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Super-Orthogonal Space-Time Turbo Transmit Diversity for CDMA

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Studies have shown that transmit and receive diversity employing a combination of multiple transmit-receive antennas (given ideal channel state information (CSI) and independent fading between antenna pairs) will potentially yield maximum achievable system capacity. In this paper, the concept of a layered super-orthogonal turbo transmit diversity (SOTTD) for downlink direct-sequence code-division multiple-access (CDMA) systems is explored. This open-loop transmit diversity technique improves the downlink performance by using a small number of antenna elements at the base station and a single antenna at the handset. In the proposed technique, low-rate super-orthogonal code-spread CDMA is married with code-division transmit diversity (CDTD). At the mobile receiver, space-time (ST) RAKE CDTD processing is combined with iterative turbo code-spread decoding to yield large ST gains. The performance of the SOTTD system is compared with single- and multiantenna turbo-coded (TC) CDTD systems evaluated over a frequency-selective Rayleigh fading channel. The evaluation is done both by means of analysis and computer simulations. The performance results illustrate the superior performance of SOTTD compared to TC CDTD systems over practically the complete useful capacity range of CDMA. It is shown that the performance degradation characteristic of TC CDTD at low system loads (due to the inherent TC error floor) is alleviated by the SOTTD system.

Keywords and phrases: transmitter diversity, space-time coding, code-division transmit diversity, layered super-orthogonal turbo transmit diversity, low-rate spreading and coding, CDMA wireless communications.

1. INTRODUCTION

Space-time (ST) processing techniques, such as receive diversity and antenna beamforming, can significantly improve the downlink and uplink capacity of cellular direct-sequence (DS) code-division multiple-access (CDMA) systems. Recent studies have explored the limits of multiple-antenna systems performance in frequency-selective multipath fading environments from an information-theoretic point of view [1, 2]. It has been shown that, with perfect receiver channel state information (CSI) and independent fading between pairs of transmit-receive antennas, maximum system capacity may potentially be achieved. When multiple receive antennas are not available, multiple transmit antennas have been proven to be an alternative form of spatial diversity that may significantly improve spectral efficiency. Other forms of transmit diversity, such as antenna selection, frequency offset, phase sweeping, and delay diversity, have been studied extensively [3, 4, 5]. Recently, space-time (ST) coding was proposed as an alternative solution for high data rate transmission in wireless communication systems [6, 7, 8, 9, 10].

Depending on whether feedback information is utilized or not, transmit diversity schemes are usually categorized as being either closed- or open-loop methods. In closed-loop schemes, CSI estimated by the receiver is fed back to the transmitter, allowing for a number of different techniques to be considered. These techniques, such as beamforming, adaptive antenna prefiltering, or antenna switching, are used to maximize the signal-to-noise ratio (SNR) at the receiver [11, 12]. When no feedback information is available, the temporal properties of the propagation environment and the transmission protocol can be used to improve the receiver's performance. Techniques utilizing these kinds of properties are commonly referred to as open-loop methods.

Foschini [2] has considered an open-loop layered space-time (ST) architecture with the potential to achieve a significant increase in capacity compared to single-channel systems. The spectrally efficient layered ST transmission process basically comprises the demultiplexing of a single primitive input data stream into n multiple equal-rate data streams. The n separately coded, chip-symbol-shaped and modulated data streams then individually drive separate multiple transmit antennas elements prior to radiation. A multiple-transmit multiple-receive ($M_T = n$, $M_R = n$)-antenna analysis (where M_T and M_R, respectively, denote the number of transmitter and receiver antenna elements) showed that the system capacity increased linearly with n, despite the random interference of the n received waves. With $n = 8$, an 1% outage probability, and 21 dB average SNR at each receiving antenna element, a spectral efficiency of 42 bps/Hz was shown to be achievable [2]. This implies a capacity increase of 40 times that of a $(M_T, M_R) = (1, 1)$ system at the same total radiated transmitter power and bandwidth. The layered ST concept basically relates to the exploitation of all available spatial and temporal dimensions provided by the layered combination of multielement transmit and/or receive antenna arrays and a vast range of available one-dimensional coding techniques to achieve maximum diversity gain through iterative processing at the receiver. For a detailed description and some illustrative examples of the layered ST architecture employing convolutional coding, as opposed to parallel concatenated iterative super-orthogonal turbo coding on each ST branch proposed in this paper, the interested reader is referred to references [2, 13].

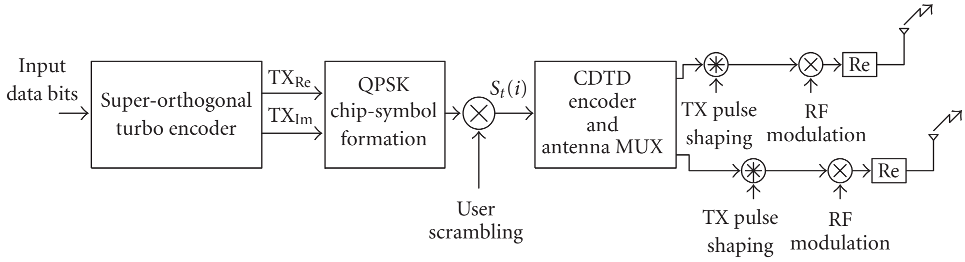
This layered ST architecture forms the basis for the class of orthogonal decomposable coded ST codes presented in this paper. The Alamouti ST block codes are members of this class of codes [3, 6]. The condition of statistically independent (uncorrelated) fading, to maintain orthogonality, is seldom achieved in practice due to the scattering environment around the mobile and base station. However, decomposition or separation of the multiantenna channel into a number of nearly independent subchannels can be realized, provided that CSI is available at the receiver [2, 12]. Maximizing the free distance of the ST coded symbols transmitted over these nearly independent spatially separated channels, a spatial-temporal coding diversity gain can be achieved, referred to as space-time gain (STG).

DS-CDMA systems exhibit maximum capacity potential when combined with forward error correction (FEC) coding [14]. In CDMA, the positive tradeoff between greater distance properties of lower rate codes and increased cross-correlation effects (due to shorter sequence length) is fundamental to the success of coded CDMA. Most FEC systems, especially those with low code rates, expand bandwidth and can be viewed as spreading systems. It has been illustrated that the maximum theoretical CDMA capacity can only be achieved by employing very low-rate FEC codes utilizing the entire bandwidth, without further spreading by the multiple-access sequence [14, 15, 16]. These are known as code-spread CDMA systems.

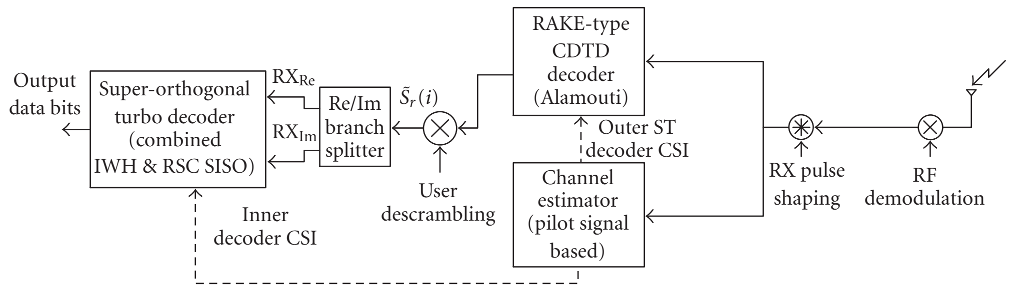
Viterbi [17] has proposed the use of orthogonal convolutional codes as low-rate coding extensions for code-spread CDMA. Recently, two new classes of low-rate codes with improved performance have been proposed. Pehkonen and Komulainen [18, 19] proposed a coding scheme that combines super-orthogonal turbo codes (SOTC) with super-orthogonal convolutional codes (SOCC) [17]. A different approach was taken by Frenger et al. [15, 16], where a class of nested rate-compatible convolutional codes (RCCC), with maximum free distance (MFD), was derived and applied to code-spread CDMA.

For nonoptimum multiuser receivers, such as the matched filter (MF) or RAKE, coding gain comes at the cost of an increased multiple-access interference (MAI) level. Note that as the spreading factor (SF) decreases, so does the potential number of users that can be accommodated, due to the smaller spreading sequence family size available. In such a case, the Gaussian approximation of the MAI does not apply, as the central limit theorem does not hold any more. However, when transmit diversity is considered, this situation (from a coding perspective) is improved due to the introduction of additional MAI as a result of the multiple transmission paths that are created through the application of the multiple transmit antenna diversity concept. Especially, when turbo coding is considered, the coding gain potential becomes significant. For a finite effective code rate (and hence a finite spreading ratio), the level of MAI, under additive white Gaussian noise (AWGN) and equal power conditions, is fixed. For a RAKE receiver with perfect CSI, the soft-input soft-output (SISO) turbo-based decoder will perform equally well in AWGN and fully interleaved multipath fading channels.

In this paper, a layered ST super-orthogonal turbo transmit diversity (SOTTD) architecture for a downlink DS-CDMA system, operating over a frequency-selective fading channel, is investigated. This open-loop transmit diversity technique is well-suited for code-spread CDMA systems where downlink performance is improved by using a small number of transmit antennas ($M_T = 3$) at the base station and a single antenna ($M_R = 1$) at the mobile handset receiver. In the proposed technique, low-rate super-orthogonal code-spread CDMA is married with code-division transmit diversity (CDTD), and at the mobile receiver, ST RAKE CDTD processing is combined with iterative turbo code-spread decoding. In Section 2, the description of the SOTTD code-spread CDMA system is presented. In Section 3, the performance of the SOTTD system is compared with single- and multiantenna turbo-coded (TC) CDTD systems. The evaluation is done by both analysis and computer simulations. Section 4 concludes the paper.



(a)



(b)

FIGURE 1: SOTTD system block diagrams: (a) transmitter, $M_T = 2$; (b) receiver, $M_R = 1$.

2. SYSTEM DESCRIPTION

2.1. Transmitter and receiver

A downlink (base station to mobile handset) dual transmit, $M_T = 2$, and single receive antenna, $M_R = 1$, multiuser DS-CDMA-based communication system with K simultaneous users is considered. The general structures of the DS-CDMA transmitter and receiver under investigation are illustrated in Figure 1.

With reference to Figure 1a, the transmitter consists of the following modules:

(1) super-orthogonal turbo encoder producing complex code-spread sequences for CDMA;
(2) Gray-coded quadrature-phase-shift-keyed (QPSK) chip-symbol formation;
(3) user-specific scrambling of QPSK chip-symbols, for example, using a IS-95-like long pseudonoise (PN) scrambling sequence;
(4) code-division transmit diversity (CDTD) encoder based on Alamouti ST block encoder and antenna multiplexer [3];
(5) transmitter chains for M_T transmit antennas, each comprising chip-pulse shaping, RF modulation, and antenna transmission of RF-modulated (real-part only) signals for the M_T transmit antennas.

With reference to Figure 1b, the receiver consists of the following blocks:

(1) receiver chain for $M_R = 1$ receive antenna, comprising RF demodulation and chip-pulse shaping;
(2) channel estimation providing the fading coefficients for each of the transmitter antennas through the transmission of known pilot signals;
(3) RAKE-type ST receiver based on the Alamouti ST block decoder and maximal-ratio combining (MRC);
(4) user-specific descrambling of QPSK chip-symbols;
(5) splitting of the spread-coded chip sequence into real and imaginary components;
(6) super-orthogonal SISO-based turbo decoder.

In the following paragraphs, details concerning the super-orthogonal turbo encoder and decoder, as well as the ST RAKE CDTD decoder, will be given.

2.2. Super-orthogonal turbo encoder description

The detailed structure of the super-orthogonal turbo encoder is shown in Figure 2. The heart of the encoding scheme is formed by the $Z = 2$ rate-(1/16) constituent encoders, consisting of the combination of a rate-(1/4) recursive systematic convolutional (RSC) encoder, a rate-(4/16) Walsh-Hadamard (WH) encoder, parallel-to-serial (P/S) converter, and puncturing modules. A definition and description of the iterative generation of WH codes, together with their correlation properties, are given in Proakis [20, Chapter 8, pages 424–425]. The combined encoder is referred to as the super-orthogonal RSC&WH encoder. These encoders are concatenated in parallel. A binary data sequence of length N is fed into the encoder. The first encoder processes the original data sequence, whereas before passing through the second encoder, the data sequence is permuted by a pseudorandom interleaver of length N. The outputs of the rate-(1/4)

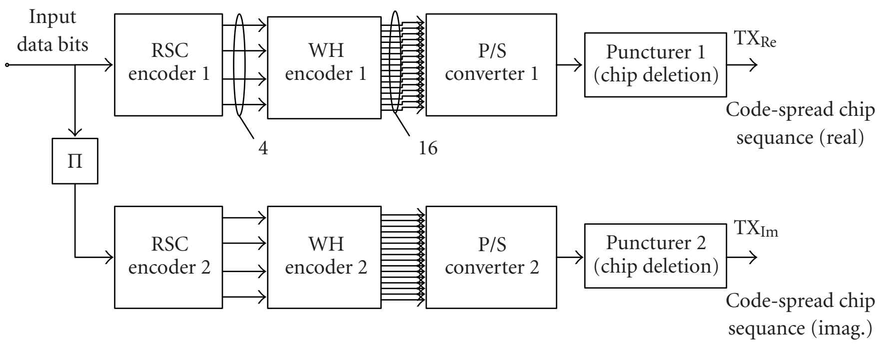


FIGURE 2: Super-orthogonal turbo encoder for $Z = 2$ constituent RSC&WH encoders.

RSC encoder is fed to the rate-(4/16) WH encoder, producing a sequence of length $L_{\mathrm{WH}} = 16$ from a set of 16 sequences. By combining the constituent encoder outputs, the code rate from the turbo encoder before puncturing is $R_c = 1/(ZL_{\mathrm{WH}}) = 1/32$.

Figure 3 depicts the rate-(1/4), 8-state RSC encoder block diagram and associated trellis diagram. The trellis diagram is important in the evaluation of code distance properties and for Viterbi decoding.

As a last stage of encoding, after P/S conversion, the outputs of the two constituent RSC&WH encoders are punctured to produce the code-spread chip sequences ($\mathrm{TX}_{\mathrm{RE}}, \mathrm{TX}_{\mathrm{IM}}$). The puncturing (chip deletion) operation can be seen as a form of rate matching to provide a wide range of spread-code rates. Note that the final code rate of the super-orthogonal turbo encoder determines the code-spread factor, G, where $G \leq 1/R_c$, in general. In the case of no puncturing, $G = 1/R_c = 2L_{\mathrm{WH}} = 32$.

The complex chip output sequences of the super-orthogonal turbo encoder is Gray-mapped into a QPSK symbol constellation. The in-phase (I) and quadrature (Q) QPSK chip-symbol sequences are complex-scrambled with a user-specific IS-95-like long complex pseudonoise (PN) scrambling sequence. The complex result of this complex scrambling process, $S_t(i)$, is fed to a code-division transmit diversity (CDTD) block encoder based on the Alamouti ST block encoder and antenna multiplexer [3, 6].

The CDTD encoder in Figure 1a maps two symbols into an orthogonalising (2×2) code matrix according to

$$D_{M_T} = \begin{bmatrix} s_t(2i-1) & s_t(2i) \\ -s_t^*(2i) & s_t^*(2i-1) \end{bmatrix}, \qquad (1)$$

where $i = 1, 2, \ldots$. The symbol $s_t(n)$ denotes the transmitted QPSK chip-symbol for time instant n.

Finally, the real part of the complex transmit pulse-shaped and RF-modulated outputs of the CDTD encoder are radiated from multiple transmit antennas, as shown in Figure 1a. In this way, low-rate super-orthogonal code-spread CDMA and an open-loop code-division transmit diversity (CDTD) technique have been combined to potentially facilitate significantly improved downlink performance through appropriate iterative ST receiver and decoder processing.

2.3. *Space-time RAKE CDTD receiver/decoder description*

Figure 4 shows the general architecture of the RAKE-type CDTD ST receiver for the SOTTD system.

It has been shown that the conditions where ST decoding yields significant diversity gains are independent of those conditions that are favorable for a RAKE-type receiver [21]. In other words, ST diversity is not adversely affected by suboptimal multipath diversity gain. Under the best conceivable conditions, the multipath components have equal expected power and arrive such that the delayed spreading codes are perfectly orthogonal. Then an L_R-finger RAKE receiver, where $L_R = J$ denotes the number of resolvable paths, would be equivalent to having J receiver antennas, and both the diversity and the expected SNR would theoretically be increased by the factor J [12, 21].

In order to maximize multipath diversity gain, the following assumptions are made.

(1) The J paths from antenna m experience independent Rayleigh fading, expressed through the channel coefficients, h_{jm}, $j = 1, 2, \ldots, J$ and $m = 1, 2$.
(2) Each pair of paths from the two transmitter antennas arrives with the same set of delays at the receiver antenna. (This assumption is justified by the fact that in the cellular personal communication frequency bands, the propagation delay between the two transmitter antenna elements is measured in nanoseconds, while the multipath delays are measured in microseconds [12]).
(3) Path delays are approximately a few chips in duration and small compared with the symbol period so that intersymbol interference can be neglected.

Multipath RAKE and ST decoding is performed on knowledge of the multipath delays and fading coefficients for each of the M_T transmitter antennas and J possible multipaths. This information is provided to the mobile receiver by the channel estimator block.

The channel estimator operates on the principle of pilot signals transmission. Increasing the number of transmitter antennas tends to give greater diversity gains, but if the total pilot power is fixed, the individual estimates for the fading coefficients deteriorate, and crosstalk increases among

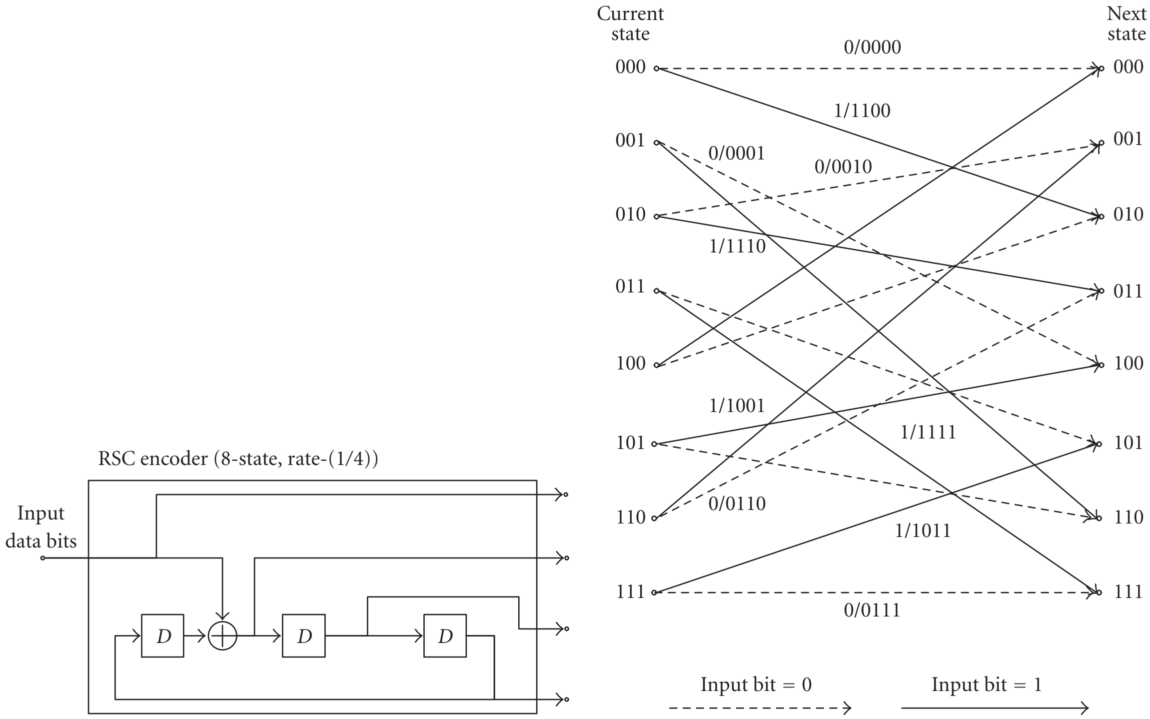


Figure 3: Constituent RSC encoder: (a) encoder's block diagram; (b) encoder's trellis diagram.

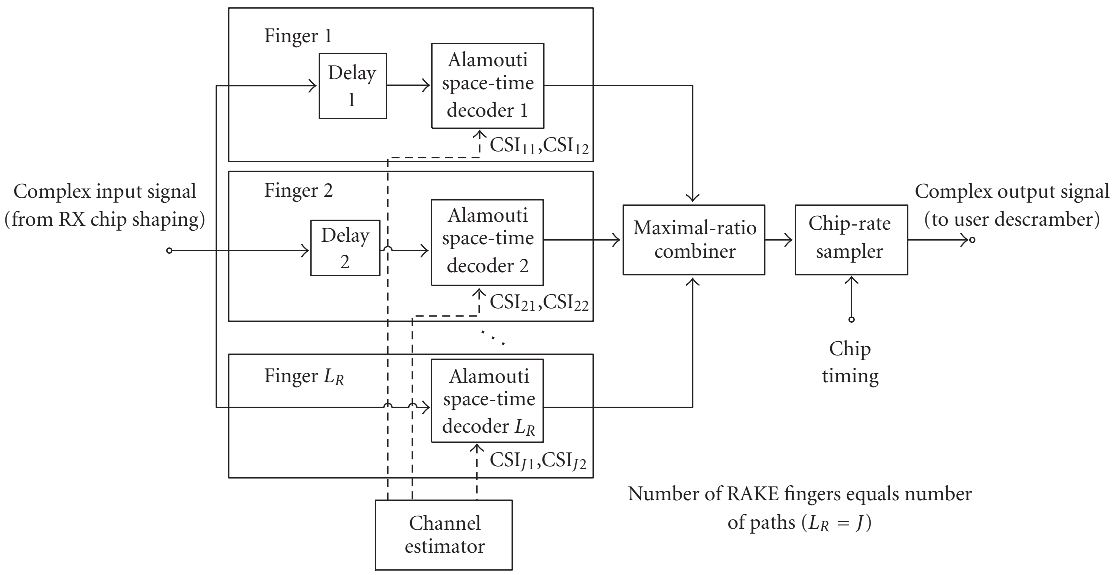


Figure 4: RAKE-type CDTD space-time (ST) receiver based on Alamouti ST block decoder.

the subchannels. Adding extra antennas requires the incorporation of additional pilot signals to enable the mobiles to accurately estimate the multiple-antenna propagation coefficients. As a rule of thumb, the individual powers of these pilot signals should be inversely proportional to the number of transmit antennas. In this paper, perfect CSI is assumed, and the channel estimation error-related RAKE ST receiver problems are not treated here.

From Figure 4, it can be seen that paths $j = 1, 2, \dots, J-1$ are delayed before ST decoding is attempted. This path delay should be equal to the time-of-arrival difference between path j and the last path J, and is done to synchronize individual path powers for maximal-ratio combining (MRC).

The ST decoder shown in Figure 4 is based on the Alamouti ST length-two block encoder and decoder [3, 6]. Recall that the encoder mapped two symbols into a (2×2) code

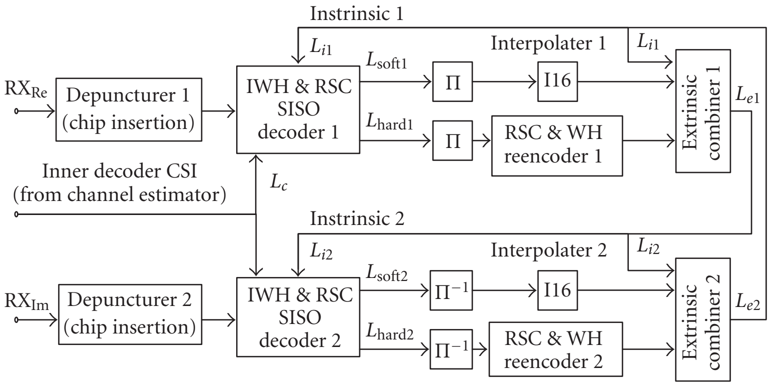


FIGURE 5: Super-orthogonal turbo decoder.

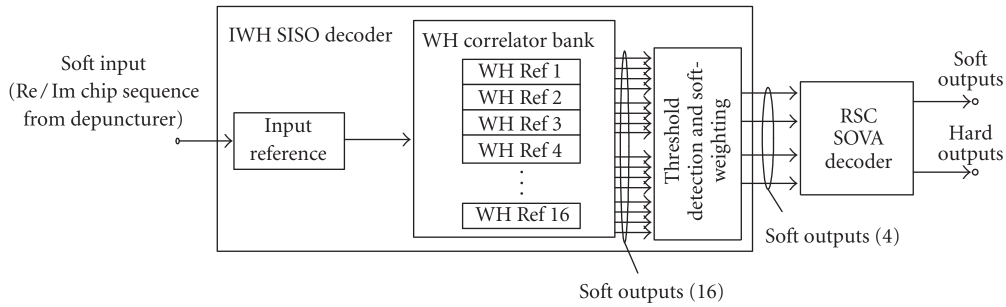


FIGURE 6: Combined inverse Walsh-Hadamard (IWH) and recursive systematic convolutional (RSC) soft-input soft-output (SISO) decoder.

matrix according to (1). Since the symbols are also orthogonal across antennas, the soft-input block decoder simply calculates:

$$\begin{aligned} s_r(2i-1) &= h_{j1}^* r(2i-1) + h_{j2} r^*(2i), \\ s_r(2i) &= h_{j2}^* r(2i-1) + h_{j1} r^*(2i). \end{aligned} \tag{2}$$

In (2), $r(n)$ denotes the received chip symbols. $s_r(n)$ is the block-decoder soft output for time instant n that determines to which quadrant in the QPSK constellation the chip symbols most likely belong. The likelihood (or confidence level) of this determination is the soft output passed on to the channel decoder after MRC.

Given perfect multipath and diversity gain, the RAKE-type ST decoder has a combined multipath and ST diversity gain of $L_R M_T$, where $L_R = J$ denotes the number of received signal paths (which are here assumed to be equal to the number of fingers employed in the RAKE receiver structure) and M_T denotes the number of transmit antennas.

2.4. Super-orthogonal turbo decoder description

Figure 5 shows the general architecture for the super-orthogonal iterative turbo decoding strategy.

Before the actual decoding takes place, for those chips that were punctured (deleted), zero values are inserted. Therefore, the decoder regards the punctured chips as erasures. The iterative decoding of the turbo coding scheme requires two component decoders using soft code-spread chip inputs and providing soft outputs. Two SISO decoders are employed in the component decoders as shown in Figure 6. The first is a SISO inverse Walsh-Hadamard (IWH) decoder and the second a SISO RSC decoder, based on the soft-output Viterbi algorithm (SOVA). Details concerning the actual decoding process will now be given, with reference to Figure 5.

Let RX_{Re} and RX_{Im} be the associated received and demodulated branch code-spread chip sequences with L_c the corresponding reliability values of the CSI. The decoder accepts a priori values $L_i(b)$ for all the information bit sequences and soft-channel outputs $L_c \cdot RX_{Re}$ and $L_c \cdot RX_{Im}$. In the IWH SISO decoder, the branch metric calculation is performed very efficiently by using soft-outputs based on the IWH transformation, which basically correlates the received soft chip-spread sequences, RX_{Re} and RX_{Im}, with the branch WH sequences. The soft outputs from the IWH SISO decoder are passed to the RSC SOVA, which produces hard (L_{hard}) and soft (L_{soft}) outputs. Without loss of generality, the indices, $z = 1, 2$, denoting the constituent component decoders (shown in Figure 5), have been omitted in this discussion.

The IWH&RSC SISO component decoders delivers a posteriori soft outputs $L(\hat{b})$ for all the information bits and extrinsic information $L_e(b)$. The latter is only determined for the current bit by its surrounding bits and the code constraints. It is therefore independent of the intrinsic information and the soft output values of the current bit.

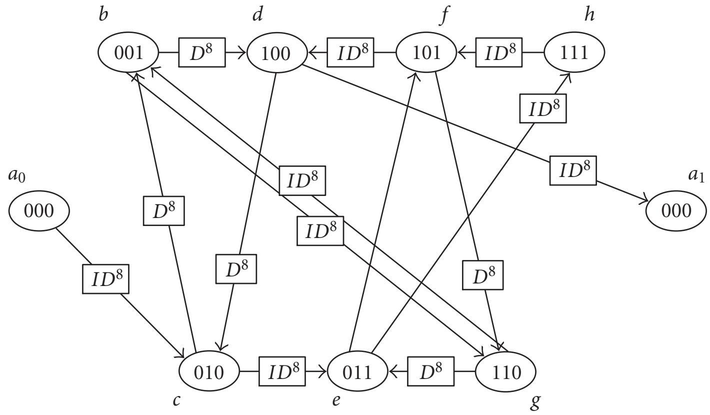


Figure 7: State diagram of combined RSC&WH constituent encoder. (Note that the state transitions are determined by RSC encoder (shown in Figure 3), while output-word Hamming distances are determined by the WH encoder.)

The extrinsic information is given by

$$L(\hat{b}) = [L_{\text{hard}} \otimes f(L_{\text{hard}}) + g(L_{\text{soft}})] - L_i(b), \tag{3}$$

where the first term, $L_{\text{hard}} \otimes f(L_{\text{hard}})$, is the reencoded chip code-spread sequence, with $f(L_{\text{hard}})$ being the function denoting the combined RSC and WH reencoding process. The symbol "$\otimes$" denotes convolution. The second term, $g(L_{\text{soft}})$, represents the interpolated soft outputs from the component SISO decoders, with interpolation factor $L_{\text{WH}} = 16$. It is important to note that all the above-mentioned sequences are vectors of length $L_{\text{WH}} = 16$.

The log-likelihood ratio (LLR) soft output of the decoder for the information bit b is written as

$$L(\hat{b}) = [L_c(\text{RX}_{\text{Re}} + \text{RX}_{\text{Im}}) + L_i(b)] + L_e(\hat{b}) \tag{4}$$

implying that there are three independent estimates that determine the LLR of the information bits, namely, the a priori values, $L_i(b)$, the soft-channel outputs of the received sequences, $L_c \cdot \text{RX}_{\text{Re}}$ and $L_c \cdot \text{RX}_{\text{Im}}$, and the extrinsic LLR's $L_e(\hat{b})$.

At the commencement of the iterative decoding process, there usually are no a priori values $L_i(b)$; hence the only available inputs to the first decoder are the soft-channel outputs obtained during the actual decoding process. After the first decoding process, the intrinsic information on b is used as independent a priori information at the second decoder. The second decoder delivers a posteriori information, which is an output produced by the first decoder too. Note that initially the LLRs are statistically independent. However, since the decoders directly use the same information, the improvement through the iterative process becomes marginal, as the LLRs become progressively more correlated.

It is important to note that the constituent RSC&WH encoders may produce similar WH codewords. Since these codewords are transmitted over different antennas, the full-rank characteristic of the system is still guaranteed. Under multipath fading scenarios, some of the orthogonality will be destroyed. The latter is not a function of the specific WH codeword transmitted at the different antennas, but rather dependent on the delay spread of the channel. Transmitting the same WH codewords over different antennas will have an effect on the channel estimation and initial synchronisation procedures.

3. PERFORMANCE EVALUATION

3.1. Union-bound BEP derivation of combined RSC and WH code

One of the objectives of this section is to shed some light on the contribution of the parallel-concatenated WH codes to the overall SOTTD systems performance. Towards this end, an upper bound is derived for the average bit error probability (BEP) performance of parallel-concatenated WH codes, stemming from the characteristics of the combined RSC&WH code.

The performance of the SOTTD system depends not on the distance properties of the WH code, but actually on the distance properties of the combined RSC&WH code. In this context, the most important single measure of the code's ability to combat interference is $d_{\min}$. Figure 7 depicts the modified state diagram of the RSC&WH constituent code under consideration. The state diagram provides an effective tool for determining the transfer function, $T(L, I, D)$, and consequently, $d_{\min}$ of the code. The exponent of D on a branch describes the Hamming weight of the encoder corresponding to that branch. The exponent of I describes the Hamming weight of the corresponding input word. L denotes the length of the specific path.

Through visual inspection, the minimum distance path, of length $L = 4$, can be identified as $a_0 \to c \to b \to d \to a_1$. This path has a minimum distance of $d_{\min} = 4 \times 8 = 32$ from the all-zero path, and differs from the all-zero path in 2 bit inputs.

Given an $(32, 1)$ RSC&WH constituent code, its input-redundancy weight enumerating function (IRWEF) is used to characterize the complete encoder [22]. The IRWEF makes implicit in each term of the normal weight enumerating function the separate contributions of the information and the parity-check bits to the total Hamming weight of the codewords. When the contributions of the information and redundant bits to the total codeword weight are separated, the IRWEF for the constituent RSC&WH code is obtained as

$$A(I,D) = 1 + 4ID^7 + 6I^2D^2 + 4I^3D^5 + I^4D^4. \quad (5)$$

When employing a turbo interleaver of length N, the IRWEF of the new constituent $(n,k) = (32N, N)$ code is given by $A^N(I,D) = [A(I,D)]^N$, for all Z constituent codes (see [22, page 157, equation (5)] for a similar approach), where n denotes the code length and k the number of encoded data symbols in the code word.

To compute an upper bound to the BEP, the IRWEF can be used with the union bound assuming maximum-likelihood (ML) soft decoding. The BEP, including the fading statistics (assumed to be slowly fading), is of the form shown in (6) [14, 21], where σ_{oc} denotes the effective SNR, and S denotes the power of the received signal:

$$P_{b|S} \leq \frac{1}{k} Q\left(\sqrt{d_{\min}\sigma_{\text{oc}}S}\right) \cdot e^{d_{\min}\sigma_{\text{oc}}S} \cdot \frac{\partial A^N(I,D)}{\partial I}\bigg|_{I=D=e^{-\sigma_{\text{oc}}S}}. \quad (6)$$

On an AWGN channel, the total effective output SNR term used in (6) is $\sigma_{\text{oc}} = R_c E_b/N_0$. Assuming that the cellular system is employing omnidirectional antennas, the total output SNR term used in (6) can be determined as in (7) [11, 21]:

$$\sigma_{\text{oc}} = \left(\frac{1}{R_c}\frac{N_0}{2E_b} + \frac{(K \cdot M_T - 1)}{3G}\right)^{-1}. \quad (7)$$

Recall that K denotes the number of simultaneous users, G is the code-spread ratio, and E_b/N_0 is the energy-per-bit-to-noise spectral density ratio. The CDMA normalized system load is given as K/G. Also, if it is assumed that the M_T transmitters have equal power, with constant correlation between the branches, and transmitted over a Rayleigh fading channel, the components of the received power vector $\mathbf{S}$ are identically distributed, with probability density function (pdf) given by (8), with $\zeta = 1 - \rho + \rho M_T L_R$ (see [21, Sections 6.3.2 to 6.3.4, pages 93–98]):

$$\begin{aligned} p_S(S) &= \frac{1}{\Omega^2\Gamma(M_T L_R)}\left(\frac{S}{\Omega^2}\right)^{M_T L_R - 1} \\ &\times \frac{\exp\left(S/(1-\rho)\Omega^2\right) \cdot {}_1F_1\left(1, M_T L_R, \rho M_T L_R S/\zeta(1-\rho)\Omega^2\right)}{\zeta(1-\rho)^{(M_T L_R - 1)}}. \end{aligned} \quad (8)$$

In the above equation, ${}_1F_1(\cdot)$ denotes the confluent hypergeometric function, Ω^2 is the average received path strength, ρ the correlation between transmit or receive branches, and L_R the number of RAKE receiver fingers.

Finally, the BEP is computed using (6) and (7), by averaging (6) over the fading statistics defined in (8).

Table 1: System parameters for analytical and simulation BEP performance analysis.

Parameter	Simulation value
Spreading ratio	$G = 32$
Operating environment	2-path frequency-selective fading
Number of users	$K = 1, 2, \ldots, G$
Number of RAKE fingers	$L_R = J = 2$
Transmit diversity technique	CDTD and SOTTD
Transmit diversity elements	$M_T = 1, 2$ $(\rho = 0)$
Interleaver length	$N = 256$

3.2. Numerical analysis of CDTD and SOTTD CDMA systems

The performance of the proposed super-orthogonal transmit diversity (SOTTD) CDMA system is compared to that of an uncoded, as well as convolutional- and turbo-coded code-division transmit diversity (CC and TC CDTD) CDMA systems. In order to calculate the BEP of the coded CDTD and SOTTD systems, the output SNR should include the transmit diversity interference term as shown in (7).

Using the system parameters outlined in Table 1, the BEP performance of a cellular CDMA system employing the different techniques has been determined numerically. The performance of single and $M_T = 2, 3$ transmit diversity systems are shown in Figure 8. From the curves, it is clear that the superior performance predicted for TC CDTD may be achieved with the SOTTD system over the complete CDMA capacity range. Also of importance is the fact that the performance degradation of TC CDTD at low system loads (due to inherent TC error floor) is alleviated by the SOTC system—hence the superior performance of SOTTD. This is explained in terms of the higher minimum free distance offered by the rate-(1/16) constituent encoders, as opposed to the use of rate-(1/2) constituent encoders in TC systems.

3.3. Simulation results

Monte-Carlo simulations were conducted to verify the BEP bounds presented above. In the computer simulations, a root-raised cosine (RRC) chip-pulse shaping with roll-off factor of $\alpha = 0.22$ was used. The length of the pulse-shaping filter was set to 8 chips, and 4 samples per chip were taken. A single receiver antenna and $J = 2$ resolvable Rayleigh fading multipaths with equal average power were assumed.

For the simulation, perfect synchronization, coherent detection, and perfect channel state information (CSI) estimation are assumed. The simulated fading channel assumed a flat Doppler power spectrum. A mobile velocity of 3 km/h was selected (corresponding to slow fading), producing nearly static fading over the frame (and interleaver) length of $N = 256$ information bits used in the simulations. The individual path gains are assumed nearly constant (quasistatic) during one frame and change independently from one another. The multipath spread was randomized and evenly distributed with a minimum resolution of

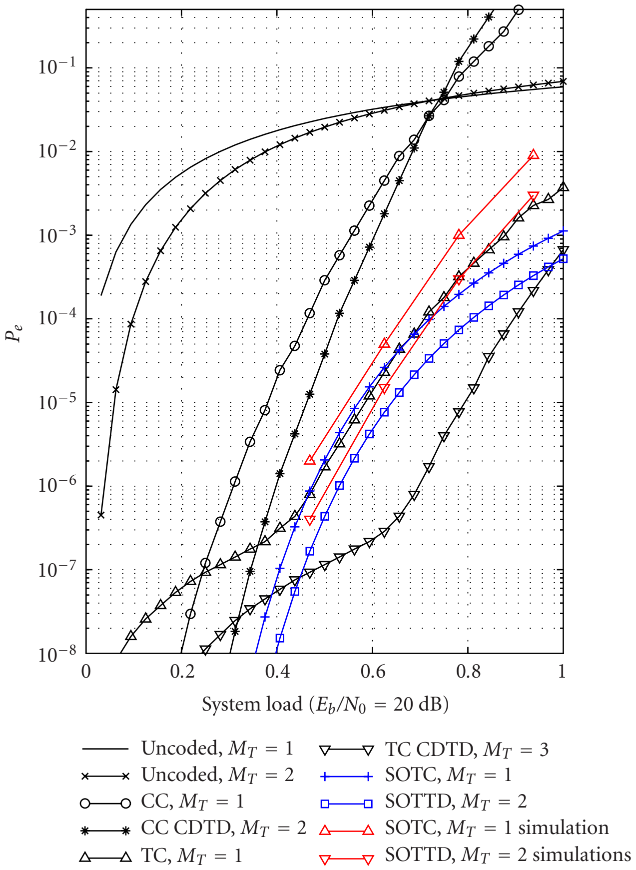


FIGURE 8: Bit error probability as a function of the load (number of users/total spreading = K/G), with the operating point at E_b/N_0 = 20 dB.

one sample. In addition, the turbo decoding configuration for $Z = 2$ constituent codes operates in serial mode, that is, "SISO decoder 1" processes data before "SISO decoder 2" starts its operation, and so on (refer to Figure 5).

Using the system parameters outlined in Table 1, the BER performance of a SOTTD CDMA system has been determined by means of simulation. Figure 8 compares the simulated SOTTD performance with the theoretical performance bounds of convolutional and turbo-coded CDTD. E_b/N_0 = 20 dB and $G = 32$, unless otherwise stated.

Concentrating on the BER curves of the SOTTD system, slight disparities between the simulation results and performance bounds can be identified for target BER values of 10^{-6} or worse. As can be seen from the graphs, the simulation curves are very close to the simulation bounds, for normalized user loads (K/G) of less than 0.75. For the conditions of low load ($P_b < 10^{-6}$), the performance of the simulated system is dominated by the performance of the suboptimal (non-ML) decoder and the practical choice of a random interleaver.

For the higher load conditions, the simulation results are also worse than the bounding performance, since the performance is limited in frequency-selective channels due to increased interference.

4. SUMMARY AND CONCLUSION

In this paper, a new concept of layered super-orthogonal turbo transmit diversity (SOTTD) has been presented for application in code-division multiple-access (CDMA) communication systems. The techniques of low-rate spreading and coding have been combined with orthogonal code-division transmit diversity (CDTD) and iterative "turbo" processing at the receiver. In contrast to layered ST turbo-coded (TC) CDTD, where a turbo encoder (and its associated iterative decoder) is required for every transmit diversity branch available, SOTTD requires a single turbo encoder-decoder pair, making it particularly attractive for CDMA wireless applications, the only requirement being that the number of constituent encoders Z be greater or equal to the transmit diversity order M_T.

From the performance results presented, it may be deduced that the proposed SOTTD system provides a very powerful and practical extension to the TC CDTD schemes, and yields superior performance compared to TC CDTD over the practically complete capacity range of CDMA. Another significant observation is the fact that the performance degradation of TC CDTD at low system loads (due to inherent TC error floor) is alleviated by the SOTTD system. This is explained in terms of the higher minimum free distance offered by the low rate-(1/16) constituent encoders, as opposed to the use of rate-(1/2) (256-state) constituent encoders in TC systems.

In conclusion, the interpretation of the performance bounds presented in this paper should be done within the confidence limits imposed by the use of the union bound, as well as the restrictions set by practical considerations, as such bounds are only valid for the case of ML decoding, and they may diverge significantly from the true performance at low values of E_b/N_0. Also, in the simulation, a suboptimal non-ML decoding algorithm was employed, as well as a pseudorandom interleaver. Furthermore, the performance of practical systems is strongly influenced by the availability of reliable CSI, which also plays a major role in the correct operation of virtually all adaptive receiver subsystems, including channel estimation, multipath decomposition and RAKE MRC, Doppler tracking, equalization, and several others. Clearly, the absence of reliable CSI will produce a noticeable degradation in the system performance. However, despite the restrictions and limitations, the results presented are close to the theoretical bounds for most of the normal CDMA operational range and thus provide useful design and comparative performance guidelines for SOTTD CDMA application scenarios.

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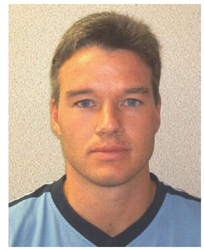

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EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

Special Issue on Image Perception

Call for Papers

Perception is a complex process that involves brain activities at different levels. The availability of models for the representation and interpretation of the sensory information opens up new research avenues that cut across neuroscience, imaging, information engineering, and modern robotics.

The goal of the multidisciplinary field of perceptual signal processing is to identify the features of the stimuli that determine their "perception," namely "a single unified awareness derived from sensory processes while a stimulus is present," and to derive associated computational models that can be generalized.

In the case of vision, the stimuli go through a complex analysis chain along the so-called "visual pathway," starting with the encoding by the photoreceptors in the retina (low-level processing) and ending with cognitive mechanisms (high-level processes) that depend on the task being performed.

Accordingly, low-level models are concerned with image "representation" and aim at emulating the way the visual stimulus is encoded by the early stages of the visual system as well as capturing the varying sensitivity to the features of the input stimuli; high-level models are related to image "interpretation" and allow to predict the performance of a human observer in a given predefined task.

A global model, accounting for both such bottom-up and top-down approaches, would enable the automatic interpretation of the visual stimuli based on both their low-level features and their semantic content.

Among the main image processing fields that would take advantage of such models are feature extraction, content-based image description and retrieval, model-based coding, and the emergent domain of medical image perception.

The goal of this special issue is to provide original contributions in the field of image perception and modeling.

Topics of interest include (but are not limited to):

- Perceptually plausible mathematical bases for the representation of visual information (static and dynamic)
- Modeling nonlinear processes (masking, facilitation) and their exploitation in the imaging field (compression, enhancement, and restoration)
- Beyond early vision: investigating the pertinence and potential of cognitive models (feature extraction, image quality)
- Stochastic properties of complex natural scenes (static, dynamic, colored) and their relationships with perception
- Perception-based models for natural (static and dynamic) textures. Theoretical formulation and psychophysical validation
- Applications in the field of biomedical imaging (medical image perception)

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Special Issue on Music Information Retrieval Based on Signal Processing

Call for Papers

The main focus of this special issue is on the application of digital signal processing techniques for music information retrieval (MIR). MIR is an emerging and exciting area of research that seeks to solve a wide variety of problems dealing with preserving, analyzing, indexing, searching, and accessing large collections of digitized music. There are also strong interests in this field of research from music libraries and the recording industry as they move towards digital music distribution. The demands from the general public for easy access to these music libraries challenge researchers to create tools and algorithms that are robust, small, and fast.

Music is represented in either encoded audio waveforms (CD audio, MP3, etc.) or symbolic forms (musical score, MIDI, etc.). Audio representations, in particular, require robust signal processing techniques for many applications of MIR since meaningful descriptions need to be extracted from audio signals in which sounds from multiple instruments and vocals are often mixed together. Researchers in MIR are therefore developing a wide range of new methods based on statistical pattern recognition, classification, and machine learning techniques such as the Hidden Markov Model (HMM), maximum likelihood estimation, and Bayes estimation as well as digital signal processing techniques such as Fourier and Wavelet transforms, adaptive filtering, and source-filter models. New music interface and query systems leveraging such methods are also important for end users to benefit from MIR research.

Although research contributions on MIR have been published at various conferences in 1990s, the members of the MIR research community meet annually at the International Conference on Music Information Retrieval (ISMIR) since 2000.

Topics of interest include (but are not limited to):

- Automatic summarization (succinct representation of music)
- Automatic transcription (audio to symbolic format conversion)
- Music annotation (semantic analysis)
- Music fingerprinting (unique identification of music)
- Music interface
- Music similarity metrics (comparison)
- Music understanding
- Musical feature extraction
- Musical styles and genres
- Optical music score recognition (image to symbolic format conversion)
- Performer/artist identification
- Query systems
- Timbre/instrument recognition

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Special Issue on Visual Sensor Networks

Call for Papers

Research into the design, development, and deployment of networked sensing devices for high-level inference and surveillance of the physical environment has grown tremendously in the last few years.

This trend has been motivated, in part, by recent technological advances in electronics, communication networking, and signal processing.

Sensor networks are commonly comprised of lightweight distributed sensor nodes such as low-cost video cameras. There is inherent redundancy in the number of nodes deployed and corresponding networking topology. Operation of the network requires autonomous peer-based collaboration amongst the nodes and intermediate data-centric processing amongst local sensors. The intermediate processing known as in-network processing is application-specific. Often, the sensors are untethered so that they must communicate wirelessly and be battery-powered. Initial focus was placed on the design of sensor networks in which scalar phenomena such as temperature, pressure, or humidity were measured.

It is envisioned that much societal use of sensor networks will also be based on employing content-rich vision-based sensors. The volume of data collected as well as the sophistication of the necessary in-network stream content processing provide a diverse set of challenges in comparison with generic scalar sensor network research.

Applications that will be facilitated through the development of visual sensor networking technology include automatic tracking, monitoring and signaling of intruders within a physical area, assisted living for the elderly or physically disabled, environmental monitoring, and command and control of unmanned vehicles.

Many current video-based surveillance systems have centralized architectures that collect all visual data at a central location for storage or real-time interpretation by a human operator. The use of distributed processing for automated event detection would significantly alleviate mundane or time-critical activities performed by human operators, and provide better network scalability. Thus, it is expected that video surveillance solutions of the future will successfully utilize visual sensor networking technologies.

Given that the field of visual sensor networking is still in its infancy, it is critical that researchers from the diverse disciplines including signal processing, communications, and electronics address the many challenges of this emerging field. This special issue aims to bring together a diverse set of research results that are essential for the development of robust and practical visual sensor networks.

Topics of interest include (but are not limited to):

- Sensor network architectures for high-bandwidth vision applications
- Communication networking protocols specific to visual sensor networks
- Scalability, reliability, and modeling issues of visual sensor networks
- Distributed computer vision and aggregation algorithms for low-power surveillance applications
- Fusion of information from visual and other modalities of sensors
- Storage and retrieval of sensor information
- Security issues for visual sensor networks
- Visual sensor network testbed research
- Novel applications of visual sensor networks
- Design of visual sensors

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Special Issue on Multirate Systems and Applications

Call for Papers

Filter banks for the application of subband coding of speech were introduced in the 1970s. Since then, filter banks and multirate systems have been studied extensively. There has been great success in applying multirate systems to many applications. The most notable of these applications include subband coding for audio, image, and video, signal analysis and representation using wavelets, subband denoising, and so forth. Different applications also call for different filter bank designs and the topic of designing one-dimensional and multidimentional filter banks for specific applications has been of great interest.

Recently there has been growing interest in applying multirate theories to the area of communication systems such as, transmultiplexers, filter bank transceivers, blind deconvolution, and precoded systems. There are strikingly many dualities and similarities between multirate systems and multicarrier communication systems. Many problems in multicarrier transmission can be solved by extending results from multirate systems and filter banks. This exciting research area is one that is of increasing importance.

The aim of this special issue is to bring forward recent developments on filter banks and the ever-expanding area of applications of multirate systems.

Topics of interest include (but are not limited to):

- Multirate signal processing for communications
- Filter bank transceivers
- One-dimensional and multidimensional filter bank designs for specific applications
- Denoising
- Adaptive filtering
- Subband coding
- Audio, image, and video compression
- Signal analysis and representation
- Feature extraction and classification
- Other applications

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EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

Special Issue on
Multisensor Processing for Signal Extraction and Applications

Call for Papers

Source signal extraction from heterogeneous measurements has a wide range of applications in many scientific and technological fields, for example, telecommunications, speech and acoustic signal processing, and biomedical pattern analysis. Multiple signal reception through multisensor systems has become an effective means for signal extraction due to its superior performance over the monosensor mode. Despite the rapid progress made in multisensor-based techniques in the past few decades, they continue to evolve as key technologies in modern wireless communications and biomedical signal processing. This has led to an increased focus by the signal processing community on the advanced multisensor-based techniques which can offer robust high-quality signal extraction under realistic assumptions and with minimal computational complexity. However, many challenging tasks remain unresolved and merit further rigorous studies. Major efforts in developing advanced multisensor-based techniques may include high-quality signal extraction, realistic theoretical modeling of real-world problems, algorithm complexity reduction, and efficient real-time implementation.

The purpose of this special issue aims to present state-of-the-art multisensor signal extraction techniques and applications. Contributions in theoretical study, performance analysis, complexity reduction, computational advances, and real-world applications are strongly encouraged.

Topics of interest include (but are not limited to):

- Multiantenna processing for radio signal extraction
- Multimicrophone speech recognition and enhancement
- Multisensor radar, sonar, navigation, and biomedical signal processing
- Blind techniques for multisensor signal extraction
- Computational advances in multisensor processing

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Special Issue on

Search and Retrieval of 3D Content and Associated Knowledge Extraction and Propagation

Call for Papers

With the general availability of 3D digitizers, scanners, and the technology innovation in 3D graphics and computational equipment, large collections of 3D graphical models can be readily built up for different applications (e.g., in CAD/CAM, games design, computer animations, manufacturing and molecular biology). For such large databases, the method whereby 3D models are sought merits careful consideration. The simple and efficient query-by-content approach has, up to now, been almost universally adopted in the literature. Any such method, however, must first deal with the proper positioning of the 3D models. The two prevalent-in-the-literature methods for the solution to this problem seek either

- Pose Normalization: Models are first placed into a canonical coordinate frame (normalizing for translation, scaling, and rotation). Then, the best measure of similarity is found by comparing the extracted feature vectors, or
- Descriptor Invariance: Models are described in a transformation invariant manner, so that any transformation of a model will be described in the same way, and the best measure of similarity is obtained at any transformation.

The existing 3D retrieval systems allow the user to perform queries by example. The queried 3D model is then processed, low-level geometrical features are extracted, and similar objects are retrieved from a local database. A shortcoming of the methods that have been proposed so far regarding the 3D object retrieval, is that neither is the semantic information (high-level features) attached to the (low-level) geometric features of the 3D content, nor are the personalization options taken into account, which would significantly improve the retrieved results. Moreover, few systems exist so far to take into account *annotation* and *relevance feedback* techniques, which are very popular among the corresponding content-based image retrieval systems (CBIR).

Most existing CBIR systems using knowledge either annotate all the objects in the database (full annotation) or annotate a subset of the database manually selected (partial annotation). As the database becomes larger, full annotation is increasingly difficult because of the manual effort needed. Partial annotation is relatively affordable and trims down the heavy manual labor. Once the database is partially annotated, traditional image analysis methods are used to derive semantics of the objects not yet annotated. However, it is not clear "how much" annotation is sufficient for a specific database and what the best subset of objects to annotate is. In other words how the knowledge *will be propagated*. Such techniques have not been presented so far regarding the 3D case.

Relevance feedback was first proposed as an interactive tool in text-based retrieval. Since then it has been proven to be a powerful tool and has become a major focus of research in the area of content-based search and retrieval. In the traditional computer centric approaches, which have been proposed so far, the "best" representations and weights are fixed and they cannot effectively model high-level concepts and user's perception subjectivity. In order to overcome these limitations of the computer centric approach, techniques based on *relevant feedback*, in which the human and computer interact to refine high-level queries to representations based on low-level features, should be developed.

The aim of this special issue is to focus on recent developments in this expanding research area. The special issue will focus on novel approaches in 3D object retrieval, transforms and methods for efficient geometric feature extraction, annotation and relevance feedback techniques, knowledge propagation (e.g., using Bayesian networks), and their combinations so as to produce a single, powerful, and dominant solution.

Topics of interest include (but are not limited to):

- 3D content-based search and retrieval methods (volume/surface-based)
- Partial matching of 3D objects
- Rotation invariant feature extraction methods for 3D objects

- Graph-based and topology-based methods
- 3D data and knowledge representation
- Semantic and knowledge propagation over heterogeneous metadata types
- Annotation and relevance feedback techniques for 3D objects

Authors should follow the EURASIP JASP manuscript format described at http://www.hindawi.com/journals/asp/. Prospective authors should submit an electronic copy of their complete manuscript through the EURASIP JASP manuscript tracking system at http://www.mstracking.com/asp/, according to the following timetable:

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Special Issue on Robust Speech Recognition

Call for Papers

Robustness can be defined as the ability of a system to maintain performance or degrade gracefully when exposed to conditions not well represented in the data used to develop the system. In automatic speech recognition (ASR), systems must be robust to many forms of signal degradation, including speaker characteristics (e.g., dialect and accent), ambient environment (e.g., cellular telephony), transmission channel (e.g., voice over IP), and language (e.g., new words, dialect switching). Robust ASR systems, which have been under development for the past 35 years, have made great progress over the years closing the gap between performance on pristine research tasks and noisy operational data.

However, in recent years, demand is emerging for a new class of systems that tolerate extreme and unpredictable variations in operating conditions. For example, in a cellular telephony environment, there are many nonstationary forms of noise (e.g., multiple speakers) and significant variations in microphone type, position, and placement. Harsh ambient conditions typical in automotive and mobile applications pose similar challenges. Development of systems in a language or dialect for which there is limited or no training data in a target language has become a critical issue for a new generation of voice mining applications. The existence of multiple conditions in a single stream, a situation common to broadcast news applications, and that often involves unpredictable changes in speaker, topic, dialect, or language, is another form of robustness that has gained attention in recent years.

Statistical methods have dominated the field since the early 1980s. Such systems tend to excel at learning the characteristics of large databases that represent good models of the operational conditions and do not generalize well to new environments.

This special issue will focus on recent developments in this key research area. Topics of interest include (but are not limited to):

- Channel and microphone normalization
- Stationary and nonstationary noise modeling, compensation, and/or rejection
- Localization and separation of sound sources (including speaker segregation)
- Signal processing and feature extraction for applications involving hands-free microphones
- Noise robust speech modeling
- Adaptive training techniques
- Rapid adaptation and learning
- Integration of confidence scoring, metadata, and other alternative information sources
- Audio-visual fusion
- Assessment relative to human performance
- Machine learning algorithms for robustness
- Transmission robustness
- Pronunciation modeling

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EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

Special Issue on

Signal Processing Technologies for Ambient Intelligence in Home-Care Applications

Call for Papers

The possibility of allowing elderly people with different kinds of disabilities to conduct a normal life at home and achieve a more effective inclusion in the society is attracting more and more interest from both industrial and governmental bodies (hospitals, healthcare institutions, and social institutions). Ambient intelligence technologies, supported by adequate networks of sensors and actuators, as well as by suitable processing and communication technologies, could enable such an ambitious objective.

Recent researches demonstrated the possibility of providing constant monitoring of environmental and biomedical parameters, and the possibility to autonomously originate alarms, provide primary healthcare services, activate emergency calls, and rescue operations through distributed assistance infrastructures. Nevertheless, several technological challenges are still connected with these applications, ranging from the development of enabling technologies (hardware and software), to the standardization of interfaces, the development of intuitive and ergonomic human-machine interfaces, and the integration of complex systems in a highly multidisciplinary environment.

The objective of this special issue is to collect the most significant contributions and visions coming from both academic and applied research bodies working in this stimulating research field. This is a highly interdisciplinary field comprising many areas, such as signal processing, image processing, computer vision, sensor fusion, machine learning, pattern recognition, biomedical signal processing, multimedia, human-computer interfaces, and networking.

The focus will be primarily on the presentation of original and unpublished works dealing with ambient intelligence and domotic technologies that can enable the provision of advanced homecare services.

This special issue will focus on recent developments in this key research area. Topics of interest include (but are not limited to):

- Video-based monitoring of domestic environments and users
- Continuous versus event-driven monitoring
- Distributed information processing
- Data fusion techniques for event association and automatic alarm generation
- Modeling, detection, and learning of user habits for automatic detection of anomalous behaviors
- Integration of biomedical and behavioral data
- Posture and gait recognition and classification
- Interactive multimedia communications for remote assistance
- Content-based encoding of medical and behavioral data
- Networking support for remote healthcare
- Intelligent/natural man-machine interaction, personalization, and user acceptance

Authors should follow the EURASIP JASP manuscript format described at http://www.hindawi.com/journals/asp/. Prospective authors should submit an electronic copy of their complete manuscripts through the EURASIP JASP manuscript tracking system at http://www.mstracking.com/asp/, according to the following timetable:

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Institute of Acoustics

NEWS RELEASE

Nominations Invited for the Institute of Acoustics 2006 A B Wood Medal

The Institute of Acoustics, the UK's leading professional body for those working in acoustics, noise and vibration, is inviting nominations for its prestigious A B Wood Medal for the year 2006.

The A B Wood Medal and prize is presented to an individual, usually under the age of 35, for distinguished contributions to the application of underwater acoustics. The award is made annually, in even numbered years to a person from Europe and in odd numbered years to someone from the USA/Canada. The 2005 Medal was awarded to Dr A Thode from the USA for his innovative, interdisciplinary research in ocean and marine mammal acoustics.

Nominations should consist of the candidate's CV, clearly identifying peer reviewed publications, and a letter of endorsement from the nominator identifying the contribution the candidate has made to underwater acoustics. In addition, there should be a further reference from a person involved in underwater acoustics and not closely associated with the candidate. Nominees should be citizens of a European Union country for the 2006 Medal. Nominations should be marked confidential and addressed to the President of the Institute of Acoustics at 77A St Peter's Street, St. Albans, Herts, AL1 3BN. The deadline for receipt of nominations is **15 October 2005**.

Dr Tony Jones, President of the Institute of Acoustics, comments, "A B Wood was a modest man who took delight in helping his younger colleagues. It is therefore appropriate that this prestigious award should be designed to recognise the contributions of young acousticians."

Further information and an nomination form
can be found on the Institute's website at
www.ioa.org.uk.

A B Wood

Albert Beaumont Wood was born in Yorkshire in 1890 and graduated from Manchester University in 1912. He became one of the first two research scientists at the Admiralty to work on antisubmarine defence. He designed the first directional hydrophone and was well known for the many contributions he made to the science of underwater acoustics and for the help he gave to younger colleagues. The medal was instituted after his death by his many friends on both sides of the Atlantic and was administered by the Institute of Physics until the formation of the Institute of Acoustics in 1974.

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EDITORS NOTES

The Institute of Acoustics is the UK's professional body for those working in acoustics, noise and vibration. It was formed in 1974 from the amalgamation of the Acoustics Group of the Institute of Physics and the British Acoustical Society (a daughter society of the Institution of Mechanical Engineers). The Institute of Acoustics is a nominated body of the Engineering Council, offering registration at Chartered and Incorporated Engineer levels.

The Institute has some 2500 members from a rich diversity of backgrounds, with engineers, scientists, educators, lawyers, occupational hygienists, architects and environmental health officers among their number. This multidisciplinary culture provides a productive environment for cross-fertilisation of ideas and initiatives. The range of interests of members within the world of acoustics is equally wide, embracing such aspects as aerodynamics, architectural acoustics, building acoustics, electroacoustics, engineering dynamics, noise and vibration, hearing, speech, underwater acoustics, together with a variety of environmental aspects. The lively nature of the Institute is demonstrated by the breadth of its learned society programmes.

For more information please visit our site at
www.ioa.org.uk.

EURASIP Book Series on Signal Processing and Communications

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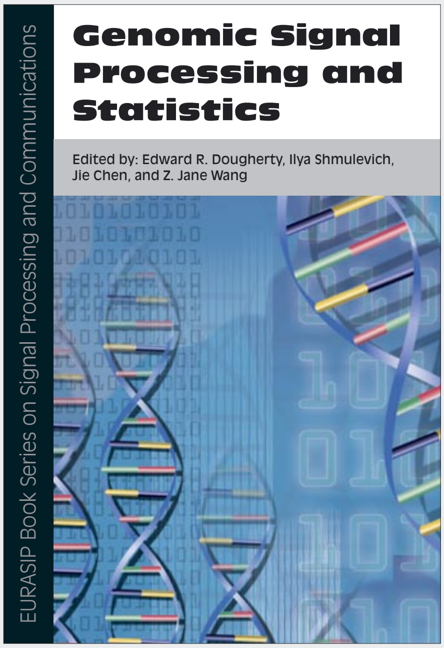
GENOMIC SIGNAL PROCESSING AND STATISTICS

Edited by: Edward R. Dougherty, Ilya Shmulevich, Jie Chen, and Z. Jane Wang



Recent advances in genomic studies have stimulated synergetic research and development in many cross-disciplinary areas. Genomic data, especially the recent large-scale microarray gene expression data, represents enormous challenges for signal processing and statistics in processing these vast data to reveal the complex biological functionality. This perspective naturally leads to a new field, genomic signal processing (GSP), which studies the processing of genomic signals by integrating the theory of signal processing and statistics. Written by an international, interdisciplinary team of authors, this invaluable edited volume is accessible to students just entering this emergent field, and to researchers, both in academia and industry, in the fields of molecular biology, engineering, statistics, and signal processing. The book provides tutorial-level overviews and addresses the specific needs of genomic signal processing students and researchers as a reference book.

The book aims to address current genomic challenges by exploiting potential synergies between genomics, signal processing, and statistics, with special emphasis on signal processing and statistical tools for structural and functional understanding of genomic data. The book is partitioned into three parts. In part I, a brief history of genomic research and a background introduction from both biological and signal-processing/statistical perspectives are provided so that readers can easily follow the material presented in the rest of the book. In part II, overviews of state-of-the-art techniques are provided. We start with a chapter on sequence analysis, and follow with chapters on feature selection, clustering, and classification of microarray data. The next three chapters discuss the modeling, analysis, and simulation of biological regulatory networks, especially gene regulatory networks based on Boolean and Bayesian approaches. The next two chapters treat visualization and compression of gene data, and supercomputer implementation of genomic signal processing systems. Part II concludes with two chapters on systems biology and medical implications of genomic research. Finally, part III discusses the future trends in genomic signal processing and statistics research.

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