André Fonseca dos Santos

Efficient Receiver Methods for Coded Systems Under Channel Uncertainty

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Efficient Receiver Methods for Coded Systems Under Channel Uncertainty

André Fonseca dos Santos

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Abstract

Traditionally, communication systems have been employing pilots in order to estimate the impulse response of multipath channels. However, with the growing demand of high data rates, the overhead due to the pilots might be a significant issue. In order to minimize such overhead, one can exploit the redundancy introduced by channel codes (e.g. convolutional and LDPC codes) for the channel estimation task. Indeed, the complete elimination of the pilot overhead exploiting solely the channel code redundancy is investigated (i.e. the code allows blind successful channel estimation).

This thesis aims to derive and analyze efficient algorithms that exploit the channel code redundancy for blind channel estimation. It investigates blind algorithms that exploit the structure imposed by channel coding on the transmitted sequence. It is shown that the sole exploitation of the code for blind channel estimation is very sensitive to the code constraints. Hence, traditional blind approaches such as the super exponential algorithms are combined with the code-based approach in order to derive efficient algorithms that do not strongly depend on the code characteristics. This combination results in a robust blind estimator that considerably relaxes the required codes constraints for the channel estimation task. It is also shown that the main advantage of exploiting the code for channel estimation is to remove all the ambiguities that are inherent to traditional blind approaches.

Additionally, a turbo receiver is proposed in order to further improve the quality of the channel estimate. Efficient and low-complex algorithms are introduced in order to exploit the soft feedback of the decoder with the role of iteratively refining the channel estimate. An algorithm where only an interference cancellation and a correlation step are performed is proposed and analytically characterized. Additionally, it is shown via Monte Carlo simulation that the proposed blind turbo receiver is able to perform as well as a turbo equalizer with perfect channel knowledge.

Moreover, with the goal of providing more insight with respect to the proposed algorithms, the Extrinsic Information Transfer (EXIT) charts are modified in order to analyze the interplay between the blind and the soft channel estimation phases. It is demonstrated that only a rough estimate of the channel in the first iteration of the turbo receiver suffices in order to perform as well as with perfect channel knowledge. Indeed, it is shown that only a negligible amount of pilots suffices. Moreover, the blind channel estimation approaches proposed here speed up the turbo receiver convergence in comparison to using a negligible amount of pilots. In summary, it is shown that the exploitation of the code reduces or even completely eliminates the necessity of employing pilots.

Zusammenfassung

Traditionell setzen Kommunikationssysteme Pilotsequenzen zur Schätzung der Impulsantwort des Mehrwege-Übertragungskanals ein. Der Verwendung von Pilotsequenzen reduziert jedoch inhärent die spektrale Effizienz der Übertragung und ist, infolge der steigenden Nachfrage nach höheren Übertragungsdatenraten, ein kritisches Problem geworden. Um diesen Verlust der spektralen Effizienz zu minimieren, kann die, durch die Fehlerschutzcodierung (z.B. Faltungscodes oder LDPC Codes) eingefügte Redundanz für die Schätzung des Übertragungskanals ausgenutzt werden. Deshalb wird in dieser Arbeit die Fragestellung untersucht, inwieweit die eingefügte Redundanz für die Kanalschätzung ausreichend ist und ob auf die Verwendung von Pilotsequenzen zur Kanalschätzung vollständig verzichtet werden kann.

Diese Arbeit zielt auf die Herleitung und Analyse von Algorithmen ab, welche die Redundanz des Fehlerschutzcodes für eine blinde Kanalschätzung ausnutzen. Es werden blinde Schätzalgorithmen untersucht, welche die von der Fehlerschutzcodierung dem Sendesignal aufgeprägte Struktur ausnutzen. Es wird gezeigt, dass die Leistungsfähigkeit blinder Schätzverfahren, welche lediglich die Codeeigenschaften ausnutzen, sehr empfindlich gegenüber der Codestruktur ist. Deshalb werden traditionelle, blinde Schätzansätze, wie z.B. Super Exponential Algorithmen mit codebasierten Ansätzen kombiniert, um effizientere Algorithmen zu entwickeln, deren Leistungsfähigkeit weniger von der Codestruktur beeinflusst wird. Diese Kombination ermöglicht ein robustes, blindes Schätzverfahren, welches die Anforderungen der blinden Kanalschätzung an den Fehlerschutzcode deutlich senkt. Weiterhin wird gezeigt, dass der Hauptvorteil einer Kanalschätzung, basierend auf der Ausnutzung der Codestruktur, darin besteht, dass die inhärente Mehrdeutigkeit von traditionellen, blinden Schätzverfahren vermieden werden kann.

Zur weiteren Verbesserung der Kanalschätzung wird ein Turbo-Empfänger entwickelt. Leistungsfähige Algorithmen mit einer geringen rechentechnischen Komplexität werden vorgestellt, um die Rückkopplung der Zuverlässigkeitswerte des Decoders für eine iterative Verbesserung der Präzision des Kanalschätzergebnisses zu erzielen. Es wird ein Algorithmus vorgestellt, welcher lediglich aus einer Interferenzauslöschung und einer nachfolgenden Korrelation besteht . Des Weiteren wird die Leistungsfähigkeit des vorgeschlagenen Turbo-Algorithmus analytisch untersucht. Zudem wird mithilfe von Monte Carlo Simulationen gezeigt, dass der vorgeschlagene blinde Turbo-Empfänger die gleiche Leistungsfähigkeit erreicht wie ein Turbo-Entzerrer mit perfekter Kanalkenntnis.

Zur weiterführenden Analyse des vorgeschlagenen Algorithmus wird der Austausch zwischen der blinden und Turbo analschätzung mithilfe von modifizierten Extrinsic Information Transfer (EXIT) Diagramme untersucht. So wird gezeigt, dass bereits die ungefähre Kenntnis über den Kanal in der ersten Iteration des Turbo-Empfängers ausreicht, um die gleiche Leistungsfähigkeit zu erzielen wie im Fall einer perfekten Kanalkenntnis. Damit wird nachgewiesen, dass bereits vernachlässigbar wenige Piloten ausreichen, um ein präzises Kanalschätzergebnis zu erzielen. Zudem wird gezeigt, dass der vorgeschlagene, blinde Kanalschätzalgorithmus die Konvergenz des Turbo-Empfängers steigert. Zusammenfassend kann festgestellt werden, dass die Ausnutzung der Codeeigenschaften die Notwendigkeit der Nutzung von Piloten zur Kanalschätzung reduziert bzw. vollständig eliminiert.

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Abbreviations

AWGN	Additive White Gaussian Noise
BER	Bit Error Rate
BICM	Interleaved Coded Modulation
BPSK	Binary Phase Shift Keying
CC	Convolutional Codes
CIR	Channel Impulse Response
CRLB	Cramer-Rao Lower Bound
ESE	Effective Soft Equalizer
EXIT	Extrinsic Information Transfer
HOS	High Order Statistics
ISI	Intersymbol Interference
ISCE	Iterative Soft Correlator Estimator
LDPC	Low Density Parity Check
LLR	Log Likelihood Ratio
LS	Logic Stings
MAP	Maximum a Posteriori
ML	Maximum Likelihood
MI	Mutual Information
MIMO	Multiple Input Multiple Output
MSE	Mean Square Error
PAM	Pulse Amplitude Modulation
QPSK	Quaternary Phase Shift Keying
QAM	Quadrature Amplitude Modulation
SCE	Soft Correlator Estimator
SE	Super Exponential
SELS	Super Exponential with Logic Strings
SELS-SOS	Super Exponential + Logic Strings + Second Order Statistics
SNR	Signal to Noise Ratio
SOS	Second Order Statistics
SSI	Self Symbol Interference
TC	Turbo Code
UWB	Ultra Wide Band

Symbols

Vectors and matrices

$\mathbf{a}$	soft weights of SCE and ISCE algorithms
$\mathbf{b}$	information bits
$\hat{\mathbf{b}}$	estimates of the information bits
$\mathbf{c}$	coded bits
$\mathbf{c}'$	interleaved coded bits
$\mathbf{e}$	equalizer filter
$\mathbf{d}$	modulated bits relative to the payload data (when the blind mode is used $\mathbf{d} = \mathbf{s}$)
$\mathbf{f}$	composite impulse response of channel and equalizer
$\mathbf{F}$	Fourrier matrix
$\mathbf{G}$	generator matrix of a channel code
$\mathbf{h}$	Channel Impulse Response (CIR)
$\mathbf{H}$	Circular convolution channel matrix
$\mathbf{H}_{\mathcal{C}}$	parity-check matrix of a channel code
$\mathbf{r}$	received symbols from the multipath channel
$\mathbf{r}_e$	extended received vector
$\mathbf{s} = [\mathbf{t}^T\, \mathbf{d}^T]^T$	vector of transmitting symbols (including pilots and payload data)
$\mathbf{S}$	Circular convolution matrix composed by the transmitted symbols $\mathbf{s}$ as first collum
$\mathbf{t}$	training (pilot) symbols
$\mathbf{T}$	Toeplitz matrix of training (pilot) symbols
$\mathbf{x}_{\mathrm{d}}$	coded, interleaved and bipolar symbols related to the payload information
$\mathbf{x}_{\mathrm{t}}$	coded, interleaved and bipolar symbols related to the training sequence
$\mathbf{x} = \mathbf{x}_{\mathrm{d}} + \mathbf{x}_{\mathrm{t}}$	coded, interleaved and bipolar symbols related to the whole frame

$\widetilde{\mathbf{x}}_{n_\mathrm{d}} = \mathbf{x}_{[(n_\mathrm{d}-1)Q+1,\ldots,n_\mathrm{d}Q]}$	subset of vector $\mathbf{x}_\mathrm{d}$ used as input of the linear modulator
$\mathbf{w}$	vector with samples of the AWGN
$\mathbf{y}$	output of equalizer
$\mathbf{y}_e$	extended equalized vector
$\mathbf{z}$	weights used for linear modulation

Vector elements

a_k	soft weight of SCE and ISCE w.r.t. x_k
b_i	information bit
d_{k_d}	QAM symbol w.r.t. to payload
e_{L_e}	equalizer tap
c_{k_d}	coded bit
h_l	channel tap
$\hat{h}'_l$	estimate of the channel tap obtained from previous turbo equalizer iteration
$\hat{h}^{it}_l$	estimate of the channel tap obtained at the *it-th* iteration of the ISCE
$\hat{h}_l$	channel tap estimate
x_{k_d}	bipolar representation of coded bit
$\hat{s}_n$	soft symbol, i.e. estimate (expected value) of the symbol s_n
t_{n_t}	training QAM symbol
$\hat{x}_k$	soft bit, i.e. estimate (expected value) of x_n
z_q	modulation weight

Symbols

$\mathcal{A}$	set of vectors with the indices of the bits belonging to parity-check equations
$\mathrm{CRBL}_{\mathrm{pilot}}$	Cramer-Rao low bound for pilot-aided estimation
$\mathrm{CRBL}_{\mathrm{full}}$	Cramer-Rao low bound for the full a-priori knowledge case
E_b	bit energy
$H(f)$	transfer function of the channel
$\hat{H}(f)$	estimate of the transfer function of the channel
$\hat{H}(f)^{\star}$	estimate of the transfer function of the channel obtained from SELS
$\hat{H}(f)^{\star\star}$	estimate of the transfer function of the channel obtained from SOS

I	amount of information bits
I_{ext}^{E}	extrinsic mutual information of equalizer
I_{a}^{E}	a-priori mutual information of equalizer
I_{ext}^{D}	extrinsic mutual information of decoder
I_{ext}^{E}	a-priori mutual information of decoder
j	complex number
l	channel tap index
L	channel length
L_f	channel + equalizer length
K_{d}	amount of coded bits relative to the payload data
K_{t}	amount of coded bits relative to the pilot symbols
K	total amount of bits in the frame (pilots+payload)
$L^{E}(\mathbf{x})$	a-posteriori information of the soft equalizer
$L_{ext}^{E}(\mathbf{x})$	extrinsic information of the soft equalizer
$L_{a}^{E}(\mathbf{x})$	a-priori information of the soft equalizer
$L_{a}^{D}(\mathbf{c})$	a-priori information of the decoder
$L^{D}(\mathbf{c})$	a-posteriori information of the decoder
$L_{ext}^{D}(\mathbf{c})$	extrinsic information of the decoder
MSE	mean square error
$\sigma_{L_{app}}^2$	variance of a-posteriori LLRs
$\sigma_{L_{ext}}^2$	variance of extrinsic LLRs
$\sigma_{L_a}^2$	variance of a-priori LLRs
$\mu_{\hat{x}}$	mean of unimodal pdf of soft bit
$\mathrm{MSE}_{\hat{h}_l}$	MSE of the estimate *l-th* channel tap
MSE'	sum of MSEs of the estimate of all channel taps
MSE	sum of MSEs of the estimate of all channel taps obtained from previous turbo iteration
MSE_{∞}	asymptotic MSE of the ISCE
m	*m-th* logic string
M	amount of logic strings
N_{d}	amount of QAM symbols relative to the payload data
N_o	noise power spectral density
N	total amount of QAM symbols in the frame (pilots+payload)
$\mathcal{N}(\mu, \sigma^2)$	Gaussian random variable with mean μ and variance σ^2
P	Logic string length or check degree
Q	number of bits per symbol
$R_{\mathcal{C}} = \frac{I}{K_{\mathrm{d}}}$	code-rate
$R_{\mathcal{T}} = \frac{\text{payload}}{N} = \frac{N_{\mathrm{d}}}{N}$	pilot-rate

SNR	Signal to Noise Ratio
u	linear modulation spacing
v	position of main tap of the channel
v_f	position of main tap of the composite response of channel and equalizer
$\mathcal{X}$	set of vectors with the indices of the bits belonging to logic strings
Δ_{ISI_k}	residual ISI at the interference canceller's output
Δ_{SSI_k}	residual Self Symbol Interference at the interference canceller's output
$\pi_{m,p}$	index of p-th bit of the m-th logic string
$\tau_{m,p}$	index of p-th bit of the m-th check equation
$\sigma_{\hat{x}}^2$	variance of unimodal pdf of soft bit
σ_a^2	variance of unimodal pdf of soft weight
σ_w^2	variance (power) of AWGN

Operators

$E\{\}$	Expectation operator
$\mathrm{Var}\{\}$	Variance operator
$*$	convolution
$\|\|(\cdot)\|\|$	Euclidian norm
$\|(\cdot)\|$	absolute value
$\log(\cdot)$	modulo two addition
$\otimes$	Kronecker product
$\angle(\cdot)$	angle of a complex number
$\mathrm{argmin}(\cdot)$	argument that minimizes a function
$E\{\cdot\}$	cumulant operator
$E\{\cdot\}$	expectation operator
$\mathfrak{F}$	Fourier transform
$\mathfrak{F}^{-1}$	inverse Fourier transform
$J(\cdot)$	maps the variance of a LLR with Gaussian distribution to the mutual information value
$J^{-1}(\cdot)$	maps the mutual information value to the variance of a LLR with Gaussian distribution
$\max(\cdot)$	maximum of a function
$\oplus$	probability
$\tanh(\cdot))$	hyperbolic tangent
$\Pr(\cdot)$	probability
$p(\cdot)$	probability density function

$\mathrm{tr}(\cdot)$	the sum of the diagonal elements of a square matrix
$\Omega(\cdot)$	time-reverse operator
$\lceil\cdot\rceil$	closest upper integer of a ratio

Chapter 1

Introduction

1.1 Motivation

It is well known that the demand for higher data rates in communications systems has been growing considerably throughout the years. The number of users joining the existing mobile networks grows significantly, resulting at the need of efficient and clever exploitation of the spectrum resources. On top of the increasing amount of users in the system, the explosion of new applications further intensifies the transmission requirements.

However, the fulfillment of the emerging requirements of data-rate brings considerable technical challenges such as the estimation and equalization of the multipath effect of the channel. Traditionally, channel estimation is performed with the employment of pilot (training) symbols. Depending on the transmission scenario, the overhead due to pilots which is required to estimate the channel with a reasonable quality, might become significantly high. Nevertheless, the overhead of pilots for channel estimation contradicts the goal of a highly efficient[1] data-rate system. Therefore, the investigation of other alternatives for channel estimation is appealing for the successful development of efficient communication systems.

In order to come up with new solutions for channel estimation some properties of the transmitted signal must be exploited. Actually, any characteristic of the transmitted signal which is known a-priori might be useful for identifying the channel. For example, the already mentioned use of pilot symbols implies that there is full a-priori knowledge about them. Therefore, the received symbols can be compared with the transmitted pilots and the channel is estimated. However, other properties of the transmitted sequence can be exploited for the channel estimation purpose. For instance, one can take advantage of the non-Gaussianity of a finite-size modulated alphabet [BK94,CC06,Fon95,Gia87,JBK97, KJ94,Men91,NM93,SN03,Sch00,SW93,TP98]. Indeed, several algorithms in the literature exploit the statistics of non-Gaussian signals for the estimation of the channel without the use of pilots. However, to the best of the author's knowledge, none of the existing blind algorithms (i. e. without pilots) based on the statistics of the transmitted signal can

[1] Effective data-rate here stands for the rate of information to be transmitted.

completely estimate the channel. There is always some remaining estimation ambiguity that is present.

Another interesting idea for channel estimation is the exploitation of knowledge of the channel code structure. Traditionally, the channel code (e.g. convolutional codes, turbo codes, LDPC codes, etc....) adds redundancy to the transmitted signal with the aim of error-correction. Nevertheless, the following fundamental question can be raised:

- since there is already redundancy added by the channel code, is it really necessary to add further redundancy by the pilots for channel estimation?
- how can the channel code redundancy be exploited for the channel estimation purpose?

Actually, the addition of pilots can be considered as an unwise manner of channel coding, where no error correction capability is added since the redundancy (pilot symbols) is uncorrelated with the information. On the other hand, the error-correcting code adds redundancy which is *correlated* with the information allowing an improvement of performance of the system w.r.t. the Bit Error Rate (BER). However, the way of exploiting this kind of redundancy might be not so straightforward as with pilot symbols.

In order to estimate the channel based on the code redundancy some property of the code which is known a-priori has to be exploited. In fact, the parity-check equations of a code are known by the receiver. Hence, in contrast to the pilot redundancy that gives *direct* knowledge of the symbols, the channel code provides *indirect* knowledge about the relation among the transmitted bits that are connected to a check equation. For instance, in [SKK03] the high order statistics imposed by the parity check-equations of the code are used to blindly estimate the channel when BPSK constellations are employed.

Another way of taking advantage of the channel code for channel estimation is to exploit the soft feedback of the decoder in a turbo receiver. The principle is to iteratively refine the estimates of the channel using the decoder decisions. In this way, the code redundancy is implicitly exploited.

Therefore, the investigation of channel estimation under coded systems must take into account the mentioned alternative in order to avoid the "waste" of further redundancy introduced by pilots.

1.2 Objective of this Thesis

The objective of this thesis can be summarized with the investigation of the answers to the following questions:

1. How to efficiently exploit the code redundancy for channel estimation?
2. By which degree can the amount of required pilots be reduced by the exploitation of the channel code redundancy?

3. Is it really necessary to use pilot symbols or does the exploitation of the code suffice?

In order to answer question 1, this thesis shows efficient algorithms that exploit the code knowledge for channel estimation. Traditional blind algorithms exploiting the non-Gaussianity of signals are combined with the channel code structure in order to blindly estimate the channel. Furthermore, low-complex and efficient algorithms that refine the channel estimates based on the decoder's soft feedback are proposed and characterized.

The performance of the receiver is characterized using the algorithms proposed through the thesis and questions 2 and 3 are investigated under some selected scenarios.

Finally, this thesis aims to investigate effective solutions for channel estimation in systems employing channel coding. Indeed, throughout the chapters it will be shown with examples that only a few pilots or even no pilots at all are required in coded systems.

1.3 Outline of the Thesis

Chapter 2 introduces the transmitter and receiver model. The turbo receiver is explained where two phases of channel estimation in the receiver are highlighted: the bootstrapping phase (when no feedback of the decoder is available) and the soft channel estimation phase (where the decoder's feedback is exploited).

In Chapter 3, the algorithms for the bootstrapping phase are considered. First, the pilot-aided and traditional blind algorithms based on the statistics of the transmitted signal are briefly reviewed. Moreover the blind channel estimation algorithm introduced in [SKK03] using channel codes in BPSK systems is reviewed. Finally, an extension to QAM constellations is proposed and its advantages and drawbacks are exposed. In order to come up with a robust blind receiver, the parity-check knowledge is combined with the statistics of second and fourth order of the transmitted signal. The simulation results show that the proposed combination estimates the channel blindly without any kind of ambiguity and without too much dependency on the employed code.

Chapter 4 proposes low-complex and efficient algorithms for the estimation of the channel based on the soft decoder's feedback. The proposed algorithms are fully characterized analytically. Finally, simulation results of the whole turbo receiver including bootstrapping and soft channel estimation are shown.

In Chapter 5 the behavior of the whole receiver is analyzed using the (Extrinsic Information Transfer) EXIT charts. It is shown that the EXIT analysis of the receiver performing channel estimation, equalization and decoding is a complicated and multidimensional problem. A framework that converts the analysis to a simple two dimensional approach is proposed. This framework allows to estimate how good the bootstrapping must be, answering Questions 2 and 3 of the previous section.

Finally, Chapter 6 summarizes and discusses the conclusions which result from this thesis. Furthermore, the opportunities for future research are pointed out.