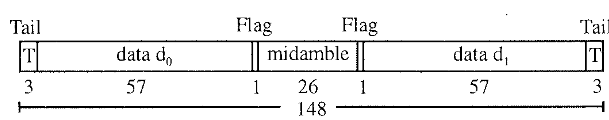

Figure 2: GSM burst structure

All 240 bits of a data frame are protected with a ½-rate convolutional code. The two polynomials are $G_0=1+D^3+D^4$ and $G_1=1+D+D^3+D^4$. 4 Tailbits are added at the end of the input sequence. Puncturing is used to fit within the GSM burst structure (see Figure 3).

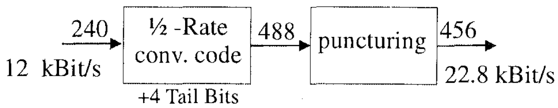

Figure 3: GSM TCH/F9.6 data channel coding.

One coded frame contains 456 bits. The data is spread over 22 consecutive bursts to reduce the effect of burst errors. Each GSM burst carries 114 bits. Figure 4 illustrates this pipelining technique. To decode one frame (e.g. frame 0), 22 bursts have to be received.

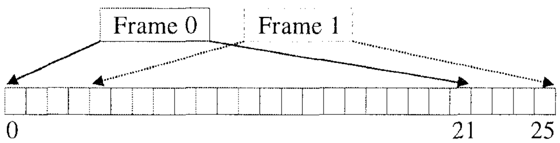

Figure 4: TCH/F9.6 interleaving scheme.

A. *COMPARISON OF SPEECH AND DATA CHANNELCODING*

In this section we will summarize the differences of speech and data channels.

Both channels use the same burst structure (114 bits) and the same frame structure for coded data (456 bits). Due to the demand of small delays, the speech channel interleaver spreads one frame over 8 bursts (22 bursts for data channels). Speech channels use an unequal error protection scheme (78 bits are uncoded (class2), 50 bits are protected with 3 parity bits (class1a), the remaining 132 bits utilize a ½ -rate convolutional code). In contrast, data channels use this ½ -rate convolutional code for all 240 bits.

III. REVIEW OF ITERATIVE DECODING

Figure 5 shows a basic digital communcation system utilizing turbo-decoding. The user data are protected by a convolutional code and spread by an interleaver. For simplification modulation aspects are omitted. We also assume a perfect channel estimation for the MLSE equalizer (maximum likelihood sequence estimation). The serial combination of two Viterbi algorithms (equalizer and convolutional decoder) is typical for turbo-equalizing.

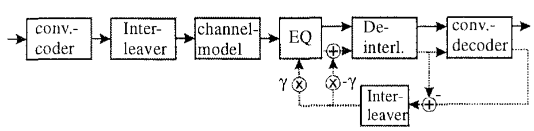

Figure 5: A simple model utilizing iterative decoding.

Older approaches only use harddecision between the equalizer and convolutional decoder. By using soft-decision algorithms (SDAs) within the equalizer we can improve the systems performance by typical 3 dB. A soft-output detector computes propabilities p_j for each bit. Usually, these values are used in the logarithmic domain.

The most famous algorithm is the SOVA by Hagenauer [6]. But there are more powerful algorithms [1],[11]. Actually, we have implemented 6 algorithms shown in Figure 6. We prefer the Max-Log-MAP because its performance is close to optimal algorithms at reasonable computational cost. Today, current cellular phones use very simple SDAs clearly below the SOVA to save cost.

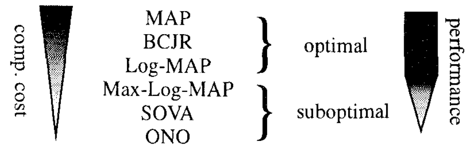

Figure 6: Several soft-output detectors.

Although the MAP and similar algorithms are optimal, the *combination* of equalizer and convolutional decoder is still suboptimal. With a special feedback structure, we can improve the performance. In the following paragraph the main ideas of the turbo-equalizer will be discussed. Details can be found in [2]. The convolutional decoder computes additional soft-information (liability information L_j) for its *input* data. The old soft-information for the input must be substracted to feedback only the *extrinsic* information through an additional interleaver. The equalizer uses this information as *a priori* data to improve its estimation. The main idea is that traditional systems (without feedback) do not exploit constraints of the convolutional code for equalizing. This means that the equalizer and convolutional decoder are only loosely combined. The equalizer tries to find the MLSE-sequence which fits best the received data, but does not take properties of the convolutional code in consideration. These properties can be used within the equalizer to prefer certain sequences and increase the total system performance. A scaling factor γ must be used to damp the *apriori* information. Currently, only empiric approaches are known to compute γ. Reasonable values are between 0.01 and 0.5. The feedback loop can be repeated n times, but with each iteration the performance gain will decrease. For our simulations, we used maximally 5 iterations.

IV. SIMULATION RESULTS

We begin our study with the simple model shown in Figure 5. We chose this model for comparision with [2] and to demonstrate the potential of iterative decoding. The convolutional coder uses the same polynomials as specified for GSM TCH/F9.6. The interleaver is a pseudorandom blockinterleaver with arbitrary blocklength. For the first simulation, we use a fixed channel h(k) with the following impulse response h(k)= sqrt([0.45, 0.25, 0.15, 0.1, 0.05]). The interleaver has a blocksize of 4096 bits. Under these conditions (which are similar to [2]) we can improve the BER at 10^{-3} by 2.5 dB (at the fifth iteration) compared to soft-decision approaches. Figure 7 shows the BER in terms of SNR. As reference, we plot the systems performance under additive white Gaussian noise conditions (AWGN).

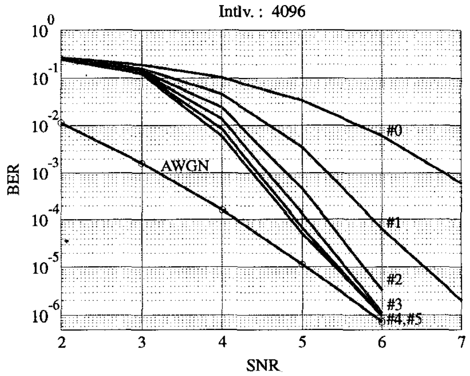

Figure 7: time-invariant, size of interleaver: 4096 (#0=soft-decision without feedback, #1= 1th iteration etc)

The top curve (#0) is the classical soft-decision algorithm without feedback. The first iteration (#1) gains about 1.5 dB. This improvement decreases with each iteration. Asymptotically, AWGN performance is reached, meaning that any channel interference can be removed. These results encouraged us to apply this approach to GSM.
But we have to remember that this simple model does not cover all aspects of a TCH/F9.6. This includes:

- diagonal interleaver (pipelining scheme)
- nonlinear phase modulation (GMSK)
- demodulator
- receiver filter (typ. 5-pole butterworth lowpass)
- channel estimation based on the midamble.

In our next step, we apply the turbo-equalizer to our full GSM compatible simulation system.

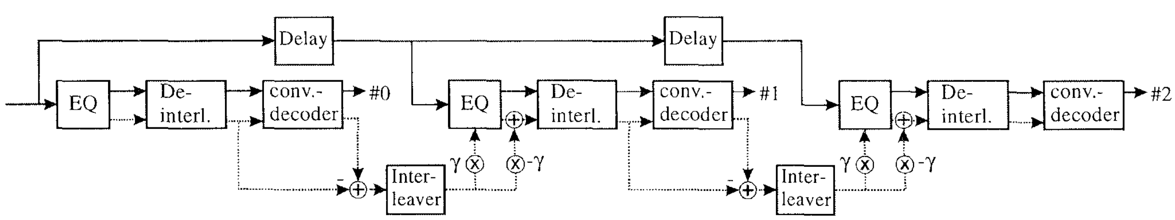

Figure 8: pipelined turbo-equalizer for TCH/F9.6. #0 is the output of the first stage (without iteration), #1 is the output of the second stage etc

A. *Modified turbo-equalizer*

Due to the pipelining interleaver structure (Figure 4) of TCH/F9.6 channels the turbo-equalizer can not be adapted directly. Let us assume, that we decode one frame with the turbo-equalizer. First, we have to receive 22 Bursts (2508) bits to include one entire frame. 1026 bits are assigned to previous frames and 1026 bits to following frames. Table 1 shows the distribution of bits assigned to the current frame, previous frames and future frames. Only 456/2508≈18% of the equalizer input are passed through the feedback loop, leading to poor performance.To overcome this problem, we introduce a new structure: Figure 8 shows a modified turbo-equalizer, which takes the pipelined interleaver into consideration. The main idea is to unroll the iteration loop. The liability information from the first stage (classical softdecision without feedback) is passed through the next stage using a FIFO-architecture (first in first out). The equalizer of the second stage has now apriori information for *every* bit. On the other hand, we have to accept a minimal delay of 6 frames (120ms) per iteration. A similar structure could be used to decode GSM *speech* frames but additional delay is much more critical for speech channels.

burst	0	1	2	3	4	5	6	7	8	9	10
current	6	12	18	24	24	24	24	24	24	24	24
previous	108	102	96	90	84	78	72	66	60	54	48
next					6	12	18	24	30	36	42

burst	11	12	13	14	15	16	17	18	19	20	21
current	24	24	24	24	24	24	24	24	18	12	6
previous	42	36	30	24	18	12	6				
next	48	54	60	66	72	78	84	90	96	102	108

Table 1: properties of the TCH/F9.6 interleaver

Figure 9 shows a simulation of a TCH/F9.6 channel under time-invariant conditions (same channel as in Figure 7). As a reference we plot the BER without channel coding (RAW). At the fourth iteration we achieve approx. 2 dB. Let us analyze this gain in some more detail: Figure 10 shows the distribution of biterrors within 120 frames at 3dB (the channel conditions remain unchanged). The results are impressive. Almost all frames (except 4) are corrected.
In a next step we use realistic *time-variant* channel models.The channel impulses are uncorrelated from each other. This is a typical situation at low speeds and ideal frequency hopping. We use standard channel profiles (rayleigh fading) specified by COST-207 [8].

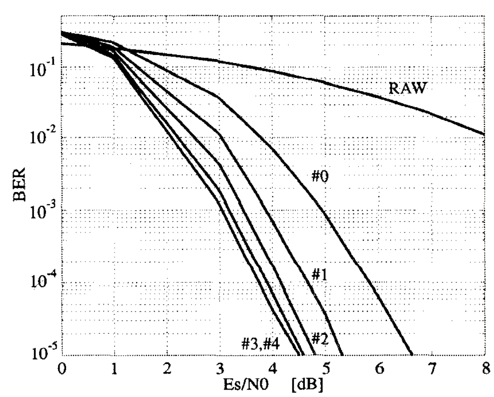

Figure 9: GSM TCH/F9.6 time-invariant

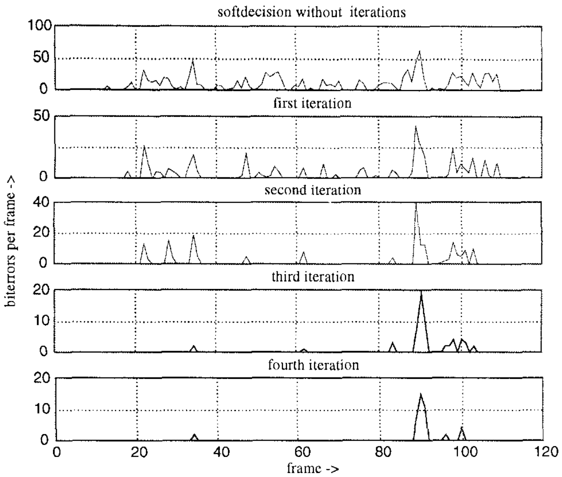

Figure 10: time-invariant at 3dB

Under time-variant conditions the performance of the turbo-equalizer dramatically degrades. Figure 11 shows a simulation for a TU50 channel. We get a performance gain of 0.5dB only!. With other channel profiles we reach similar results. There are specific situations where we can improve the BER considerably. But averaged over many channels, the improvement is small.

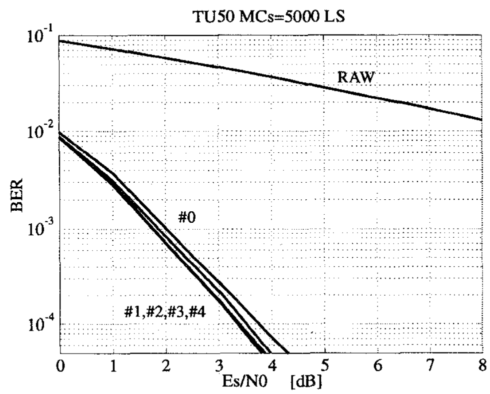

Figure 11: TCH/F9.6 (time-variant TU50 channel)

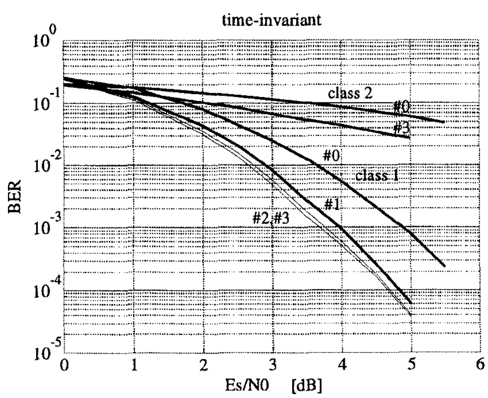

Figure 12: performance of the turbo-equalizer (TCH/FS, time-invariant)

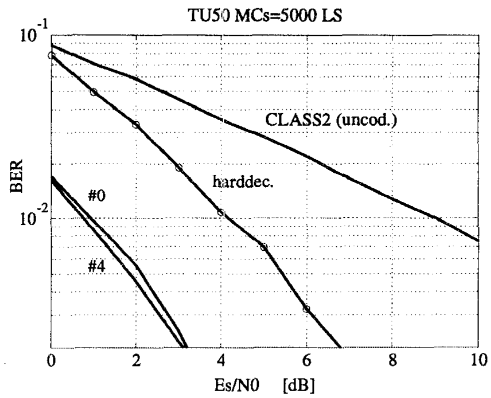

Figure 13: performance of the turbo-equalizer (TCH/FS, time-variant TU50).

In [3] we published results for GSM speech channels. The effect was similar: under realistic time-variant channels the gain was neglibile. For comparison with data channels we repeat these results (see Figure 12 and Figure 13). At time-invariant conditions we get 1.2dB (CLASS1) for speech channels and 2 dB for data channels. Under time-variant scenarios we reached 0.1dB (CLASS1) for speech and 0.5dB for data channels.

V. CONCLUSION

The turbo-equalizer has high potential and is promising for modern digital communication systems. However, if it is used with an existing digital transmission system, constraints are imposed due to the existing coding schemes. When the turbo-equalizer is applied to GSM traffic channels (speech and data), only a small gain is achieved for realistic, time-variant channels. Special properties of the GSM transmission scheme affect the iterative concept. The use of small interleavers is a crucial, negative factor in the use of the turbo-equalizer. They occur due to the demand of being compatible with the existing specification.

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