ABSTRACT
We utilize Extrinsic Information Transfer (EXIT) charts to optimize the power allocation in a multiuser CDMA system. We investigate two methods to obtain the optimal power levels: the first minimizes the total power; the second minimizes the reduction in implementation complexity. Which on the other hand provides tremendous dynamic range caused by fixed-point arithmetic, causes much degradation, for example, reduced or less specialized hardware. This transition usually requires an implementation in more adaptable hardware. The practical success of the iterative turbo decoding algorithm has inspired its adaptation to other code classes, notably serially concatenated codes, and has rekindled interest in low-density parity-check codes, which give the definitive historical precedent in iterative decoding. The serial concatenated configuration holds particular interest for communication systems, since the “inner encoder” of such a configuration can be given more general interpretations, such as a “parasitic” encoder induced by a convolutional channel or by the spreading codes used in CDMA. The corresponding iterative decoding algorithm can then be extended into new arenas, giving rises to turbo equalization or turbo CDMA, among doubtless other possibilities. Such applications demonstrate the power of iterative techniques which aim to jointly optimize receiver components, compared to the traditional approach of adapting such components independently of one another. Algorithms are often developed and tested in floating-point environments on GPPs in order to show the achievable optimal performance. Besides shortest development time, there are no requirements on, for example, processing speed or power consumption, and hence this platform is the best choice for the job. However, speed or power constraints might require an implementation in more or less specialized hardware. This transition usually causes much degradation, for example, reduced dynamic range caused by fixed-point arithmetic, which on the other hand provides tremendous reduction in implementation complexity.

II.CHANNEL CODING AND DECODING
This chapter deals with basics of channel coding and its decoding algorithms. Following is a brief description of the simple communication model that is assumed in the sequel. This model also helps to understand the purpose of channel coding. Then, two popular coding approaches are discussed more thoroughly: convolutional coding together with Gray-mapped signal constellations and set-partition coding. Decoding algorithms are presented from their theoretical background along with a basic complexity comparison.

Consider the block diagram of the simplified communication system in Figure 2.1. It consists of an information source (not explicitly drawn) that emits data symbols \{u_k\}. A channel encoder adds some form of redundancy, possibly jointly optimized with the modulator, to these symbols to yield the code symbol sequence \{c_k\}, where \(c_k\) denotes an Mary transmission symbol. Linear modulation is assumed, that is, modulation is based on a linear superposition of (orthogonal) pulses. The signal sent over the channel is therefore

\[
    s(t) = \sum_k c_k \cdot w(t - kT_s),
\]

Where \(w(\cdot)\) is the pulse waveform and \(T_s\) is the symbol time. The waveform channel adds uncorrelated noise \(n(t)\) to the signal, which results in the waveform \(r(t)\) at the receiver. For the remainder, the disturbance introduced by the channel is assumed to be additive white Gaussian noise (AWGN). That is,

\[
    \mathbb{E}\{n(t)\} = 0
\]

\[
    \mathbb{E}\{|n(t)|^2\} = N_0/2.
\]

The received waveform \(r(t)\) is demodulated to yield a discrete sequence of (soft) values \(\hat{u}_k\). Based on these values, the channel decoder puts out an estimate \(\hat{u}_k\) for the data symbols \(u_k\).
Figure 2.1: A simplified communication system.

According to Shannon [85], reliable communication with arbitrarily low bit error rate (BER) in the AWGN channel can be achieved for transmission rates below

\[ C = \frac{1}{2} \log_2 \left( 1 + \frac{2E_b}{N_0} \right) \text{(bits/dimension)}. \]

If there are \( J \) orthogonal signal dimensions per channel use, the transmission rate of a (coded) communication system is defined as

\[ R_d = \frac{\log_2 M}{J} \cdot R_c \text{ (bits/dimension)}, \quad (2.1) \]

where \( M \) is the number of possible symbols per channel use and \( R_c < 1 \) denotes the code rate of the channel code in data bits/code bits. For example, a communication system with a channel code of rate \( R_c = 1/2 \) per channel use and a 16-QAM constellation, that is, \( M = 16 \) and \( J = 2 \), has a transmission rate of \( R_d = 1 \) bit/dimension.

\[ E_s = \frac{1}{M} \sum_{i=1}^{M} |c_i|^2. \]

For equiprobable signaling, the energy devoted to a transmission symbol is expressed as or, alternatively, the energy per data bit is

\[ E_b = \frac{E_s}{\log_2 M \cdot R_c}. \quad (2.2) \]

### 2.1 CHANNEL CODING:

A good channel code reduces the necessary \( E_b \) to achieve the same BER over a noisy channel as an uncoded transmission system of equal transmission rate \( R < C \). This reduction is referred to as coding gain. The BER of many communication systems can be estimated in closed form based on the union bound [78]. Essentially, BER depends on the two-signal error probability, that is, the probability that one signal is mistaken for another upon decoding, and the minimum distance between signals. This probability resembles

\[ p_e \approx \frac{2K}{M} Q \left( \frac{d_{\text{min}}}{N_0} \right). \quad (2.3) \]

where \( K \) is the number of signal pairs that lie at distance \( d_{\text{min}} \) apart from each other and \( Q(\cdot) \) is the complementary error function defined as

\[ Q(x) = \frac{1}{\sqrt{2\pi}} \int_{x}^{\infty} \exp(-u^2/2) \, du. \]

In practice, BER is estimated by computer simulations of the underlying communication model.

From Equation 2.3 the task of the channel code (together with the modulator) becomes apparent: either increase \( d_{\text{min}} \), or decrease \( 2K/M \), or both. Then, \( E_b \) can be lowered for the same BER.

There are two major classes of binary channel codes: block codes and convolutional codes. In the context of this thesis, only the latter codes are considered since they are widely applied in today’s communication systems. Nevertheless, the rediscovery of low-density parity-check codes [49] might reclaim some share from convolutional-based coding in these systems in the near future.

### 2.2 DECODING ALGORITHMS:

From the considerations in Section 2.1.1, the trellis created by a convolutional encoder can be interpreted as finite-state discrete-time Markov source. Denote by \( X_k \in [0, N - 1], k \in \mathbb{Z} \), a possible state of the encoder at time \( k \). At the receiver side, the probability of a trellis transition from state \( X_k \) to \( X_{k+1} \) and the outcome \( y_k \) is given by

\[ p(x_{k+1}, y_k | x_k) = p(y_k | x_k, x_{k+1}) \cdot \Pr(x_{k+1} | x_k) \]

(2.4)

Here \( p(y_k | x_k, x_{k+1}) \) is the likelihood function of the received symbol \( y_k \) given the transition \( (x_k, x_{k+1}) \) and \( \Pr(x_{k+1} | x_k) \) is the transition’s \textit{a priori} probability. For convolutional codes, there are \( c \) code symbols along a trellis branch and thus \( y_k = (y_0, k \cdot \cdot \cdot , y_{c-1}, k) \). Depending on the code rate \( R_c \) and the transmission scheme, these \( y_i,k \) stem from one or several i.i.d. code symbols. For TCM codes, there are subsets along the branches. These subsets consist of two-dimensional signals and \( y_i \) is a two-dimensional signal. When a demodulated noisy value \( y_k \) is received from an AWGN channel with variance \( \sigma^2 = N_0/2 \), the likelihood function becomes

\[ p(y_k | x_k, x_{k+1}) = \frac{1}{\sqrt{\pi N_0}} \exp \left( -\frac{|y_k - c_{k}|^2}{N_0} \right). \]

One can take the logarithm of Equation 2.8 and scale with \(-N_0\) to yield the branch metric (BM)

\[ \lambda(x_{k+1}, x_k) = -N_0 \log p(X_{k+1}, y_k | X_k) \]

\[ = |y_k - c_k| \cdot N_0 \log \frac{1}{\sqrt{\pi N_0}} \]

(2.5)

The first term in Equation 2.5 corresponds to the squared Euclidean distance between the received symbol \( y_k \) and the expected symbol \( c_k \) along the
branch \((X_k, X_{k+1})\). The second term is the weighted a priori probability of the branch. The constant can be neglected in the calculations since it contributes equally to all \( \lambda(\cdot) \). Based on the previous notations, consider a received symbol sequence \( y = \{ y_k \} \). Since the channel is memoryless, maximum likelihood (ML) and maximum a posteriori (MAP) sequence estimates can be expressed as finding the I that achieves respectively:

\[
\min_i \| y - c_i \|^2
\]  
(2.6)

And

\[
\min_i \left\{ \| y - c_i \|^2 - N_0 \sum_i \log \Pr(X_{i+1} | X_i) \right\}
\]  
(2.7)

Clearly, ML and MAP decoders would estimate the same symbol sequence if all symbols were equally likely, that is, the a priori probability is equal for all branches. Then, the second term in Equation 2.11 is the same for all branches \((X_i, X_{i+1})\), and can thus be removed in calculating the branch metrics. If there is a priori information about the transition, though, the decoding might give different results for ML and MAP. In any case, ML minimizes the sequence error probability, whereas MAP can be set up so as to minimize the bit error probability [8].

III. TURBO CODES

In information theory, turbo codes (originally in French Turbo codes) are a class of high-performance forward error correction (FEC) codes developed in 1993, which were the first practical codes to closely approach the channel capacity, a theoretical maximum for the code rate at which reliable communication is still possible given a specific noise level. Turbo codes are finding use in (deep space) satellite communications and other applications where designers seek to achieve reliable information transfer over bandwidth- or latency-constrained communication links in the presence of data-corrupting noise. Turbo codes are nowadays competing with LDPC codes, which provide similar performance.

SOFT DECISION APPROACH:

The decoder front-end produces an integer for each bit in the data stream. This integer is a measure of how likely it is that the bit is a 0 or 1 and is also called soft bit. The integer could be drawn from the range \([-127, 127]\), where:

- 127 means "certainly 0"
- 100 means "very likely 0"
- 0 means "it could be either 0 or 1"
- 100 means "very likely 1"
- 127 means "certainly 1"
- etc.

This introduces a probabilistic aspect to the data-stream from the front end, but it conveys more information about each bit than just 0 or 1.

4.1 EXISTING SYSTEM:

The turbo decoding algorithm for error-correction codes is known not to converge, in general, to a maximum likelihood solution, although in practice it is usually observed to give comparable performance. The quest to understand the convergence behavior has spawned numerous inroads, including extrinsic information transfer (or EXIT) charts, density evolution of intermediate quantities, phase trajectory techniques, Gaussian approximations which simplify the analysis, and cross-entropy minimization, to name a few. Some of these analysis techniques have been applied with success to other configurations, such as turbo equalization. Connections to the belief propagation algorithm have also been identified, which approach in turn is closely linked to earlier work(6) on graph theoretic methods. In this context, the turbo decoding algorithm gives rise to a directed graph having cycles; the belief propagation algorithm is known to converge provided no cycles appear in the directed graph, although less can be said in general once cycles appear. Interest in turbo decoding and related topics now extends beyond the communications community, and has been met with useful insights from other fields; some references in this direction include which draws on nonlinear system analysis, which draws on computer science, in addition to (predating turbo codes) and (more recent) which inject ideas from statistical physics, which in turn can be rephrased in terms of information geometry. Despite this impressive pedigree of analysis techniques, the “turbo principle” remains difficult to master analytically and, given its fair share of specialized terminology if not a certain degree of mystique, is often perceived as difficult to grasp to the non specialist. In this spirit, the aim of this paper is to provide a reasonably self-contained and tutorial development of iterative decoding for parallel and serial concatenated codes. The paper does not aim at a comprehensive survey of available analysis techniques and implementation tricks surrounding iterative decoding, but rather chooses a particular ad- vantage point which steers clear of unnecessary sophistication and avoids approximations.

4.2 PROPOSED SYSTEM:

The project work focuses on joint optimization of the power and decoding schedule is prohibitively complex so we break the optimization in two parts and first optimize power levels of each user then optimize the decoding schedule using the optimized power levels. Large gains in power efficiency and complexity can be achieved simultaneously. Furthermore, our optimized receiver has a lower convergence threshold and requires less iterations to achieve convergence than a conventional receiver. We show that our proposed optimization results in a more consistent quality of service (QoS).

IV. SYSTEM DESCRIPTION

The major advantage of dynamic scheduling over static scheduling is that the method compensates for performance better/worse than expected (average) due to differences in channel conditions over decoding blocks, or differences in the decoding trajectory. Using dynamic scheduling we have a more reliable receiver or similar complexity.

V. IMPLEMENTATION

Implementation of any software is always preceded by important decisions regarding selection of the platform, the language used, etc. These decisions are often influenced by several factors such as real environment in which the system works, the speed that is required, the security concerns, and other implementation specific details. There are three major implementation decisions that have been made before the implementation of this project. They are as follows:

1. Selection of the platform (Operating System).
2. Selection of the programming language for development of the application.
3. Coding guideline to be followed.

3.3 IMPLEMENTATION REQUIREMENTS:

SOFTWARE REQUIREMENT:
- The language chosen for this project is Java Swing and software used is NetBeans 6.8.
- Operating System used: Microsoft windows XP

RESULTS:

VI. CONCLUSION

We have optimized a turbo MUD receiver for unequal power turbo-coded CDMA system through EXIT chart analysis. The results in prior works were used to derive effective EXIT functions for FEC decoders and an interference canceller which enabled analysis of the system as in the equal power case. We utilized a nonlinear constrained optimization as in prior work to optimize the power levels of groups of users in the system. We modified the algorithm proposed in prior work to dynamically derive the optimal decoding schedule for the IMUD receiver. We then showed through simulation that this power optimized system using dynamic scheduling achieves
similar BER performance as a conventional receiver with significant complexity savings. Furthermore it outperforms the statically derived optimal schedule through reducing the variance of the per packet BER. We also proposed a method for estimating the SNR in an AWGN CDMA channel and showed that power and schedule may be optimized without any trade-off. Finally, we determined that a combination of static and dynamic scheduling offers the best benefit for the cost.

REFERENCES

12. 3GPP TS 25.104 V5.9.0; 3rd generation partnership project; technical specification group radio access network;base station (BS) radio transmission and reception (FDD) (release 5),” Sept. 2004.