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TRANSMISSION WITH TURBO TRELLIS-CODED MODULATION ON RADIO CHANNEL WITH THE BANDWIDTH OF 25 kHz

Design of telecommunication systems that operate on radio channels with the bandwidth of 25 kHz with relatively high data rate is not an easy task. Such a systems should allow to transmit either voice or digital data. In this paper, we discuss one of a few possible solutions, which allow realization of such a telecommunication system - TTCM (*Turbo Trellis-Coded Modulation*). The paper begins with a short description of basic features of TTCM systems and two decoding algorithms. The second part contains necessary information about square-root raise cosine filter, which operates as a pulse shaping filter. In this part it is also shown, how to choose appropriate constellation in case of use of a non-linear power amplifier. Finally, obtained results with their short analysis are presented.

Keywords: TTCM, Turbo Trellis-Coded Modulation, 25 kHz channel

1. CONCEPT OF TURBO-TRELLIS CODED MODULATION.

1.1 Introduction

TTCM (*Turbo Trellis-Coded Modulation*) is a combination of Trellis Coded Modulation with the concept of turbo-coding. Two main concept of TTCM are known: parallel and serial concatenation. Parallel concatenation of TTCM coding was introduced by Robertson and Wörz [1 - 3]. The block diagram [4] is shown in Fig. 1.1. As we can see this concept is very similar to the classical parallel turbo-coding, but there are same major differences [2]:

– the interleaver operates on group of m bits instead of single bits,
– achieving the desired spectral efficiency requires mandatory puncturing of the parity bits and is not quite as straightforward as in binary turbo-codes,

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– there are some restrictions on component codes and interleavers,
– component encoders are trellis encoders.

These differences are also applied to the serial realization of TTCM.

Parallel concatenation is seen as a direct extension of classical turbo-coding. In this solution we use two TCM encoders that work in parallel. The input data to the first TCM encoder is a given data sequence. The input data to the second TCM encoder is the same data sequence after interleaving. The final sequence of transmitted symbols is generated by selecting symbols alternately from the two encoders (output symbols of the second one may be de-interleaved but this operation can be carried out in the decoder as well). Those symbols are subsequently mapped and shaped with the filter in the transmitter. Finally such a signal is transmitted. Parallel TTCM has found only small interest, because both encodes affect the modulator.

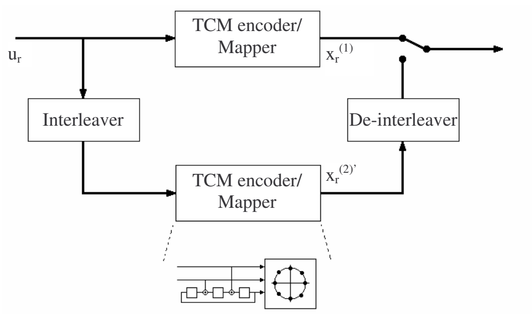

Fig. 1.1. Block diagram of parallel TTCM encoder with 8PSK encoders [4]

A decoder is structured analogously to the classical turbo-decoder and is shown in Fig 1.2. The decoder operates on the received sequence of symbols $\mathbf{y}$. This sequence is alternately fed to the first and second SISO (*Soft Input Soft Output*) decoders which are related to the first and second encoder, respectively. The SISO decoders can operate according to many decoding algorithms. In simulation model the MAP (*Maximum A posteriori Probability*) algorithm and the SOVA (*Soft Output Viterbi Algorithm*) algorithm are used.

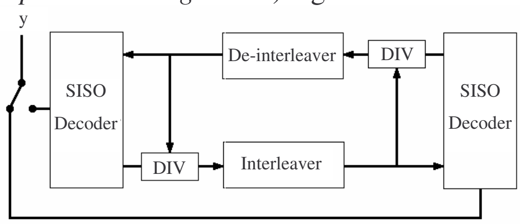

Fig. 1.2. Block diagram of parallel TTCM decoder [4]

The second concept of TTCM is a serial concatenation TTCM. This concept uses cascade coding, where the inner encoder codes the output data of the outer encoder. In serial TTCM a binary, convolutional encoder usually plays the role of the outer encoder. The output signal of the outer encoder is interleaved and put into the inner encoder.

Serial TTCM finds a wide interest, particularly in the context of space-time coding. Detailed information about parallel and serial concatenated TTCM can be found in [1-4]. Necessary information about calculating *a priori* information for the first decoding step is introduced in [3].

1.2. Decoding strategies

As mentioned before various decoding algorithms can be used. They can differ in computation complexity and obtained performance. The best results can be obtained with the MAP algorithm. The purpose of the MAP algorithm is to calculate *a posteriori* probability such as $\Pr(u_r \mid \mathbf{y})$ or $\Pr(x_r \mid \mathbf{y})$, where $\mathbf{y}$ is the received sequence observed at the output of the channel, whose input is the transmitted sequence $\mathbf{x}$. The algorithm does not compute these probabilities directly, but computes the probability of the event that the encoder traversed a specific transition in the trellis: $\Pr(s_r = i, s_{r+1} = j \mid \mathbf{y})$, where s_r is state in time instant r, s_{r+1} is state in next time instant. These probabilities are computed as:

$$\Pr(s_r = i, s_{r+1} = j \mid \mathbf{y}) = \frac{1}{\Pr(\mathbf{y})}\alpha_{r-1}(i)\gamma_r(j,i)\beta_r(j) \tag{1}$$

Values α and β are internal variables of the algorithm and are computed during the forward and backward recursion of the algorithm, respectively. The γ values are conditional transition probabilities and are the inputs to the algorithm.

The other example of decoding algorithm is SOVA. This algorithm has a lower computation complexity than the MAP algorithm and is slightly worse in the performance, as it will be shown in the next section. The SOVA algorithm that is used in simulations is bi-directional algorithm and was introduced in [5]. Detailed information about MAP and SOVA can be found in [4] and [5].

2. PULSE SHAPING AND CONSTELATION SELECTION

2.1 Pulse shaping

After comprehensive studies on shaping filters the square-root raised cosine filter with the roll-of factor of 0,25 has been chosen. Thus, the power spectral density *G(f)* of the transmitted signal is given by the formula:

$$G(f)=\begin{cases}1 & for \quad 0\le |f| \le \dfrac{1-\alpha}{2T}\\ \dfrac{1}{2}\left\{1+\cos\left[\dfrac{\pi\cdot T}{\alpha}\left(|f|-\dfrac{1-\alpha}{2T}\right)\right]\right\} & for \quad \dfrac{1-\alpha}{2T}\le |f| \le \dfrac{1+\alpha}{2T}\\ 0 & for \quad |f|\ge \dfrac{1-\alpha}{2T}\end{cases} \tag{2}$$

Defining the signal bandwidth for which the 10 dB loss from the maximum is observed, we are able to determine the symbol data rate $f_{symb} = 1/T$ from the following formula:

$$\frac{1}{2}\left\{1+\cos\left[\frac{\pi\cdot T}{\alpha}\left(|f|-\frac{1-\alpha}{2T}\right)\right]\right\}\ge\frac{1}{10} \tag{3}$$

This leads us to the result:

$$f_{symb}\le\frac{W}{0{,}59\cdot\alpha+1} \tag{4}$$

So in 25 kHz channel we can transmit up to 21.786 ksymb/s. For implementation simplicity 21.5 ksymb/s has been selected. The theoretical bandwidth of the signal transmitted with this data-rate can be calculated from the formula:

$$W=\frac{\alpha+1}{T}=f_{symb}\cdot(\alpha+1) \tag{5}$$

and it is equal to 26,875 kHz. Thus, transmitting with this data-rate causes the theoretical transmitted signal spectrum to extend our 25 kHz band by 7,5%.

2.2. Selecting constellation and TCM encoder

Our next task is to choose which constellation should be used to obtain the best performance in form of a bit error rate (BER). Since we transmit 21.5 ksymb/s and we aim at transmitting at the data rate of 64 kb/s we must send at least 3 information bits per symbol. Since parallel TTCM adds one parity bit to each symbol we have to use modulation that allows to transmit 4 bit/symb. The most commonly used are 16-QAM and 16-PSK. The crucial problem is which of these modulations allow us to have lower BER with using non-linear power amplifier.

For modeling of a non-linear power amplifier the Rapp model can be used that is given by formula:

$$R_{out} = \frac{R_{in}}{\sqrt[2p]{1 + \left(R_{in} / V_{sat} \right)^{2p}}} \tag{6}$$

where: R_{in} is the amplifier input signal, R_{out} is the amplifier output signal, V_{sat} is the amplifier saturation level and p is a parameter (typical value is 2).

For simulation we can assume $V_{sat} = 1$ and $p = 2$. In order to compare 16-QAM with 16-PSK simulation, the model in Simulink, which is shown in Fig. 2.1 has been applied.

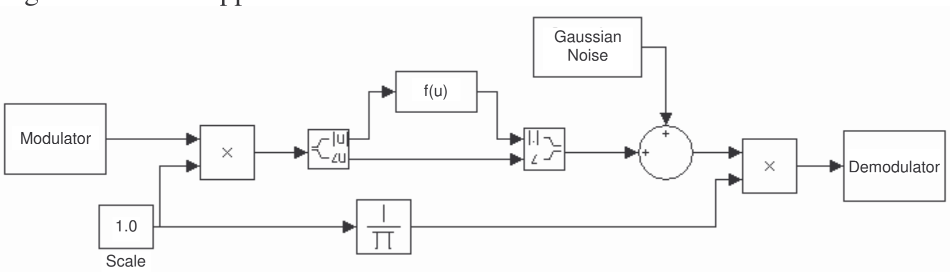

Fig. 2.1. Block diagram of Simulink simulation model

Our investigations over the above model have indicated that 16-QAM is a better choice so in our model 16-QAM has been applied.

The next problem is to choose the appropriate TCM encoder. In literature many encoders can be found that can be used in our telecommunication system. Examples of such encoders are presented in [3]. These encoders are optimized to maximize the Euclidean minimum distance. Among these, which use 16-QAM modulation, two encoders are shown in [3] which can be selected. The first one is an 8-state encoder and the second one is a 16-state encoder. Since the main purpose is to implement TTCM encoder and decoder on a given DSP we should choose the 8-state encoder because of too high computation complexity of the other one. Parameters of this coder are (in octal notation):

$$\mathrm{H_0(D) = 11, H_1(D) = 02, H_2(D) = 04, H_3(D) = 10}$$

2.3. Additional assumptions to the simulation model

Some additional assumptions and remarks that have not been mentioned yet and may have influence on obtained results are that:

– the s-random interleaver with size N = 512 or N = 3072 is applied,
– probabilities of transmitted symbols are equal,
– receiver and transmitter are in perfect synchronization,
– results are obtained for an AWGN channel and for a static two-path channel with the Zero Forcing linear equalizer,
– so far simulation model uses a linear power amplifier (a non-linear power amplifier will be used in a DSP implementation with a real RF front-end).

3. RESULTS

All presented results (BER) are an average value of 10 simulation runs (all input parameters such as: interleaver size or SNR are the same). Each simulation run is terminated if number of errors reaches 100.

Figs. 3.1. and 3.2 present the BER curves. In each figure four curves represent results for 1, 2, 4 and 8 iterations of the decoding algorithm.

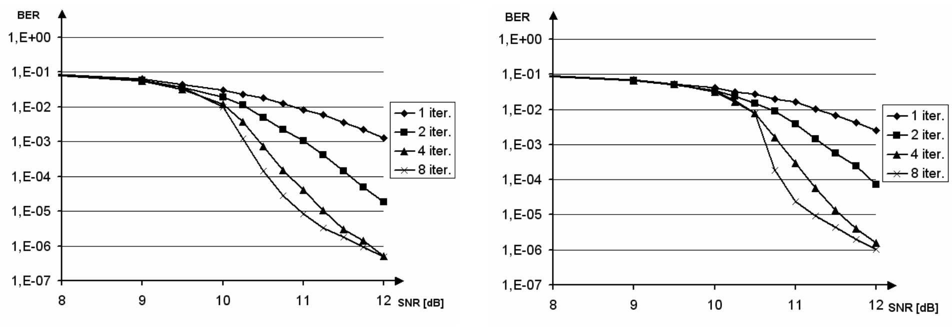

Fig. 3.1. Results for the MAP algorithm, N = 3072, AWGN channel (left) and two-path channel (right)

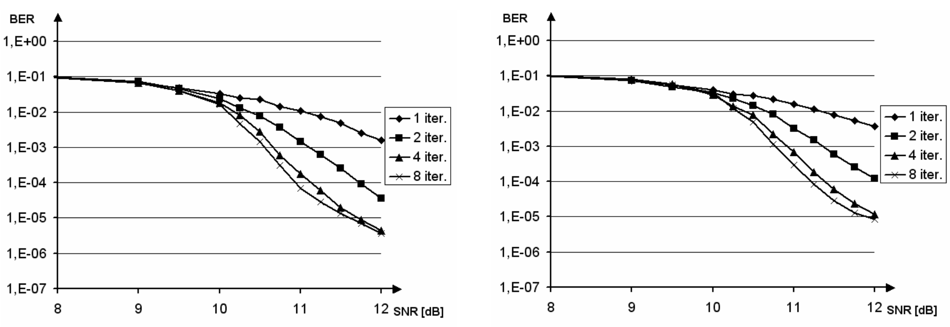

Fig. 3.2. Results for the SOVA algorithm, N = 512, AWGN channel (left) and two-path channel (right)

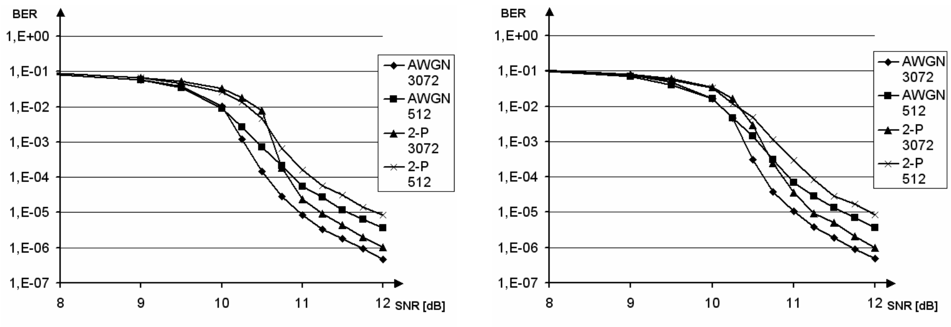

Fig. 3.3. Comparison of results for the MAP algorithm (left)
and for the SOVA algorithm (right)

In Fig. 3.3 on the left we can see the results for the MAP algorithm. The first observation allows us to note that the smaller interleaver size is the worse results we obtain. It takes place either in AWGN channel or in two-path channel. The difference for BER = 10^{-5}, is about 0.75 dB, while for BER = 10^{-3} is about 0.25 dB. This means that for even lower BERs the difference will probably increase. Let us also note that the decoder performance when two-path channel is applied is worse than that for an AWGN channel. For BER = 10^{-2} loss is 0.3-0.5 dB depending on the interleaver size. However, for BER = 10^{-5} the difference decreases to about 0.25dB. Difference between these two channels decreases to zero when the SNR increase.

In Fig. 3.3 on the right we can see the results for the SOVA algorithm. After short analysis we can conclude that the SOVA decoder performs similarly to the MAP decoder.

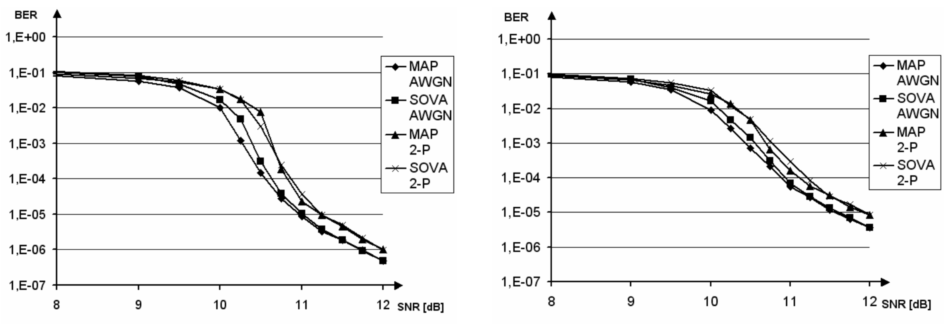

Fig. 3.4. Comparison of results for interleaver size N = 3072 (left)
and for N = 512 (right)

The next step is to compare our results for given interleaver size. This comparison is shown in Fig. 3.4. both for N = 3072 and for N = 512. For a given interleaver size application of he SOVA algorithm instead of the MAP one deteriorates the BER curves. However, this difference is not big and in AWGN channel changes from 0.2 dB for BER = 10^{-3} to 0 dB for BER = 10^{-6}. For two-path channel the difference between MAP and SOVA is imperceptible except SNR=10.5 dB, where the results for SOVA are slightly better. This difference is probably caused by inaccuracy of estimation.

4. CONCLUSIONS

In this paper we have reported the primary results of our studies on application of TTCM in 64 kbit/s data transmission on radio channels with 25 kHz bandwidth. The proposed system makes it possible to achieve BER = 10^{-6} for SNR equals about 12 dB. This value is not constant and depends on the size of applied interleaver. Due to iterative turbo-decoding, the proposed system is very time and memory consuming and many DSP may not meet computation requirements, which are very high. Additionally, we must remember that processor has to not only to decode received data but it has also many other tasks to perform. In order to decrease computation requirements a few solutions can be applied which however result in decrease of the achieved performance. Examples of such solutions are presented in [4] and [5]. This means that the designer of telecommunication system has to decide how large increase of BER can be tolerated.

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