

# Matrix-Interleaved Serially Concatenated Block Codes for Speech Transmission in Fixed Wireless Communication Systems

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**Abstract**—In this paper, we study a class of serially concatenated block codes (SCBC) based on matrix interleavers, to be employed in fixed wireless communication systems. The performances of SCBC-coded systems are investigated under various interleaver dimensions. Numerical results reveal that the matrix interleaver could be a competitive candidate over conventional block interleaver for frame lengths of 200 bits; hence, the SCBC coding based on matrix interleaver is a promising technique to be employed for speech transmission applications in many international standards such as pan-European Global System for Mobile communications (GSM), Digital Cellular Systems (DCS) 1800, and Joint Detection Code Division Multiple Access (JD-CDMA) mobile radio systems, where the speech frame contains around 200 bits.

**Keywords**—Matrix Interleaver, serial concatenated block codes (SCBC), turbo codes, wireless communications.

## I. INTRODUCTION

INTERLEAVED concatenated codes show a great advancement in coding theory for their outstanding performance. Classic turbo codes introduced by Berrou *et al.* in 1993 [1], are based on parallel concatenation of convolutional codes (PCCCs). It has been noted for many PCCCs that at high signal-to-noise ratios (SNRs), the bit-error-rate (BER) curve flattens a little. As one of the methods to obtain low BER performance at high SNRs, turbo codes based on serial concatenation have been proposed in [2]. Serially concatenated block codes (SCBCs) or serially concatenated convolutional codes (SCCCs) are implemented as cascade of a recursive systematic convolutional (RSC) outer encoder, an interleaver scrambles the ordering of the outer codewords's bits, and an RSC inner encoder whose input bit sequences are the time-shifted versions of the outer encoder's codewords. SCBCs can be decoded at the receiver by iterating the *Bahl-Jelinek* algorithm [3] on the convolutional component codes. They achieve remarkable BER performance at low SNRs with this decoding algorithm; however, it is shown that the performance of these codes may not be astonishing at very high SNRs due to the domination by effective free distance ( $d_{min}$ ) [4]. This BER flattening at high SNRs (i.e. where increase in transmit power does not lead to a significant reduction of the BER) could

be enhanced, by increasing component code's constraint length, increasing interleaver size, or designing proper permutation pattern for interleaver's mapping. For speech communications, using a large constraint length in component codes, and employing large interleaver sizes are not desirable due to the substantial increase in processing and transmission times, while the increase in  $d_{min}$  can be realized through an appropriate choice of the interleaver permutation pattern without introducing further delays [5].

In this paper, to provide remarkable BER performance without increasing the latency, effective free distance ( $d_{min}$ ) is increased through the utilization of matrix interleavers. It will be shown that this system outperforms the system based on conventional block interleavers for typical frame size in speech transmission standards.

## II. SCBC-CODED WIRELESS COMMUNICATION SYSTEMS

### A. SCBC Coding

The schematic of a rate 1/3 SCBC encoder is depicted in Fig.1 (a). It is composed of two cascaded RSC convolutional component encoders, the outer encoder with rate 1/2 and the inner encoder with rate 2/3, linked by an interleaver of length  $N$ . The interleaver is used to spread out the error bursts exhibited during the multipath propagation in wireless channels. A crucial step in the design of turbo codes which plays a key role in its BER performance is the design of interleaver's permutation pattern. In this case, we used Matrix interleaver which permutes input bit sequences by filling a matrix row by row and reading out column by column as shown in Fig. 2. At the receiver, the de-interleaver writes the information column-wise and reads out row-wise. Denoting the column size (also called depth) and row size (also called span) by  $n$  and  $m$  respectively, the matrix interleaver is referred to as  $(n,m)$  interleaver.

It is worthwhile to mention here that for low frame sizes, conventional block interleavers shown to have superior performance than the random (pseudo-random) interleavers [6]. In section III, we will show that matrix interleavers outperform block interleavers for such frame lengths.

The SCBC decoder is depicted in Fig.1 (b). The core of the decoding procedure is the Maximum a posteriori probability (MAP) decoding based on *Bahl-Jelinek Algorithm* [3]. Each

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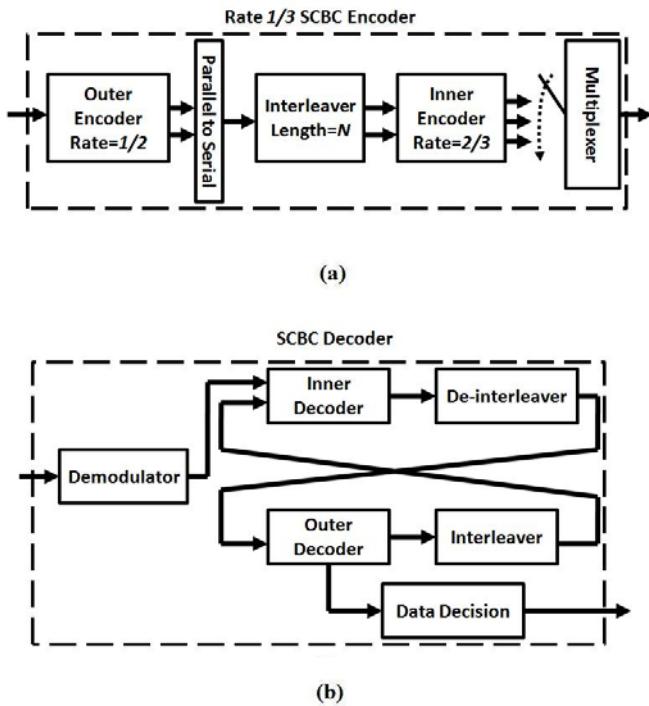


Fig.1. Structure of the SCBC. (a) Encoder. (b) Decoder.

convolutional component decoder is separately decoded using the most recent decoding information from the other component decoder, known as extrinsic information  $L(u_k)$ . This extrinsic information is a posteriori log-likelihood ratio (LLR) of the systematic bit, which is computed by each component decoder, and can be regarded as a type of diversity in that it can refine the decoder outputs in each iteration [7]:

$$L(u_k) = \log \frac{P(u_k=1|y_1^N)}{P(u_k=0|y_1^N)} \quad (1)$$

Where  $u_k$  is the information bit value (0 or 1) at time  $K$ ,  $y_1^N$  is the decoder output sequence, and  $P(u_k = 0, 1)$  is a posteriori probability of bit value 0 or 1. The iterative process continues until a stopping criterion (based on the cross entropy between the distributions of estimated outputs of the decoders at each iteration [8]-[9], or in some other schemes [10] based on the signature code of estimated outputs) is fulfilled.

### B. International Standards for Speech Transmission

To successfully use SCBCs for speech transmission applications in wireless communication systems, the interleaver size must be designed in such a way that the maximum latency in speech transmission is not exceeded. As far as the main constraint for the application of SCBCs in real-time speech transmissions is their delay, they must take into account the size of speech frames in order to avoid a further buffering delay penalty (it should be noted that since the interleaver's input is the outer code's output in SCBCs, the interleaver size is  $2 \times \text{frame size}$  in this case). In speech transmission which is typically based on finite frames, the

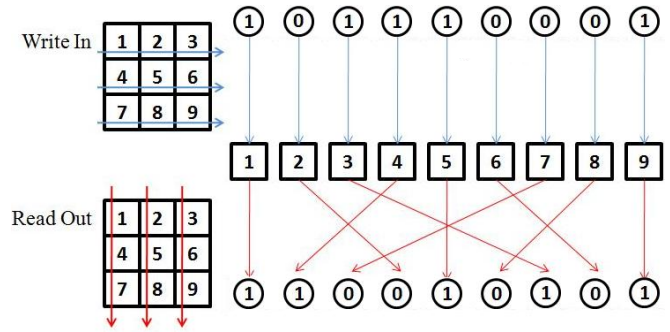


Fig.2. Operation of (3,3) matrix interleaver for interleaver size  $N=9$ .

speech frames contain less than 200 bits to be processed by the channel encoder [11]. In many international standards, such as pan-European Global System for Mobile communications (GSM) and Digital Cellular System (DCS) 1800 [12], speech frames contain 189 bits. In a case of joint detection code division multiple access (JD-CDMA) mobile radio systems, speech frames contain 192 bits [13]. Moreover, in the case of DS-SS digital cellular systems known as Interim Standard (IS-95), speech frames contain 192 bits [14].

### C. Propagation Environment

Wireless communication takes place over multipath fading channels. The transmitted signal travels through different paths with different reflections from a transmitter to a receiver. Depending on the propagation environment, the line-of-sight (LOS) paths in radio link may exist, or might be blocked by different objects such as high buildings, trees, hills, or mountains. The terms Rayleigh and Rician fading refer to a particular type of statistical response curve that mathematically describe the variation of the received signal's amplitude, caused by the characteristics of the transmission path. When LOS exists, Rician distribution is used to describe the probability density function (PDF) of the received signal's amplitude. Meanwhile, in a case of blocked LOS, Rayleigh distribution is used. In Rician fading, the ratio of the power in the LOS component to the power in the diffuse component is referred to as  $K$  factor. The stronger the LOS component, the rarer the occurrence of deep fades [15]. With increasing the direct ray,  $K$  increases, and for very large  $K$ , it approximates a Gaussian distribution. Obviously, for  $K \rightarrow 0$  (i.e. almost no direct ray), the Rician distribution becomes Rayleigh.

For mobile wireless communication systems, the rapid movement of receiver in a vehicle toward or away from the transmitter introduces a signal frequency change called a Doppler shift, which causes the spectral broadening of the received signal. We also have a Doppler shift in fixed wireless communication systems due to the scatter components. Different statistical models are used to specify the amount of spectral broadening of the received signals in different environments. In this work, we use Rounded power spectral density (PSD) [16] for statistical modeling of Doppler spectrum in fixed wireless communication systems.

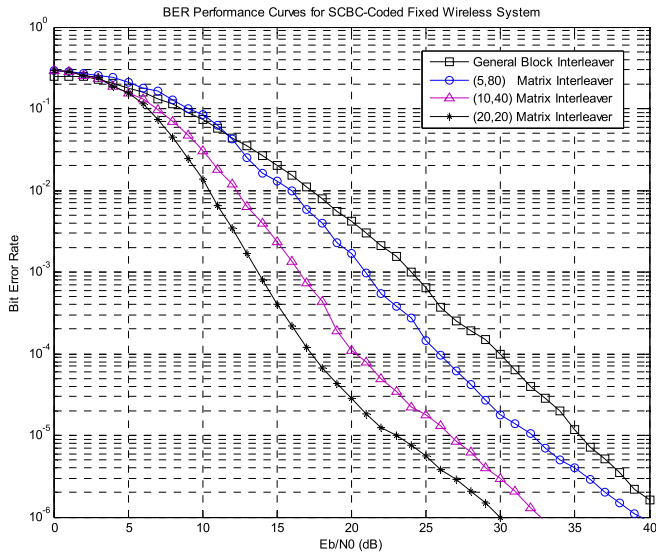


Fig. 3. Bit error probability versus energy per bit/noise power spectral density ( $E_b/N_0$ ) for matrix and conventional block interleavers with DBPSK, over Rayleigh-multipath fading channel with Rounded Doppler PSD ( $f_d \ll 10\text{Hz}$ ).

### III. SIMULATION RESULTS

With the aid of aforementioned discussions, the performance evaluation of SCBC-coded fixed wireless communication systems based on matrix interleavers with various dimensions are evaluated numerically in this section, and the results are compared with the similar system based on conventional block interleaver. The modulation format used for the transmission over wireless channel is chosen to be differential binary-phase-shift-keying (DBPSK) signaling. Due to the robustness of differential PSK modulations over coherent PSK, they are usually preferred choice for high Doppler environments with deep fades [17]. In addition, they preserve the signal's amplitude against amplitude nonlinearities and provide reasonable spectral efficiency, compared with the amplitude-based and frequency-based signaling formats. The SCBC uses RSC component codes, in which a rate 1/2 outer code employs 8-state RSC component code with a generator polynomial matrix  $G(17,13)$ , and a rate 2/3 inner code, consists of two 8-state component codes with  $G(17,13,0)$  and  $G(0,17,17)$  generator polynomial matrices, resulting overall rate of 1/3. A moderate code rate (1/3) is used in this work to guarantee both spectral and power efficiencies. Assuming the omni-directional antennas are deployed in both ends of the link, the performance of SCBC system with the parameters discussed previously is simulated based on a worst-case scenario where the LOS is blocked; thus, Rayleigh distribution is used for multipath fading modeling and Rounded PSD is used for Doppler spectrum modeling. In this case, we assumed Doppler shift of  $f_d \ll 10\text{Hz}$  at the receiver. Moreover, the additive noise introduced by hardware components at the front end receiver is assumed to be additive-white-gaussian-noise (AWGN).

Fig. 3 shows the bit error probability versus the energy per

bit/noise power spectral density ( $E_b/N_0$ ) for the discussed SCBC-coded Fixed wireless system. The *Log-MAP algorithm* [18] with 12 decoding iterations is used at the decoder. According to the discussions on the speech frame sizes of international standards, the *frame size=200* is chosen to provide generic results; hence, the interleaver size will be  $N=400$ . It is seen from the results that by using various matrix dimensions, the system based on matrix interleavers outperforms system based on block interleaver. Although the pseudo-random interleavers shown to have superior performance than the block interleavers for large frame sizes, in many cases their inferior performance in low frame sizes have been confirmed as in [6]. Therefore, the results imply that for low frame sizes around the typical international speech frame standards, considerable BER performance improvement can be achieved without changing the hardware requirements and introducing further delays by using the matrix interleaver instead of the other conventional interleaver types. In particular, with large and comparable depth and span, remarkable power efficiency improvement of up to 9 dB could be achieved at  $BER = 1.5 \times 10^{-6}$  by using (20,20) interleaver instead of (5,80) interleaver. It should be noted that we can achieve lower bit error values at such moderate SNRs (i.e.  $E_b/N_0$ ), by using higher interleaver sizes, employing component code with higher constraint length, and using original MAP algorithm (without approximation), by relaxing the time constraints, which is not desirable in most speech applications.

### IV. CONCLUSION

For the typical speech frame sizes, the matrix interleavers have shown a superior performance over the most well-known and the most-used interleaving schemes; thus, the turbo coding based on serial concatenation in cooperation with matrix interleavers, was presented for fixed wireless communication systems, taking into account the effect of AWGN at the receiver front end, and considering Rayleigh-faded wireless channel.

The performance of the discussed system could be further investigated for the other communication systems with different channel conditions such as mobile wireless communication systems, and multiple-input multiple-output (MIMO) channels.

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