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Iterative List Decoding of Concatenated Source-Channel Codes

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Whenever variable-length entropy codes are used in the presence of a noisy channel, any channel errors will propagate and cause significant harm. Despite using channel codes, some residual errors always remain, whose effect will get magnified by error propagation. Mitigating this undesirable effect is of great practical interest. One approach is to use the residual redundancy of variable-length codes for joint source-channel decoding. In this paper, we improve the performance of residual redundancy source-channel decoding via an iterative list decoder made possible by a nonbinary outer CRC code. We show that the list decoding of VLCs is beneficial for entropy codes that contain redundancy. Such codes are used in state-of-the-art video coders, for example. The proposed list decoder improves the overall performance significantly in AWGN and fully interleaved Rayleigh fading channels.

Keywords and phrases: joint source-channel coding, variable-length codes, list decoding, iterative decoding.

1. INTRODUCTION

Variable-length codes (VLCs) for entropy coding are by now a central part of most data compression techniques, which are in turn essential for many communications applications, including text, voice, images, and video. While VLCs achieve significant compression, they also introduce dependencies in the data structure through their variable length, thus leading to error propagation in the decoded sequence.

One of the techniques that has been used to combat this undesirable effect is joint source-channel decoding. It is known that even the most efficient symbol-by-symbol compression (Huffman code) does not always achieve the entropy limit, therefore redundancy often remains in compressed data. This redundancy can, in principle, be used to assist the decoder.

Taking this argument one step further, it has been proposed to leave redundancy intentionally in entropy codes, for the purposes of resilience against channel noise. For example, the video coding standard H.263+ and its descendants use a reversible variable-length code (RVLC) [1] whose compression efficiency is less than Huffman codes. However, the RVLC allows bidirectional symbol-based decoding which is useful in the presence of channel errors. This approach has been generalized by designing entropy codes with prespecified minimum distance [2, 3].

The error resilience of entropy codes can be used to "clean up" any residual errors from the traditional error control coding (see Figure 1). For example, in the case of RVLC, one may start decoding from the end of the sequence whenever an error is observed. This is a separable approach to decoding. However, we know today that serially concatenated codes offer significantly improved performance if the decoding operation is done *jointly*, via the soft-input soft-output (SISO) decoding algorithm. This principle has been applied to finite-alphabet source-channel codes by Bauer and Hagenauer [4, 5], and further analyzed in [6, 7].

In this paper, we propose an improvement over the method of Bauer and Hagenauer by introducing a list decoder for source-channel decoding, made possible by a nonbinary CRC outer code. We implement this list decoder via an iterative decoding procedure similar to serial concatenated codes (Figure 2).

We briefly summarize and review the issues of iterative source-channel decoding in Section 2. We introduce list decoding of the concatenated code in Section 3. We present some analytical and experimental results in Section 4 and offer concluding remarks in Section 5.

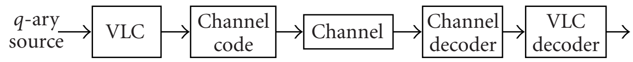

FIGURE 1: Conventional concatenated source-channel decoder.

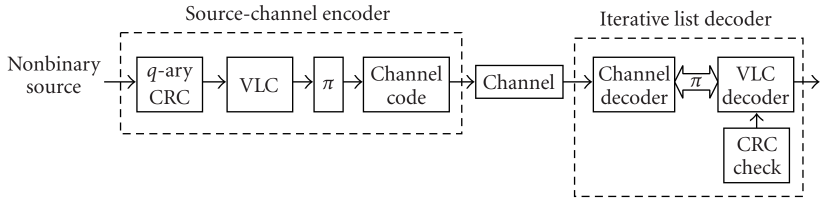

FIGURE 2: Proposed list iterative joint source-channel coding system.

2. SERIAL CONCATENATION OF VLC AND CHANNEL CODES

For the clarity of exposition, we first consider the system of Figure 2 in the absence of the CRC and list decoding component. The simplified system consists of an outer (VLC) code and an inner channel code, separated by an interleaver π. The source and channel codes are jointly (iteratively) decoded at the receiver. As mentioned previously, this method relies on residual redundancy in the VLC, in particular, sometimes redundancy is retained in the VLC on purpose, for example, in RVLCs. Thus, for the purposes of this section, we treat both codes in terms of their distance properties.

We treat the outer code, $\mathcal{C}^o$, as a channel code. The key difficulty of the analysis, which requires a generalization of the well-known work of [8], is that VLCs are nonlinear.

The proceeding analysis closely follows that of [7]. Assume a sequence of K symbols is encoded, and the average length of the outer entropy code symbols is ℓ_{ave}. Hence, the output bit sequence of $\mathcal{C}^o$ has a variable length with size $N_{\min}$ to $N_{\max}$. Code $\mathcal{C}^o$ is partitioned in a way such that all codewords of $\mathcal{C}^o$ with length $N \in [N_{\min}, N_{\max}]$ form a subcode denoted by $\mathcal{C}_N$. In other words, to avoid dealing with variable lengths, we partition the set of all composite codewords into sets of equal length [2]. We define the free distance of $\mathcal{C}^o$, d_f^o as the minimum of the free distances of $\mathcal{C}_N$'s.

The number of inner codewords with output weight h and input weight ℓ is shown by $A^i_{\ell,h}$. Assume the outer subcode $\mathcal{C}_N$ has $A^o_\ell(N)$ pairs of codewords with Hamming distance ℓ. Using the uniform interleaver notion of [8], and thanks to linearity of the inner code, the number of pairs of codewords of the overall concatenated code, with Hamming distance h, is

$$A_h(N) = \sum_{\ell=d_f^o}^{N} \frac{A^o_\ell(N) A^i_{\ell,h}(N)}{\binom{N}{\ell}}. \tag{1}$$

The pairwise error probability (PEP) of a pair of codewords with Hamming distance h is $P_h = Q(\sqrt{2hE_s/N_0})$. Using (1), the PEP of the concatenated code is

$$\begin{aligned} P_E &\leq \sum_{N=N_{\min}}^{N_{\max}} \Pr(N) \sum_{h=d_f}^{N/R^i} A_h(N) P_h \\ &= \sum_{N=N_{\min}}^{N_{\max}} \sum_{h=d_f}^{N/R^i} \sum_{\ell \geq d_f^o} \Pr(N) \frac{A^o_\ell(N) A^i_{\ell,h}(N)}{\binom{N}{\ell}} Q\left(\sqrt{2h\frac{E_s}{N_0}}\right), \end{aligned} \tag{2}$$

where d_f is the free distance of the concatenated code, R^i is the rate of the inner channel code, and $\Pr(N)$ is the probability of the codewords of $\mathcal{C}_N$. We note that the above union bound can be used with different choices of inner and outer codes, for example, a convolutional or turbo code as inner code [4, 9, 10], or Huffman code or RVLC as outer code. A similar development is possible for symbol error rate [7], which we do not present here for the sake of brevity.

Iterative decoding of the concatenated source-channel code is performed via soft-input soft-output (SISO) modules for the inner and outer codes. For the outer code, the SISO module is performed over a bit-level trellis representation of VLC, similar to the one originally proposed by Balakirsky [11].

3. LIST DECODING OF SERIALLY CONCATENATED VLC AND CHANNEL CODES

A list decoder provides an ordered list of the L most probable sequences in maximum-likelihood sense. Then, an outer error detecting code, usually a cyclic redundancy check (CRC) code, verifies the validity of the candidates and selects the error-free sequence, if exists, among the candidates. Two variations of the list Viterbi algorithm (LVA) are reported in [12].

An ordinary ML (Viterbi) decoder makes an error whenever the codeword closest to the received waveform is an erroneous codeword. For the list decoder to make an error, the correct sequence must lie outside of the L nearest neighbors

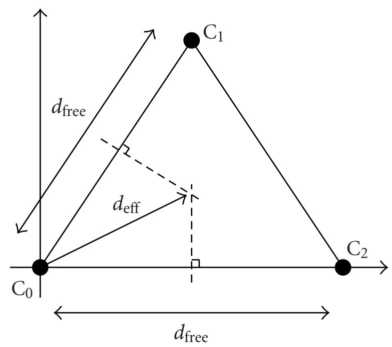

FIGURE 3: Asymptotic analysis of list Viterbi algorithm.

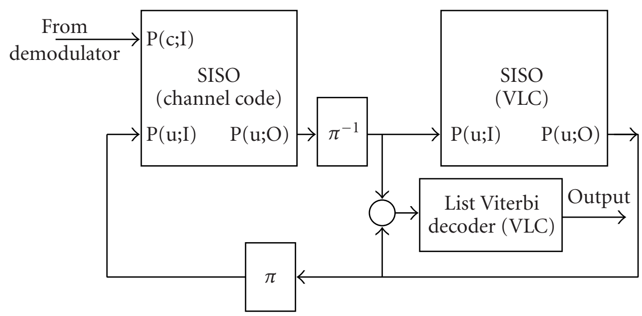

FIGURE 4: Iterative list decoding of VLC and channel code.

of the received sequence. This error is less probable than the corresponding error in the ML decoder.

In a list decoder, the distance between the received sequence and all the candidates determines the performance. Therefore, determining the exact performance is mathematically intractable. But it is possible to calculate the asymptotic coding gain, for example, see [12]. In the case of AWGN channel, a geometrical argument reveals that the asymptotic coding gain is $G = 10\log(2L/(L+1))$ dB for a list of length L. However, the actual gain is often less due to the multiplicity of the set of L nearest neighbors, which is neglected in the analysis [12].

3.1. List decoding of variable-length codes

List decoders can also be applied for variable-length encoded sequences, given an appropriate trellis (e.g., the bit-level trellises mentioned earlier). Our list decoding is constructed with the help of a non binary CRC code, which verifies the validity of the L most probable paths in the VLC trellis. The alphabet set of the CRC code must cover all codewords of the VLC (size q). If q is a power of a prime, it is possible to construct a q-ary CRC code, otherwise the size of VLC should be extended to the nearest power of a prime. One can use the a priori knowledge that these additional symbols are never present in the data sequence, but only (possibly) present in the parity sequence.

The asymptotic error rate for a list of size $L = 2$ is based on a simple geometric construction due to Seshadri and Sundberg [12] (see Figure 3). When the three codewords are pairwise equidistant, it produces a worst-case error probability. In this case, the minimum-magnitude noise resulting in an error is shown by the vector terminating at the circumcenter of the triangle. This vector represents the effective minimum distance, denoted by d_{eff}, which is larger than $d_{\text{free}}/2$, explaining the list decoding gain, which is equal to $10\log(2L/(L+1))$ dB, as calculated in [12].

This value of asymptotic gain, however, ignores the multiplicities of the minimum distance, and in our case minimum-distance error event has high multiplicities.[1] Therefore, we augment the asymptotic analysis of [12, 13] for $L = 2, 3$ list decoder of VLCs so that multiplicities are taken into account. We denote by N_{free} the multiplicity of the minimum-distance errors.[2] The number of codeword triplets at minimum-distance that include the transmitted codeword is $N_{\text{eff}} = N_{\text{free}}(N_{\text{free}} - 1)/2$. Thus, for $L = 2$ and assuming an AWGN channel, coding gain is the difference $\Delta\gamma = \gamma_1 - \gamma_2$, where γ_1 and γ_2 are the two values of E_b/N_0 such that

$$N_{\text{eff}}Q\left(\sqrt{2d_{\text{eff}}\gamma_2}\right) = N_{\text{free}}Q\left(\sqrt{2d_{\text{free}}\gamma_1}\right). \quad (3)$$

Simulations show that the coding gain thus obtained is more accurate than results that ignore multiplicities, for example, [12, 13] (see Section 5). The disadvantage is that the equation above does not admit a closed-form solution.

Similarly, worst-case analysis can be repeated for $L = 3$ list decoder to calculate d_{eff}. To obtain a more realistic approximation of the coding gain, we consider the multiplicity of the worst case of the set of three codewords, which is $N_{\text{eff}} = N_{\text{free}}(N_{\text{free}}-1)(N_{\text{free}}-2)/6$, given $N_{\text{free}} \geq 3$. The coding gain is calculated in a similar way as $L = 2$.

3.2. Proposed iterative list decoder

We now introduce an approximated list decoder for the concatenation of VLCs and channel codes. Our proposed iterative list decoder is demonstrated in Figure 4. After the last iteration, the final soft-output sequence produced by the SISO is decoded by the list Viterbi algorithm. The trellis used in this final decoder is similar to the one used in SISO-VLC.

The asymptotic analysis of the list decoder of turbo codes in [13] shows that the coding gain of list turbo decoder is higher than the coding gain of convolutional list decoder. Specifically, due to the low probability of multiple free-distance error events in a turbo-encoded sequence, the asymptotic coding gain is determined by the second minimum distance, yielding higher gain [13]. For the case of serially concatenated VLCs and convolutional codes, we show experimentally in Section 4 that significant improvements in coding performance can be achieved.

[1]More information on the distance spectrum of VLCs is available in [2], and two examples are given in [4].

[2]The multiplicities of VLCs, in general, are not integer-valued since we must average the multiplicities of the subcodes. In our analysis, we round the multiplicities up to simplify the calculation.

Table 1: Variable-length codes used in Section 4.

s	$P_S(s)$	C_1	C_2 [4]	C_3
0	0.33	00	00	11
1	0.30	11	11	001
2	0.18	10	010	0100
3	0.10	010	101	0101100
4	0.09	011	0110	0001010
$E[L]$	$H = 2.14$	2.19	2.46	3.61
d_{free}	—	1	2	3

Table 2: Convolutional codes used in Section 4 (from [8]).

CC_1: rate $\frac{1}{2}$	$\left(1, \dfrac{1+D^2}{1+D+D^2}\right)$
CC_2: rate $\frac{1}{2}$	$\left(1, \dfrac{1+D+D^3}{1+D}\right)$
CC_3: rate $\frac{2}{3}$	$\begin{pmatrix} 1 & 0 & \dfrac{1+D^2}{1+D+D^2} \\ 0 & 1 & \dfrac{1+D}{1+D+D^2} \end{pmatrix}$

3.3. *Nonbinary CRC*

Wicker [14] provides a comprehensive background on Galois fields, rings of polynomials on Galois fields, and the construction of cyclic codes. We give here a quick summary of the key results as well as the procedure for designing nonbinary CRCs.

Cyclic codes are built using a generator polynomial on the underlying Galois field GF(q). If the number of symbols in our application is not a power of a prime, the next higher appropriate q must be chosen, since for a field GF(q), q must be either a prime or a power of a prime. Cyclic codes are built from a generator polynomial $g(X)$ on GF(q). The codewords are all the multiples of $g(X)$ modulo $X^n - 1$, where $g(X)$ is a degree-r polynomial that divides $X^n - 1$.

CRC codes are shortened cyclic codes that can encode up to $n - r$ information symbols. CRC codes have excellent error detection capability. The CRC code with a generator of degree r detects all burst errors of length r or less, and the probability that the CRC will not detect a random error is q^{-r}. Due to the lack of a convenient way to calculate the error spectrum of a CRC code, ad hoc methods have been used for code design in the binary case.

Unfortunately the existing ad hoc techniques for binary CRC design are not particularly helpful for the q-ary case, but nevertheless, the general structural properties, error coverage, and burst error detection properties remain the same across different underlying Galois fields. Therefore, even though we cannot design CRC with specified minimum distance, still it is possible to arrive at codes that have very respectable error detection performance. For example, for the 5-ary code used in the next section, a possible choice for generator polynomial is the primitive polynomial $X^8 + 4X^6 + X^4 + X^3 + X^2 + 3X + 3$ which requires 8 parity symbols for data sequences up to 390617 symbols. The undetected codeword error probability for this code is only 2.56×10^{-6}.

4. EXPERIMENTAL RESULTS

Table 1 shows the 5-ary source used in our experiments and various codes designed for this source. C_1 is a Huffman code, C_2 is an RVLC for this source reported in [4], C_3 is a high-redundancy code designed by us because we observed that the free distance of the outer code is a crucial factor in performance, as seen by the asymptotic behavior of the multiplicities A_h in (1). It is noteworthy that despite the differences, the trellises of the different codes have roughly the same order of complexity, due to sparseness of the VLC trellises.

Table 2 shows the recursive convolutional codes employed as inner code in our schemes. In our experiments, a packet of K symbols is entropy-encoded, interleaved, channel encoded, and transmitted using binary phase-shift keying (BPSK) modulation over an AWGN channel or fully interleaved Rayleigh fading channel.

4.1. *Iterative decoding*

Figure 5a shows union bounds[3] and simulation results for the concatenated code C_2+CC_1. The calculation of the multiplicities for a nonlinear, variable-length code is a lengthy and time-consuming process, thus we present *truncated bounds* calculated with the first 10 terms of the multiplicities of the outer code that are available in [4]. The decoding experiment was performed with 10 iterations, with packet lengths of 20 and 200.

We consider two outer VLCs: code C_2 with free distance 2 and code C_3 with free distance 3, to build codes C_2+CC_1 and C_3+CC_3 with overall rates 0.445 and 0.404, respectively.[4] The symbol error rate (SER) of the two concatenated codes is shown in Figure 5b for $K = 2000$ symbols. In a wide range of E_b/N_0, the code $C_3 + CC_3$ outperforms $C_2 + CC_1$ and demonstrates a sharper drop in error rate. Other simulations have shown that in terms of frame error rate (FER), $C_3 + CC_3$ provides significant coding gain, about 1.4 dB at FER $= 10^{-3}$.

For $C_2 + CC_1$, we noticed that the higher number of iterations does not provide much of coding gain. We use the density evolution technique to give insight into the progress of iterative decoder. After an experimental verification that the LLR histograms are indeed Gaussian, we evaluated the approximate density evolution for $C_2 + CC_1$ and $C_3 + CC_3$ (Figure 6). The two lower curves in each plot correspond to

[3]Union bounds work in the high E_b/N_0 regions, and they are calculated for the optimal (ML) decoder, and iterative decoding is not optimal. This explains the deviations of simulation from union bounds.

[4]The equivalent code rate of a VLC is defined as the average length of the VLC divided by the average length of the Huffman code.

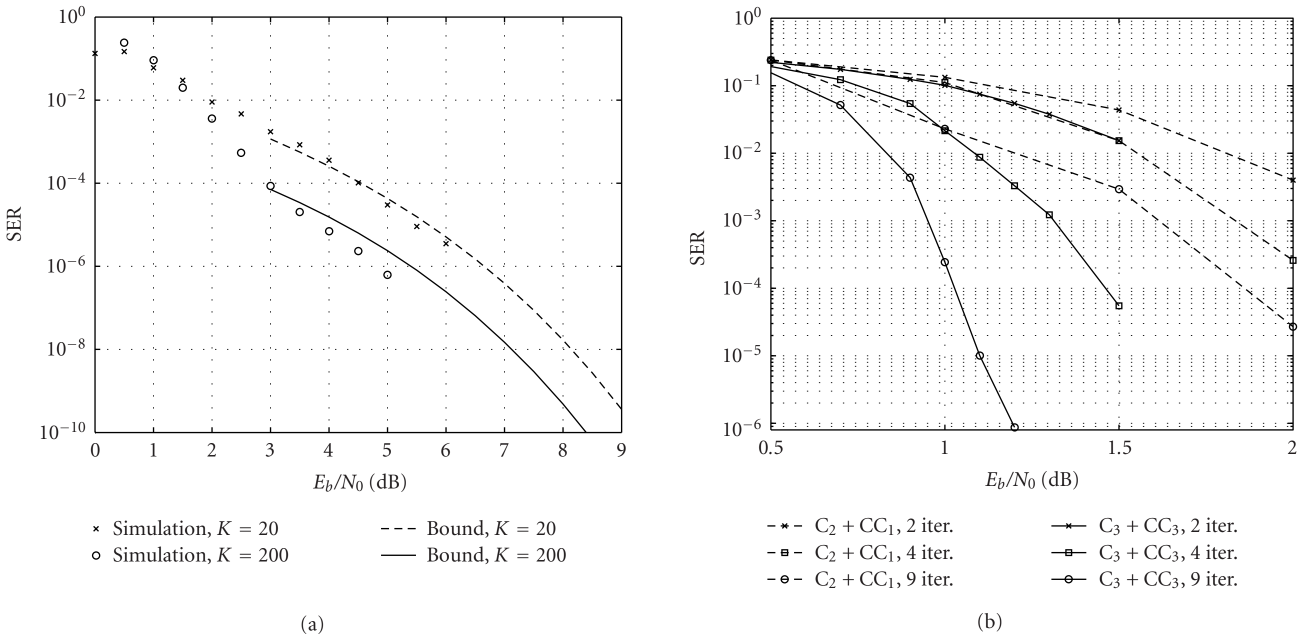

FIGURE 5: (a) Performance and union bounds of $C_2 + CC_1$, $K = 20$ and 200 symbols; (b) performance of $C_2 + CC_1$, and $C_3 + CC_3$, $K = 2000$.

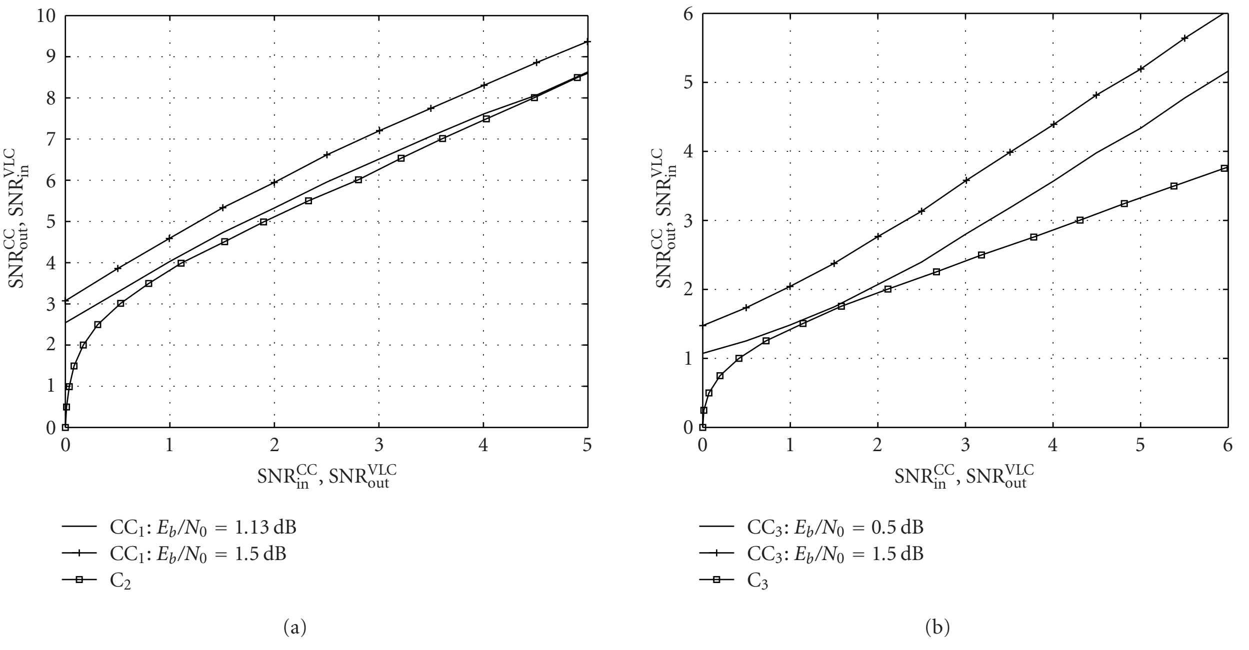

FIGURE 6: Approximate Gaussian density evolution of $C_2 + CC_1$ and $C_3 + CC_3$, $K = 2000$.

the iterative decoding threshold [15]. The code $C_3 + CC_3$ has lower threshold than $C_2 + CC_1$ (0.5 dB compared to 1.15 dB).

Borrowing the notion of iterative decoder tunnel from [15], we observe that the wider tunnel of $C_3 + CC_3$ provides a fast convergence with a few iterations: the higher the channel E_b/N_0, the fewer the iterations needed for convergence. These observations are in agreement with Figure 5b.

4.2. Iterative list decoding

We first evaluated the accuracy of our analysis for the performance of list decoding, which takes multiplicities into account. We used code C_2, with $K = 200$ symbols in the AWGN channel. The coding gain at FER $= 10^{-4}$ is calculated as 1 dB for $L = 2$ and 1.4 dB for $L = 3$. These values are a better

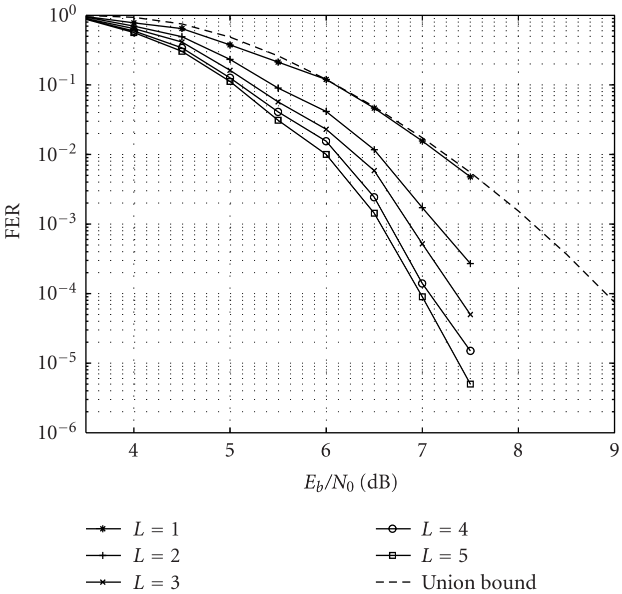

FIGURE 7: List decoding of C_2 in AWGN channel, K=200.

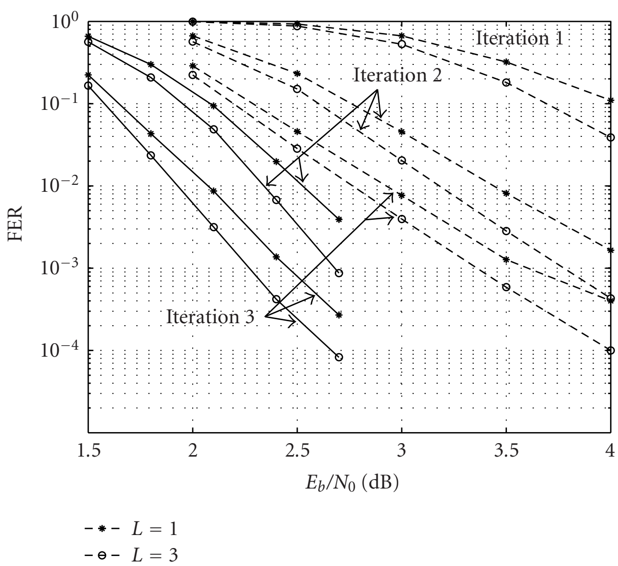

FIGURE 8: Iterative list decoding of $C_2 + CC_1$ (dashed) and $C_3 + CC_3$ (solid line) in AWGN channel, K=500.

match to simulations (Figure 7) than the coding gain predicted by [12].

Consider the two codes $C_2 + CC_1$ and $C_3 + CC_3$. Figure 8 presents the FER of the iterative list decoder at the first, second, and third iterations with $L = 1, 3$ in AWGN channel with $K = 500$. $C_3 + CC_3$ outperforms $C_2 + CC_1$. Figure 9 reports the FER of the concatenated codes in a fully interleaved Rayleigh channel with $K = 200$. At this frame size, the difference between the two concatenated codes is less pronounced, but still $C_3 + CC_3$ has lower error rate (except in the first iteration). List decoding has higher coding gain under fully interleaved Rayleigh channel, because of added diversity arising from increased equivalent free distance of the code [12].

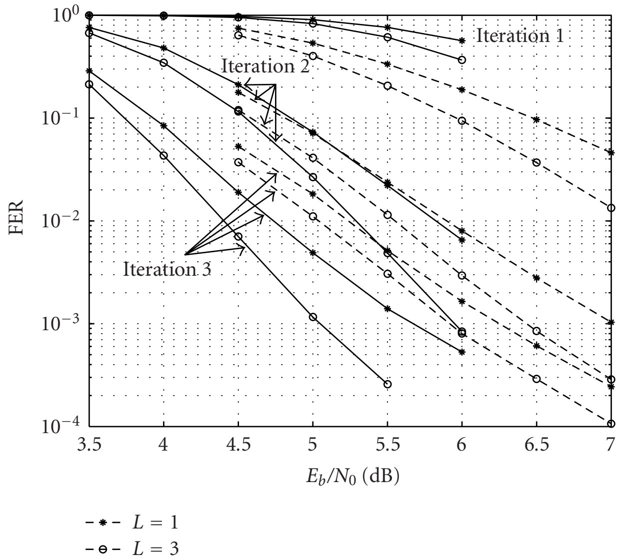

FIGURE 9: Iterative list decoding of $C_2 + CC_1$ (dashed) and $C_3 + CC_3$ (solid line) in fully interleaved Rayleigh channel, K=200.

The coding gain of $C_2 + CC_2$ at the fifth iteration for $L = 2$ is about 1.5 dB in Rayleigh fading, and 0.75 dB with $L = 5$ in AWGN channel. We refer the interested reader to [6] for further results on this code.

5. CONCLUSION

We propose an iterative list decoder for VLC-based source-channel codes. The iterative decoding of source-channel codes is made possible by the residual redundancy in the source code. Some source coders, such as H.263+, include additional redundancy for error resilience, making a source-channel decoder more desirable. It is shown that the amount of the redundancy in the VLC plays an important role in the performance of the code, given a total rate constraint. The list decoder is made possible by a nonbinary CRC code which also provides a stopping criterion for the iterative decoder. At a given iteration of the iterative decoder, the proposed list decoder improves the overall performance of the system. Extensive experimental results are provided in AWGN and fully interleaved Rayleigh channels.

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EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

Special Issue on

Advances in Blind Source Separation

Call for Papers

Almost every multichannel measurement includes mixtures of signals from several underlying sources. While the structure of the mixing process may be known to some degree, other unknown parameters are necessary to demix the measured sensor data. The time courses of the source signals and/or their locations in the source space are often unknown a priori and can only be estimated by statistical means. In the analysis of such measurements, it is essential to separate the mixed signals before beginning postprocessing.

Blind source separation (BSS) techniques then allow separation of the source signals from the measured mixtures. Many BSS problems may be solved using independent component analysis (ICA) or alternative approaches such as sparse component analysis (SCA) or nonnegative matrix factorization (NMF), evolving from information theoretical assumptions that the underlying sources are mutually statistically independent, sparse, smooth, and/or nonnegative.

The aim of this special issue is to focus on recent developments in this expanding research area.

The special issue will focus on one hand on theoretical approaches for single- and multichannel BSS, evolving from information theory, and especially on nonlinear blind source separation methods, and on the other hand or their currently ever-widening range of applications such as brain imaging, image coding and processing, dereverberation in noisy environments, and so forth.

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Special Issue on Tracking in Video Sequences of Crowded Scenes

Call for Papers

Object tracking in live video is an enabling technology that is in strong demand by large application sectors, such as video surveillance for security and behavior analysis, traffic monitoring, sports analysis for enhanced TV broadcasting and coaching, and human body tracking for human-computer interaction and movie special effects.

Many techniques and systems have been developed and demonstrated for tracking objects in video sequences. The specific goal of this special issue is to provide a status report regarding the state of the art in object tracking in crowded scenes based on the video stream(s) of one or more cameras. The objects can be people, animals, cars, and so forth. The cameras can be fixed or moving. Moving cameras may pan, tilt, and zoom in ways that may or may not be communicated to the tracking system.

All papers submitted must address at least the following two issues:

- Processing of live video feeds

For many applications in surveillance/security and TV sports broadcasting, the results of processing have value only if they can be provided to the end user within an application-defined delay. The submitted papers should present algorithms that are plausibly applicable to such incremental ("causal") processing of live video feeds, given suitable hardware.

- Handling of crowded scenes

Crowded-scene situations range from relatively simple (e.g., players on a planar field in a soccer match) to very difficult (e.g., crowds on stairs in an airport or a train station). The central difficulties in crowded scenes arise from the constantly changing occlusions of any number of objects by any number of other objects.

Occlusions can be resolved to some degree using a single video stream. However, many situations of occlusion are more readily resolved by the simultaneous use of several cameras separated by wide baselines. In addition to resolving ambiguities, multiple cameras also ease the exploitation of 3D structure, which can be important for trajectory estimation or event detection.

Topics of interest include principles and evaluation of relevant end-to-end systems or important components thereof, including (but not limited to):

- Handling of occlusions in the image plane in single-camera scenarios
- Handling of occlusions in a world coordinate system (3D, possibly degenerated to 2D) in single- or multi-camera scenarios
- Fusion of information from multiple cameras and construction of integrated spatiotemporal models of dynamic scenes
- 3D trajectory estimation
- Tracking of multiple rigid, articulated, or nonrigid objects
- Automatic recovery of camera pose from track data
- Detection and recognition of events involving multiple objects (e.g., offside in soccer)

Papers must present a thorough evaluation of the performance of the system or method(s) proposed in one or more application areas such as video surveillance, security, sports analysis, behavior analysis, or traffic monitoring.

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Special Issue on

Advances in Subspace-Based Techniques for Signal Processing and Communications

Call for Papers

Subspace-based techniques have been studied extensively over the past two decades and have proven to be very powerful for estimation and detection tasks in many signal processing and communications applications. Such techniques were initially investigated in the context of super-resolution parametric spectral analysis and the related problem of direction finding. During the past decade or so, new potential applications have emerged, and subspace methods have been proposed in several diverse fields such as smart antennas, sensor arrays, system identification, time delay estimation, blind channel estimation, image segmentation, speech enhancement, learning systems, and so forth.

Subspace-based methods not only provide new insight into the problem under investigation but they also offer a good trade-off between achieved performance and computational complexity. In most cases they can be considered as low cost alternatives to computationally intensive maximum likelihood approaches.

The interest of the signal processing community in subspace-based schemes remains strong as is evident from the numerous articles and reports published in this area each year. Research efforts are currently focusing on the development of low-complexity adaptive implementations and their efficient use in applications, numerical stability, convergence analysis, and so forth.

The goal of this special issue is to present state-of-the-art subspace techniques for modern applications and to address theoretical and implementation issues concerning this useful methodology.

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

- Efficient and stable subspace estimation and tracking methods
- Subspace-based detection techniques
- Sensor array signal processing
- Smart antennas
- Space-time, multiuser, multicarrier communications
- System identification and blind channel estimation
- State-space model estimation and change detection
- Learning and classification
- Speech processing (enhancement, recognition)
- Biomedical signal processing
- Image processing (face recognition, compression, restoration)

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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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EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

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 EMBEDDED SYSTEMS

Special Issue on

Signal Processing with High Complexity: Prototyping and Industrial Design

Call for Papers

Some modern applications require an extraordinary large amount of complexity in signal processing algorithms. For example, the 3rd generation of wireless cellular systems is expected to require 1000 times more complexity when compared to its 2nd generation predecessors, and future 3GPP standards will aim for even more number-crunching applications. Video and multimedia applications do not only drive the complexity to new peaks in wired and wireless systems but also in personal and home devices. Also in acoustics, modern hearing aids or algorithms for de-reverberation of rooms, blind source separation, and multichannel echo cancelation are complexity hungry. At the same time, the anticipated products also put on additional constraints like size and power consumption when mobile and thus battery powered. Furthermore, due to new developments in electroacoustic transducer design, it is possible to design very small and effective loudspeakers. Unfortunately, the linearity assumption does not hold any more for this kind of loudspeakers, leading to computationally demanding nonlinear cancelation and equalization algorithms.

Since standard design techniques would either consume too much time or do not result in solutions satisfying all constraints, more efficient development techniques are required to speed up this crucial phase. In general, such developments are rather expensive due to the required extraordinary high complexity. Thus, de-risking of a future product based on rapid prototyping is often an alternative approach. However, since prototyping would delay the development, it often makes only sense when it is well embedded in the product design process. Rapid prototyping has thus evolved by applying new design techniques more suitable to support a quick time to market requirement.

This special issue focuses on new development methods for applications with high complexity in signal processing and on showing the improved design obtained by such methods. Examples of such methods are virtual prototyping, HW/SW partitioning, automatic design flows, float to fix conversions, automatic testing and verification, and power aware designs.

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