DUAL REPEAT PUCTURED TURBO CODES IMPLEMENTATION ON FADING CHANNELS

By

SELVA KUMAR.P

Reg. No. 0920107022

of

KUMARAGURU COLLEGE OF TECHNOLOGY

(An Autonomous Institution affiliated to Anna University of Technology, Coimbatore)

COIMBATORE – 641 049

A PROJECT REPORT

Submitted to the

FACULTY OF ELECTRONICS AND COMMUNICATION ENGINEERING

In partial fulfillment of the requirements

for the award of the degree

of

MASTER OF ENGINEERING

IN

COMMUNICATION SYSTEMS

APRIL 2011

BONAFIDE CERTIFICATE

Certified that this project report entitled "**DUAL REPEAT PUNCTURED TURBO CODES IMPLEMENTATION ON FADING CHANNELS**" is the bonafide work of Mr. P. Selvakumar [Reg. no. 0920107022] who carried out the research under my supervision. Certified further, that to the best of my knowledge the work reported herein does not form part of any other project or dissertation on the basis of which a degree or award was conferred on an earlier occasion on this or any other candidate.

(Prof. Govindaraju. S)
Project Guide

(Dr. Rajeswari Mariappan)
Head of the Department

The candidate with university Register no. 0920107022 is examined by us in the project viva-voce examination held on ..21..04..2011..

Internal Examiner

External Examiner

ACKNOWLEDGEMENT

I wish to acknowledge my sincere gratitude and heartfelt thanks to our beloved principal **Dr. S. Ramachandran Ph.D.,** for having given me the adequate support and opportunity for completing this project work successfully.

I express my sincere thanks to **Dr. Rajeswari Mariyappan Ph.D.,** the ever active, Head of the Department of Electronics and Communication Engineering, who rendered guidance throughout this project

In particular, I wish to thank and everlasting gratitude to the project coordinator **D. Mohanageetha (Ph.D).,** Associate Professor, Department of Electronics and Communication Engineering for her expert counseling and guidance to make this project to a great deal of success. With his careful supervision and ensured me in the attaining perfection of work.

I extend my heartfelt thanks to my internal guide **Prof. S. Govindaraju M.E.,** Professor, Department of Electronics and Communication Engineering for his ideas and suggestion, which have been very helpful for the completion of this project work.

Last, but not the least, I would like to express my gratitude to my family members, friends and to all my staff members of Electronics and Communication Engineering department for their encouragement and support throughout the course of this project.

Turbo Codes is a powerful branch of Error-Control-Coding that has exhibited performances close to the Shannon limit, whilst offering a reasonable level of computational complexity. Repeat-Punctured Turbo Codes(RPTCs) exploit the fact that the use of large interleavers correspond to better Bit-Error Rate (BER) performances for the conventional turbo code, and makes the use of repetition and puncturing to improve the performance of the conventional turbo code for moderate to high signal-to-noise ratios (SNRs).

Motivated by Turbo Codes and Repeat-Punctured Turbo Codes, the project work attempts use of new scheme, Dual-Repeat-Punctured Turbo Codes (DRPTC) on the Fading channels. DRPTCs primarily make use of repetition and puncturing and the structure of the conventional turbo code to yield a more superior performance. The decoding in principle follows the technique of turbo decoding but with some changes to yield the performance improvement. To compare Performance analysis of AWGN channel and the Fading channel has also been correct out.

LIST OF FIGURES

ARQ -------- Automatic Repeat Request

AWGN -------- Additive White Gaussian Noise

BER -------- Bit Error Rate

BPSK --------- Binary Phase Shift Keying

CDMA --------- Code Division Multiple Access

DMC --------- Discrete Memoryless Channel

FER ---------- Frame Error Rate

FEC ---------- Forward Error Correction

IODS ---------- Input-Output Distance Spectrum

MAP --------- Maximum A Posteriori Algorithm

MER ---------- Message-Error-Rate

NRC ----------- Non-Recursive Convolutional Code

NSC ----------- Non-Systematic Convolutional Code

PCCC ----------- Parallel Concatenated Convolutional Codes

PSK ----------- Phase Shift Keying

QPSK ----------- Quadrature Phase Shift Keying

RSC ------------ Recursive Systamatic Convolutional Code

RV ------------ Random Variable

RPTC ------------ Repeat Punctured Turbo Codes

DRPTC ----------- Dual Repeat Punctured Turbo Codes

SCCC ----------- Series Concatenated Convolutional Codes

SISO ----------- Soft Input Soft Output

SNR ------------ Signal to Noise Ratio

SOVA ------------ Soft-Output Viterbi Algorithm

TCM ------------ Trellis Coded Modulation

TDMA ----------- Time Division Multiple Access

CHAPTER 1

INTRODUCTION

1.1 NEED FOR CODING

The development of digital communication is given impetus by three prime needs.

1. Greatly increased demands for data transmission of every form, from computer data banks to remote entry data terminals for a variety of applications, with ever increasing accuracy.

2. Rapid evolution of synchronous artificial satellite relays which facilitate worldwide communication at very high data rates, but whose launch costs, and consequent power and bandwidth limitations, impose a significant economic incentive on the efficient use of the channel resources.

3. Data communication networks which must simultaneously service many different users with a variety of rates and requirements, in which simple and efficient multiplexing of data and multiple access of channels is a primary economic concern.

These requirements and the solid-state electronic technology needed to support the development of efficient, flexible and error-free digital system evolved simultaneously. With unique intuition, Shannon perceived that, the goals of error-free digital communication on noisy channels and of maximally efficient conversion of analog signals to digital were dual facets of the same problem and that they share a common framework and hence, virtually a common solution.

The task facing the designer of a digital communication system is providing a cost-effective system for transmitting from a sender (at one end of the system) at a rate and level of reliability that are acceptable to a user (at the other end). The two key system parameters available to the designer are transmitted signal power and channel bandwidth. These two parameters together with the power spectral density of receiver noise, determine the signal energy bit-to-noise power density ratio E_b/N_o. For fixed E_b/N_o only the practical option available is to use coding.

Another practical motivation for the use of coding is to reduce the required E_b/N_o for a fixed bit error rate. This reduction in E_b/N_o may, in turn be exploited to reduce the required transmitted power or to reduce the hardware costs by requiring a smaller antenna size.

Increasing demand for information exchange is characteristic of modem civilization. The transfer of information from the source to its destination has to be done in such a way that the quality of the received information should be as close as possible to the quality of the transmitted information. A typical communication system may be represented by the block diagram shown in Fig. 1.1.

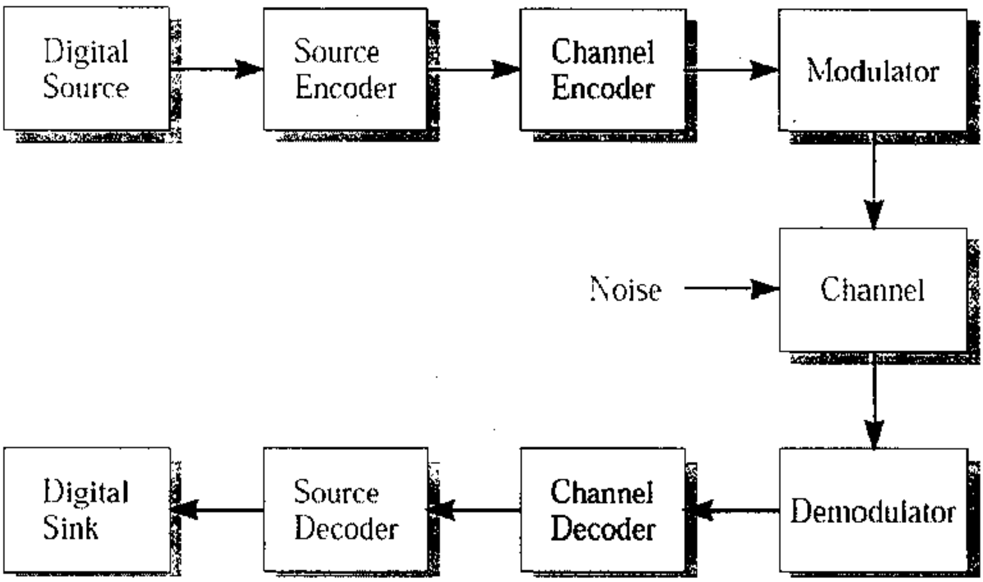

Fig. 1.1 Block diagram of a communication system

The information to be transmitted can be machine generated (e.g., images, computer data) or human generated (e.g., speech). Regardless of its source, the information must be translated into a set of signals optimized for the channel over which we want to send it. The first step is to eliminate the redundant part in order to maximize the information transmission rate. This is achieved by the source encoder block in Fig. 1.1. In order to ensure the secrecy of the information we want to transmit, an encryption scheme must be used.

The data must also be protected against perturbations introduced by the communication channel which could lead to misinterpretation of the transmitted message at the receiving end. This protection can be achieved through error control strategies: forward error correction (FEC), i.e., using error correcting codes that are able to correct errors at the receiving end, or automatic repeat request (ARQ) systems. The modulator block generates a signal suitable for the transmission channel.

In communication systems, the information is represented as a sequence of binary bits. The binary bits are then mapped (modulated) to analog signal waveforms and transmitted over a communication channel. The communication channel introduces noise and interference to corrupt the transmitted signal. At the receiver, the channel corrupted transmitted signal is mapped back to binary bits. The received binary information is an estimate of the transmitted binary information. Bit errors may result due to the transmission and the number of bit errors depends on the amount of noise and interference in the communication channel.

1.2 GOAL OF THE PROJECT

To implement the Dual Repeat punctured turbo codes on both AWGN and fading channels using iterative decoding algorithms and compare the performance.

1.3 ORGANIZATION OF THE PROJECT:

- **Chapter 2** discusses about the Error control codes
- **Chapter 3** deals about the Channel coding techniques
- **Chapter 4** discusses about the Modulation technique
- **Chapter 5** deals about the fading channels
- **Chapter 6** explains the Turbo codes
- **Chapter 7** Implements the DRPTC over fading channels
- **Chapter 8** shows simulation results and discussion
- **Chapter 9** Conclusion and Future scope.

CHAPTER 2

ERROR CONTROL CODING

2.1 INTRODUCTION

Error-control coding is a discipline under the branch of applied mathematics called Information Theory, discovered by Claude Shannon in 1948 [6]. Prior to this discovery, conventional wisdom said that channel noise prevented error-free communications. Shannon proved otherwise when he showed that channel noise limits the transmission rate, not the error probability. Shannon showed that every communications channel has a capacity, C (measured in bits per second), and as long as the transmission rate, R (also in bits per second), is less than C, it is possible to design a virtually error-free communications system using error control codes.

Shannon's contribution was to prove the existence of such codes. He did not tell us how to find them. After the publication of Shannon's famous paper [6], researchers scrambled to find codes that would produce the very small probability of error that he predicted. Progress was disappointing in the 1950s when only a few weak codes were found.

In the 1960s, the field split between the algebraists who concentrated on a class of codes called *block codes* and the probabilists, who were concerned with understanding encoding and decoding as a random process. The probabilists eventually discovered a second class of codes, called *convolutional codes*, and designed powerful decoders for them. In the 1970s, the two research paths merged and several efficient decoding algorithms were developed. With the advent of cheap microelectronics, decoders finally became practical and in 1981, the entertainment industry adopted a very powerful error control scheme for the new compact disc players.

Today, error-control coding in its many forms is used in almost every new military communications system including the Joint Tactical Information Distribution System (JTIDS) and the Jam Resistant Secure Communications (JRSC) system employed on the Defense Satellite Communications System (DSCS).

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2.2 ELEMENTS OF DIGITAL COMMUNICATION SYSTEM

ENCODER AND DECODER - The encoder adds redundant bits to the sender's bit stream to create a codeword. The decoder uses the redundant bits to detect and/or correct as many bit errors as the particular error-control code will allow.

MODULATOR AND DEMODULATOR - The modulator transforms the output of the encoder, which is digital, into a format suitable for the channel, which is usually analog (e.g., a telephone channel). The demodulator attempts to recover the correct channel symbol in the presence of noise. When the wrong symbol is selected, the decoder tries to correct any errors that result.

COMMUNICATIONS CHANNEL - The part of the communication system that introduces errors. The channel can be radio, twisted wire pair, coaxial cable, fiber optic cable, magnetic tape, optical discs, or any other noisy medium.

ERROR-CONTROL CODE – A set of codewords used with an encoder and decoder to detect errors, correct errors, or both detect and correct errors.

BIT-ERROR-RATE (BER) - The probability of bit error. This is often the figure of merit for a error-control code. We want to keep this number small, typically less than 10^{-4}. Bit-error rate is a useful indicator of system performance on an independent error channel, but it has little meaning on bursty or dependent error channels.

MESSAGE-ERROR-RATE (MER) - The probability of message error. This may be a more appropriate figure of merit because the smart operator wants all of his messages error-free and could care less about the BER.

UNDETECTED MESSAGE ERROR RATE (UMER) - The probability that the error detection decoder fails and an errored message (codeword) slips through undetected. This event happens when the error pattern introduced by the channel is such that the transmitted codeword is converted into another valid codeword. The decoder can't tell the difference and must conclude that the message is error-free. Practical error detection codes ensure that the UMER is very small, often less than 10^{-16}.

RANDOM ERRORS - Errors that occur independently. This type of error occurs on channels that are impaired solely by thermal (Gaussian) noise. Independent-error channels are also called memory less channels because knowledge of previous channel symbols adds nothing to our knowledge of the current channel symbol.

BURST ERRORS - Errors that are not independent. For example, channels with deep fades experience errors that occur in bursts. Because the fades make consecutive bits more likely to be in error, the errors are usually considered dependent rather than independent. In contrast to independent-error channels, burst-error channels have memory.

ENERGY PER BIT - The amount of energy contained in one information bit. This is not a parameter that can be measured by a meter, but it can be derived from other known parameters. Energy per bit (E_b) is important because almost all channel impairments can be overcome by increasing the energy per bit. Energy per bit (in joules) is related to transmitter power P_t (in watts), and bit rate R (in bits per second), in the following way:

$$E_b = P_t / R$$

If transmit power is fixed, the energy per bit can be increased by lowering the bit rate. Thus the reason why lower bit rates are considered more robust.

CODING GAIN - The difference (in dB) in the required signal-to-noise ratio to maintain reliable communications after coding is employed. Signal-to-noise ratio is usually given by E_b/N_0, where N_0 is the noise power spectral density measured in watts/Hertz (joules).For example, if a communications system requires a E_b/N_0 of 12 dB to maintain a BER of 10^{-5}, but after coding it requires only 9 dB to maintain the same BER, then the coding gain is 12 dB minus 9 dB = 3 dB.

CODE RATE - Consider an encoder that takes k information bits and adds r redundant bits (also called parity bits) for a total of $n = k + r$ bits per codeword. The code rate is the fraction k/n and the code is called a (n, k) error-control code. The added parity bits are a burden (i.e. overhead) to the communications system, so the system designer often chooses a code for its ability to achieve high coding gain with few parity bits.

2.3 ERROR-CONTROL CODING TECHNIQUES

2.3.1 AUTOMATIC REPEAT REQUEST (ARQ)

An error detection code by itself does not control errors, but it can be used to request repeated transmission of errored codewords until they are received error-free. This technique is called automatic repeat request, or *ARQ*. In terms of error performance, ARQ outperforms forward error correction because codewords are always delivered error-free.

This advantage does not come free – we pay for it with decreased throughput. The chief advantage of ARQ is that error detection requires much simpler decoding equipment than error correction. ARQ is also adaptive since it only re-transmits information when errors occur. On the other hand, ARQ schemes require a feedback path which may not be available. They are also prone to duping by the enemy. A pulse jammer can optimize its duty cycle to increase its chances of causing one or more errors in each codeword. Ideally (from the jammer's point of view), the jammer forces the communicator to retransmit the same codeword over and over, rendering the channel useless.

There are two types of ARQ: stop and wait ARQ and continuous ARQ.

STOP-AND-WAIT ARQ: With stop-and-wait ARQ, the transmitter sends a single codeword and waits for a positive acknowledgement (ACK) or negative acknowledgement (NAK) before sending any more codewords. The advantage of stop-and-wait ARQ is that it only requires a half duplex channel. The main disadvantage is that it wastes time waiting for ACKs, resulting in low throughput.

CONTINUOUS ARQ: Continuous ARQ requires a full duplex channel because codewords are sent continuously until a NAK is received. A NAK is handled in one of two ways: With go back-N ARQ, the transmitter retransmits the errored codeword plus all codewords that followed until the NAK was received. The parameter N is determined from the round trip channel delay. For geosynchronous satellite channels, N can be very large because of the 540 millisecond round trip delay.

The transmitter must store N codewords at a time and large values of N result in expensive memory requirements. With selective-repeat ARQ, only the errored codeword is retransmitted, thus increasing the throughput over go back-N ARQ. Both types of continuous ARQ offer greater throughput efficiency than stop-and-wait ARQ at the cost of greater memory requirements.

2.3.2 FORWARD ERROR CORRECTION (FEC)

Forward error correction is appropriate for applications where the user must get the message right the first time. The one-way or broadcast channel is one example. Today's error correction codes fall into two categories: block codes and convolutional codes.

2.4 LINEAR BLOCK CODES

In coding theory, **block codes** are one of the channel codes, which enable reliable transmission of digital data over unreliable communication channels subject to channel noise.

A block code transforms a message m consisting of a sequence of information symbols over an alphabet Σ into a fixed-length sequence c of n encoding symbols, called a code word. In a **linear block code**, each input message has a fixed length of $k < n$ input symbols. The redundancy added to a message by transforming it into a larger code word enables a receiver to detect and correct errors in a transmitted code word, and – using a suitable decoding algorithm – to recover the original message.

The redundancy is described in terms of its information rate, or more simply – for a linear block code – in terms of its code rate, k/n. The error correction performance of a block code is described by the minimum Hamming distance d between each pair of code words, and is called the distance of the code.

2.5 CONVOLUTIONAL CODES

With convolutional codes, the incoming bit stream is applied to a K-bit long shift register. For each shift of the shift register, b new bits are inserted and n code bits are delivered, so the code rate is b/n. The power of a convolutional code is a function of its constraint length, K. Large constraint length codes tend to be more powerful. Unfortunately, with large constraint length comes greater decoder complexity.

There are several effective decoding algorithms for convolutional codes, but the most popular is the Viterbi algorithm, discovered by Andrew Viterbi in 1967. Viterbi decoders are now available on single integrated circuits (VLSI) from several manufacturers. Viterbi decoders are impractical for long constraint length codes because decoding complexity increases rapidly with constraint length. For long constraint length codes ($K > 9$), a second decoding algorithm called sequential decoding is often used.

A third decoding technique, feedback decoding, is effective on burst-error channels, but is inferior on random error channels. In general, convolutional codes provide higher coding gain than block codes for the same level of encoder/decoder complexity.

One drawback of the codes we have looked at so far is that they all require bandwidth expansion to accommodate the added parity bits if the user wishes to maintain the original unencoded information rate. In 1976, Gottfried Ungerboeck discovered a class of codes that integrates the encoding and modulation functions and does not require bandwidth expansion [4]. These codes are called Ungerboeck codes or trellis coded modulation (TCM).

2.6 TURBO CODES

Turbo codes (originally in French Turbocodes) 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.

CHAPTER 3

CHANNEL CODING

3.1 FUNDAMENTALS OF CHANNEL CODING

Channel coding is often used in digital communication systems to protect the digital information from noise and interference and reduce the number of bit errors. Channel coding is mostly accomplished by selectively introducing redundant bits into the transmitted information stream. These additional bits will allow detection and correction of bit errors in the received data stream and provide more reliable information transmission. The cost of using channel coding to protect the information is a reduction in data rate or an expansion in bandwidth.

3.2 WHY CHANNEL CODING?

- Severe transmission conditions in terrestrial mobile radio communication due to multipath fading.
- Very low signal-to-noise ratio for Satellite communications due to limited transmits power in downlink.
- Compressed data (e.g., audio and video signals) is very sensitive to transmission errors and should be protected from them.
- Channel coding is applied to ensure adequate transmission quality (bit or frame error rate).

3.3 PRINCIPLES OF CHANNEL CODING

- Adding redundancy to data stream enables receiver to detect or even correct transmission errors.
- Commonly used codes are linear block codes (e.g., Hamming Codes, BCH and Reed-Solomon-Codes) and Convolutional codes.

Channel coding protects digital data from errors by selectively introducing redundancies in the transmitted data. Channel codes that are used to detect errors are called *error detection codes*, while codes that can be detect and correct errors are called *error correction codes.*

In 1948, Shannon demonstrated that by proper encoding of the information, errors induced by a noisy channel can be reduced to any desired level without sacrificing the rate of information transfer, Shannon's channel capacity formula is applicable to the AWGN channel and is given by,

$$C = B \log_2 (1+S/N) \quad \ldots\ldots\ldots\ldots\ldots\ldots (1)$$

Where C is the channel capacity (bits per second), B is the transmission bandwidth (Hz), P is the received power (W), and N is the single sided noise power density (W/Hz).The received power at a receiver is given as

$$P = E_b R_b \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (2)$$

Where E_b is the average bit energy, and R_b is the transmission rate. Equation (1) can be normalized by the transmission bandwidth and is given by

$$C/B = \log_2 (1 + E_b R_b/N_o B) \ldots\ldots\ldots\ldots (3)$$

Where C/B denotes bandwidth efficiency.

The basic purpose of error detection and error correction techniques is to introduce redundancies in the data to improve wireless link performance. The introduction of redundant bits increases the raw data rate used in the link, and hence it increases the bandwidth requirement for a fixed source data rate. This reduces the Bandwidth efficiency of the link in high SNR conditions, but provides excellent BER performance at low SNR values.

3.4ADVANTAGES OF CHANNEL CODING

It is well known that the use of orthogonal signaling allows the probability of error to become arbitrarily small by expanding the signal set, i.e., by making the number of waveforms M tends to infinity, provided that the SNR per bit exceeds the Shannon limit of $SNR \geq -1.6$ dB.

The wide band signals could be used to achieve error free communication, as long as sufficient SNR exists, and this is partly why CDMA is adopted for 3G. Error control coding waveforms , on the other hand, have bandwidth expansion factors that grow only linearly with code block length. Error correction coding thus offers advantages in bandwidth limited application, and also provides link protection in power limited applications.

A channel coder operates on digital message by encoding the source information into a code sequence for transmission through the channel.

3.5 WHAT CODING CAN AND CANNOT DO

The traditional role for error-control coding was to make a troublesome channel acceptable by lowering the frequency of error events. The error events could be bit errors, message errors, or undetected errors. Coding role has expanded tremendously and today coding can do the following:

- **REDUCE THE OCCURRENCE OF UNDETECTED ERRORS**. This was one of the first uses of error-control coding. Today's error detection codes are so effective that the occurrence of undetected errors is, for all practical purposes, eliminated.
- **REDUCE THE COST OF COMMUNICATIONS SYSTEMS**. Transmitter power is expensive, especially on satellite transponders. Coding can reduce the satellite's power needs because messages received at close to the thermal noise level can still be recovered correctly.
- **OVERCOME JAMMING**. Error-control coding is one of the most effective techniques for reducing the effects of the enemy's jamming. In the presence of pulse jamming, for example, coding can achieve coding gains of over 35 dB [8].
- **ELIMINATE INTERFERENCE**. As the electromagnetic spectrum becomes more crowded with man-made signals, error-control coding will mitigate the effects of unintentional interference. Despite all the new uses of error-control coding, there are limits to what coding can do. On the Gaussian noise channel, for example, Shannon's capacity formula sets a lower limit on the signal-to-noise ratio that we must achieve to maintain reliable communications.

Shannon's lower limit depends on whether the channel is power-limited or bandwidth-limited. The deep space channel is an example of a power-limited channel because bandwidth is an abundant resource compared to transmitter power. Telephone channels, on the other hand, are considered bandwidth-limited because the telephone company adheres to a strict 3.1 kHz channel bandwidth.

For strictly power-limited (unlimited bandwidth) channels, Shannon's lower bound on E_b/N_0 is 0.69, or –1.6 dB. In other words, we must maintain an E_b/N_0 of at least -1.6 dB to ensure reliable communications, no matter how powerful an error-control code we use.

For bandwidth-limited channels with Gaussian noise, Shannon's capacity formula can be written as the following [3]:

$$E_b/N_0 \geq 2^{r-1} / r \quad \text{.........................} \quad (4)$$

Where r is the spectral bit rate in bits/s/Hz. For example, consider a bandwidth-limited channel operating with uncoded quadrature phase shift keying (a common modulation technique with 2 bits/symbol and a maximum spectral bit rate of $r = 2$ bit/s/Hz) and a required BER of 10^{-5}. We know that without coding, this communications system requires an E_b/N_0 of 9.6 dB.

Shannon's formula above says that to maintain reliable communications at an arbitrarily low BER, we must maintain (for $r = 2$ bits/s/Hz) an E_b/N_0 of at least 1.5 (1.8 dB). Therefore, if we need to lower the required E_b/N_0 by more than 7.8 dB, coding can't do it. We must resort to other measures, like increasing transmitter power. In practice, the situation is worse because we have no practical code that achieves Shannon's lower bound. A more realistic coding gain for this example is 3 dB rather than 7.8 dB.

Another limitation to the performance of error-control codes is the modulation technique of the communication system. Coding must go hand-in-hand with the choice of modulation technique for the channel. Even the most powerful codes cannot overcome the consequences of a poor modulation choice.

CHAPTER 4

FADING CHANNELS

4.1 INTRODUCTION

Due to multipath propagation the received signal may contain multiple copies of the original signal which may have slight variation in phase, amplitude or frequency. This is termed as Fading. Small scale fading or simple fading is used to describe the rapid fluctuation of the amplitude, phase, or multipath delays of a radio over short period of time or travel distance, so that large scale path loss effect may be ignored.

Fading is caused by interference between two or more versions of the transmitted signal which arrive at the receiver at slightly different times. These waves are called multipath waves. These waves combine at the receiver to give a resultant signal which can vary widely in amplitude and phases, depending on the distribution of the intensity and relative propagation time of the wave and the band width of the transmitted signal.

For most practical channels, where signal propagation takes place in the atmosphere and near the ground, the free-space propagation model is inadequate to describe the channel and predict system performance. In a wireless mobile communication system, a signal can travel from transmitter to receiver over multiple reflective paths; this phenomenon is referred to as *multipath* propagation. The effect can cause fluctuations in the received signal's amplitude, phase, and angle of arrival, giving rise to the terminology *multipath fading*. Another name, *scintillation*, which originated in radio astronomy, is used to describe the multipath fading caused by physical changes in the propagating medium, such as variations in the density of ions in the ionospheric layers that reflect high-frequency (HF) radio signals. Both names, fading and scintillation refer to a signal's random fluctuations or fading due to multipath propagation.

The main difference is that scintillation involves mechanisms (e.g., ions) that are much smaller than a wavelength. The end-to-end modeling and design of systems that mitigate the effects of fading are usually more challenging than those whose sole source of performance degradation is AWGN.

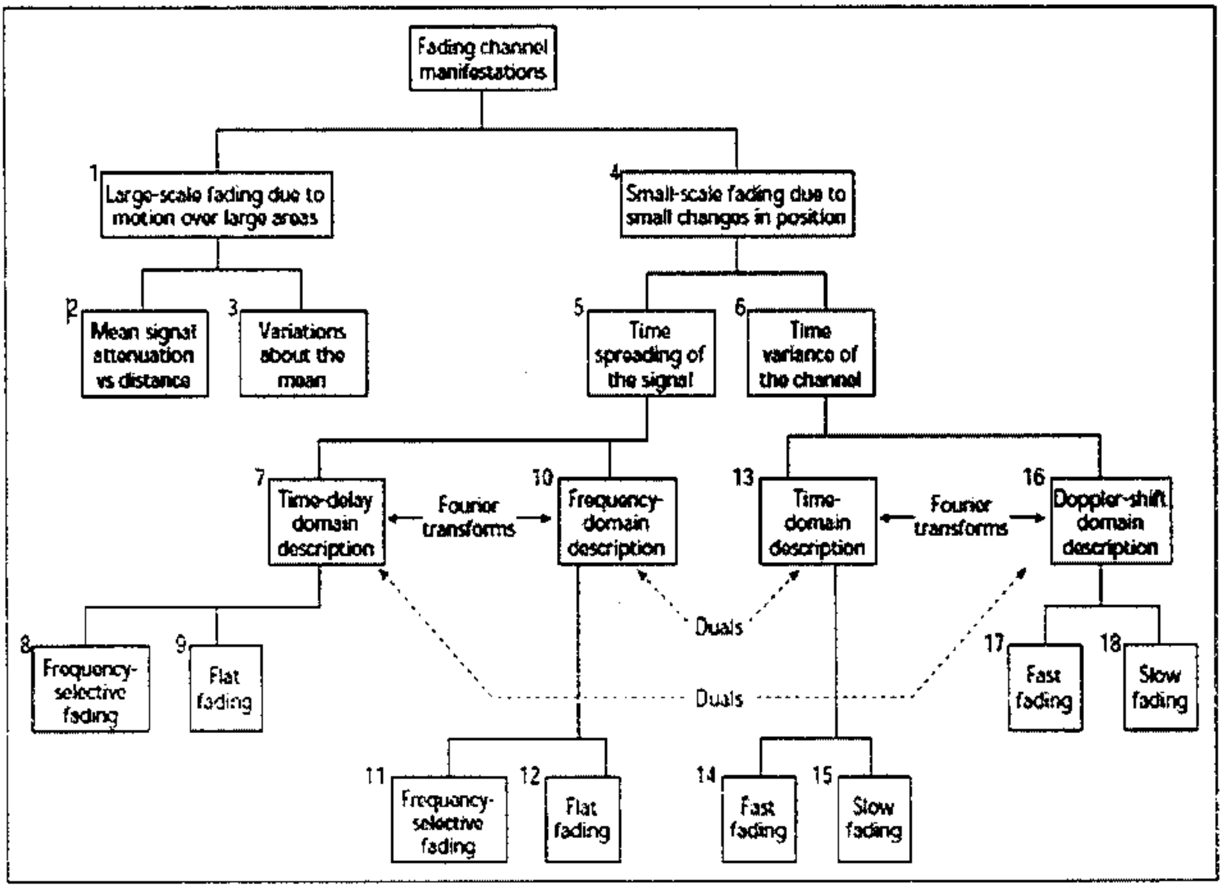

Fig. 4.1 Fading channels

4.2 DIFFERENT FADING CHANNELS

Based on multipath time delay spread

- Flat Fading
- Frequency Selective Fading

Based on Doppler spread

- Fast Fading
- Slow fading

4.2.1 FLAT FADING

If the Channel has a constant gain and linear phase over a band width which is smaller than the bandwidth of transmitted signal, then the received signal will undergo Flat Fading. Flat fading channels are also known as Amplitude varying channels and are sometimes referred as narrow band channels, since bandwidth of the applied signal is narrow as compared to the channel flat fading bandwidth.

4.2.2 FREQUENCY SELECTIVE FADING

Frequency selective fading channels are much more difficult to model than flat fading channels, since each multipath signal must be considered to be a linear filter. It is for this reason, the wideband multipath measurements are made and models are developed from the measurements. Here, the spectrum S (f) of the transmitted signal has a bandwidth which is greater than the coherence bandwidth B_c of the channel. When the channels are viewed in the frequency domain, the channel becomes frequency selective. Frequency selective fading is caused by multipath delays which approach or exceed the symbol period of the transmitted signal. Frequency selective fading channels are also known as wideband channels.

4.2.3 FAST FADING

Fast fading deals with the rate of change of channel due to motion. It is the channel in which amplitude of the delta function varies faster than the rate of change of the transmitted baseband signal. The amplitudes, phases & time delays of any one of the multipath components vary faster than the rate of change of the transmitted signal. In practice, Fast fading only occurs for very low data rates.

4.2.4 SLOW FADING

In a slow fading channel the channel impulse response changes at a rate much slower than the transmitted baseband signal S (t). In this case, the channel may be assumed to be static over one or several reciprocal bandwidth intervals. In frequency domain, this implies that the Doppler spread of the channel is much less than the bandwidth of the baseband signals.

4.3 FADING EFFECTS

Multipath fading introduces:

- Delay spread due to power delay
- Doppler-spread due to time varying nature of the channel caused the relative motion between the mobile and the base station.
- Fade amplitude statistics.

Time dispersion due to multipath causes transmitted signal to undergo either flat or frequency flat fading. If the radio channel has constant gain and linear phase response over a bandwidth which is greater than the bandwidth of transmitted signal, than the received signal will undergo Flat fading. Here spectral characteristics of transmitted signals are preserved at the receiver. However, the strength of the received signal changes with time, due to fluctuation in gain of channel caused by multipath.

The Channel has a constant gain and linear phase over a band width which is smaller than the bandwidth of transmitted signal, then the received signal will undergo Flat Fading. It is due to time dispersion of transmitted symbols within the channel.

4.4 TIME VARIANT MULTIPATH FADING CHANNEL

Fundamentally, communication channels are time varying, multipath fading channels. In a radio communication system, there are many paths for a signal to travel from a transmitter to a receiver. Sometimes there is a direct path where the signal travels without being obstructed. In most cases, components of the signal are reflected by the ground and objects between the transmitter and the receiver such as buildings, vehicles, and hills or refracted by different atmospheric layers.

These components travel in different paths and merge at the receiver. Each path has a different physical length. Thus, signals on each path suffer different transmission delays due to the finite propagation velocity. The superposition of these signals at the receiver results in destructive of constructive interference, depending on the relative delays involved. The fact that the environment changes as time passes leads to signal variation. This is called *time variant*. Signals are also influenced by the motion of a terminal.

A short distance movement can cause an apparent change in the propagation paths and in turn the strength of the received signals.

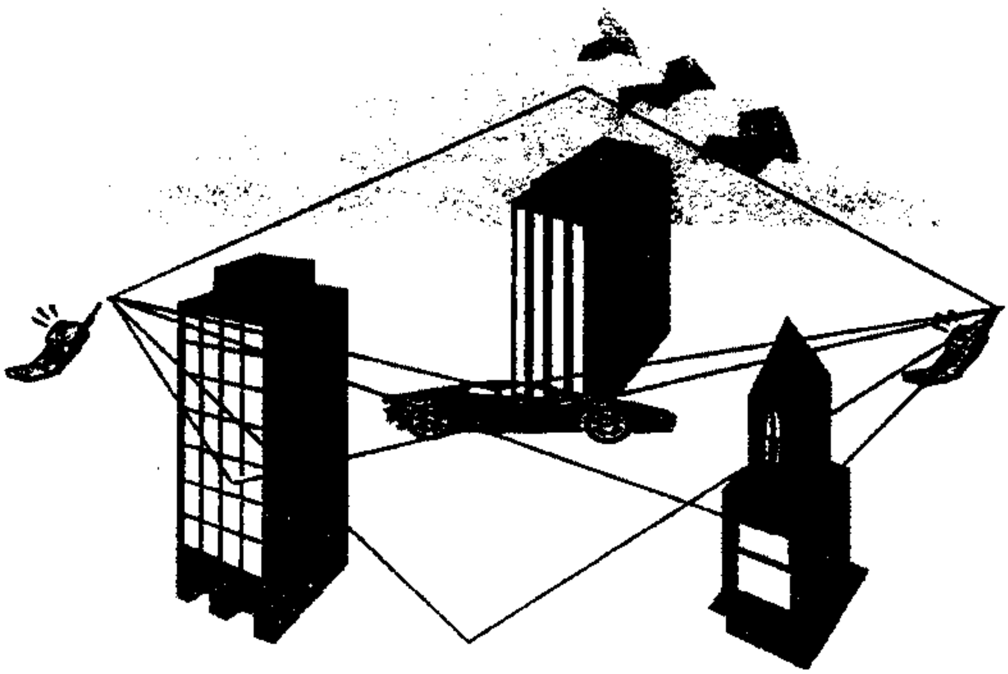

Fig. 4.2 Multipath fading channel

4.5 RAYLEIGH FADING

Rayleigh fading models assume that the magnitude of a signal that has passed through such a communication channel will vary randomly, or fade, according to a Rayleigh Distribution — the radial component of the sum of two uncorrelated Gaussian random variables

Rayleigh fading is viewed as a reasonable model for tropospheric and ionospheric signal propagation as well as the effect of heavily built-up urban environments on radio signals. Rayleigh fading is most applicable when there is no dominant propagation along a line of sight between the transmitter and receiver. If there is a dominant line of sight, Rician fading may be more applicable.

Probability density function:

$$p_R(r) = \frac{2r}{\Omega} e^{-r^2/\Omega},\ r \geq 0 \quad \text{............. (5)}$$

Where $\Omega = E(R^2)$.

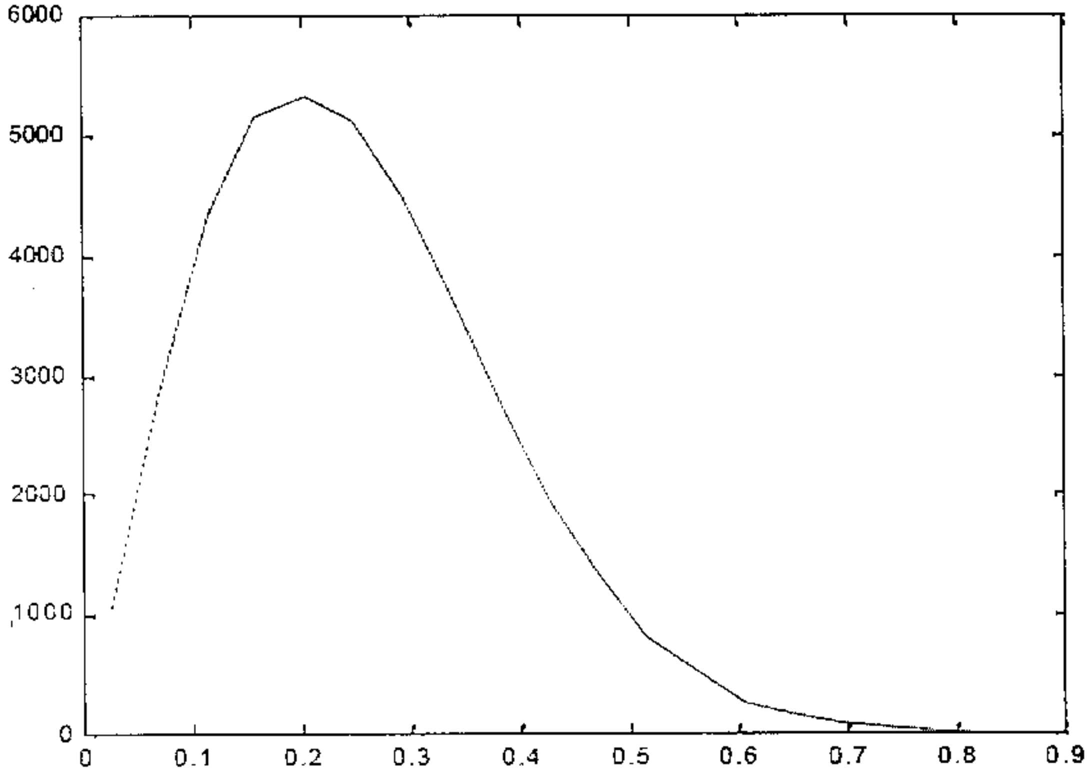

Fig. 4.3 Rayleigh distribution

PROPERTIES

- The Rayleigh distribution lends itself to analysis, and the key features that affect the performance of a wireless network have analytic expressions
- If a channel is not changing with time, clearly it does not fade and instead remains at some particular level. Separate instances of the channel in this case will be

uncorrelated with one another owing to the assumption that each of the scattered components fades independently. Once relative motion is introduced between any of the transmitter, receiver and scatterers, the fading becomes correlated and varying in time.

4.6 RICIAN FADING

Rician fading is a stochastic model for radio propagation anomaly caused by partial cancellation of a radio signal by itself — the signal arrives at the receiver by two different paths (hence exhibiting multipath interference), and at least one of the paths is changing (lengthening or shortening). Rician fading occurs when one of the paths, typically a line of sight signal, is much stronger than the others. In Rician fading, the amplitude gain is characterized by a Rician distribution.

Rayleigh fading is the specialized model for stochastic fading when there is no line of sight signal, and is sometimes considered as a special case of the more generalized concept of Rician fading. In Rayleigh fading, the amplitude gain is characterized by a Rayleigh distribution.

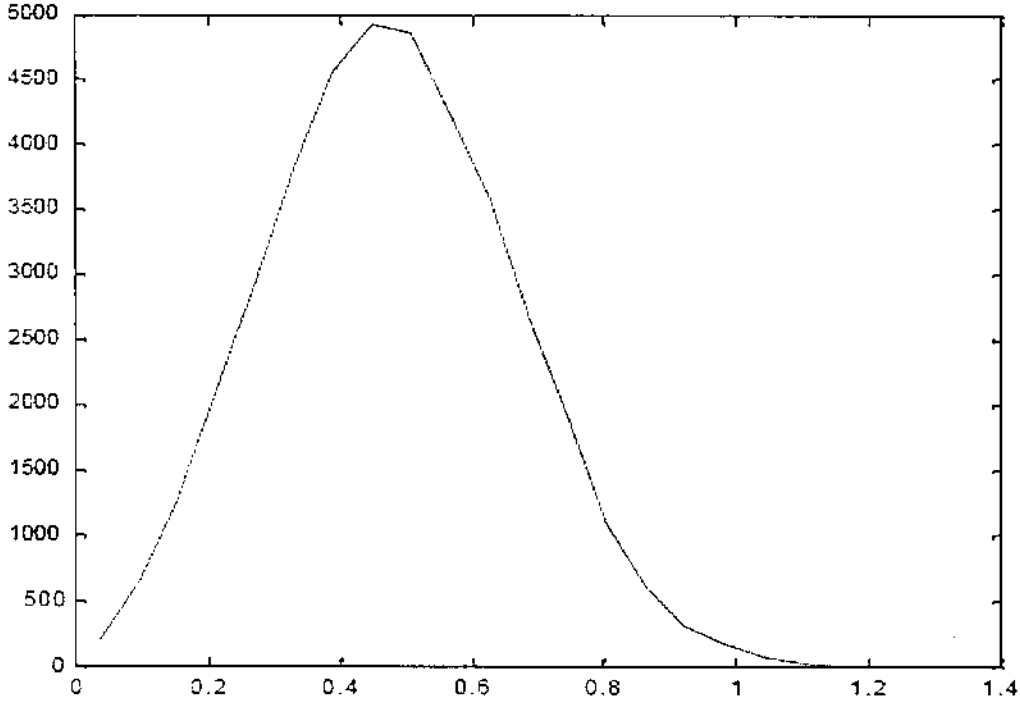

Fig. 4.4 Ricean distribution

CHAPTER 5

DIGITAL MODULATION TECHNIQUE

5.1 WHY DIGITAL MODULATION?

The move to digital modulation provides more information capacity, compatibility with digital data services, higher data security, better quality communications, and quicker system availability. Developers of communications systems face these constraints:

- Available bandwidth
- Permissible power
- Inherent noise level of the system

The RF spectrum must be shared, yet every day there are more users for that spectrum as demand for communications services increases. Digital modulation schemes have greater capacity to convey large amounts of information than analog modulation schemes.

5.2 TRADING OFF SIMPLICITY AND BANDWIDTH

There is a fundamental tradeoff in communication systems. Simple hardware can be used in transmitters and receivers to communicate information. However, this uses a lot of spectrum which limits the number of users. Alternatively, more complex transmitters and receivers can be used to transmit the same information over less bandwidth. The transition to more and more spectrally efficient transmission techniques requires more and more complex hardware. Complex hardware is difficult to design, test, and build. This tradeoff exists whether communication is over air or wire, analog or digital.

5.3 INDUSTRY TRENDS

Over the past few years a major transition has occurred from simple analog Amplitude Modulation (AM) and Frequency/Phase Modulation (FM/PM) to new digital modulation techniques. Examples of digital modulation include

- QPSK (Quadrature Phase Shift Keying)
- FSK (Frequency Shift Keying)
- MSK (Minimum Shift Keying)
- QAM (Quadrature Amplitude Modulation)

Another layer of complexity in many new systems is multiplexing. Two principal types of multiplexing (or "multiple accesses") are TDMA (Time Division Multiple Access) and CDMA (Code Division Multiple Access). These are two different ways to add diversity to signals allowing different signals to be separated from one another.

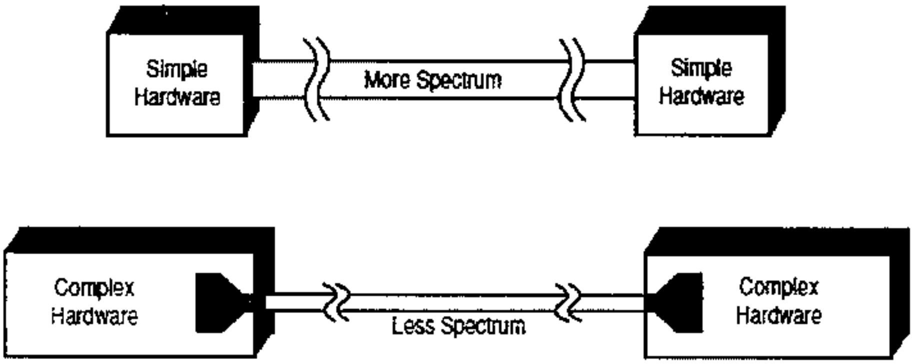

Fig. 5.1The fundamental Tradeoff

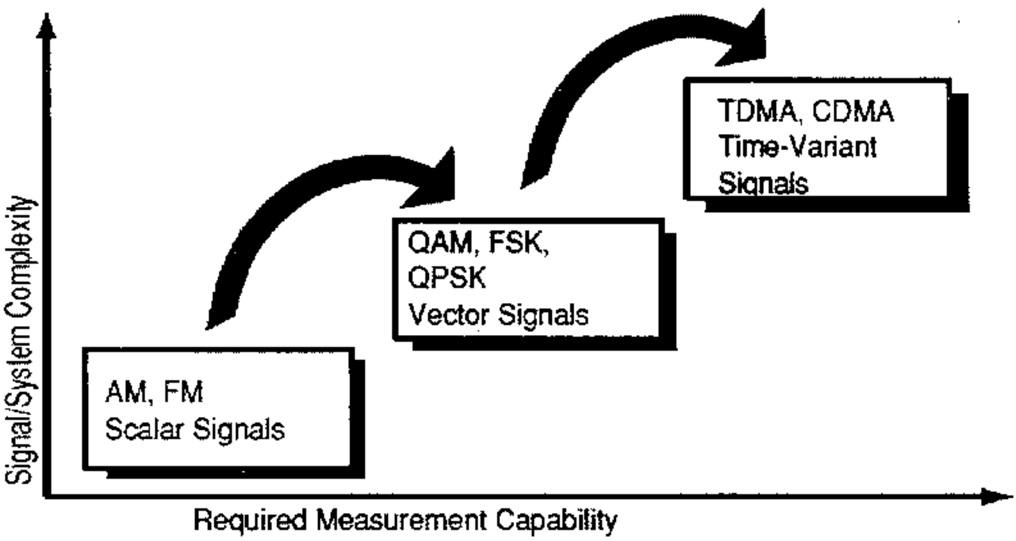

Fig. 5.2 Trends in the Industry

5.4 I/Q FORMATS

In digital communications, modulation is often expressed in terms of *I* and *Q*. This is a rectangular representation of the polar diagram. On a polar diagram, the *I* axis lies on the zero degree phase reference, and the *Q* axis is rotated by 90 degrees. The signal vector's projection onto the *I* axis is its "I" component and the projection onto the *Q* axis is its "Q" component.

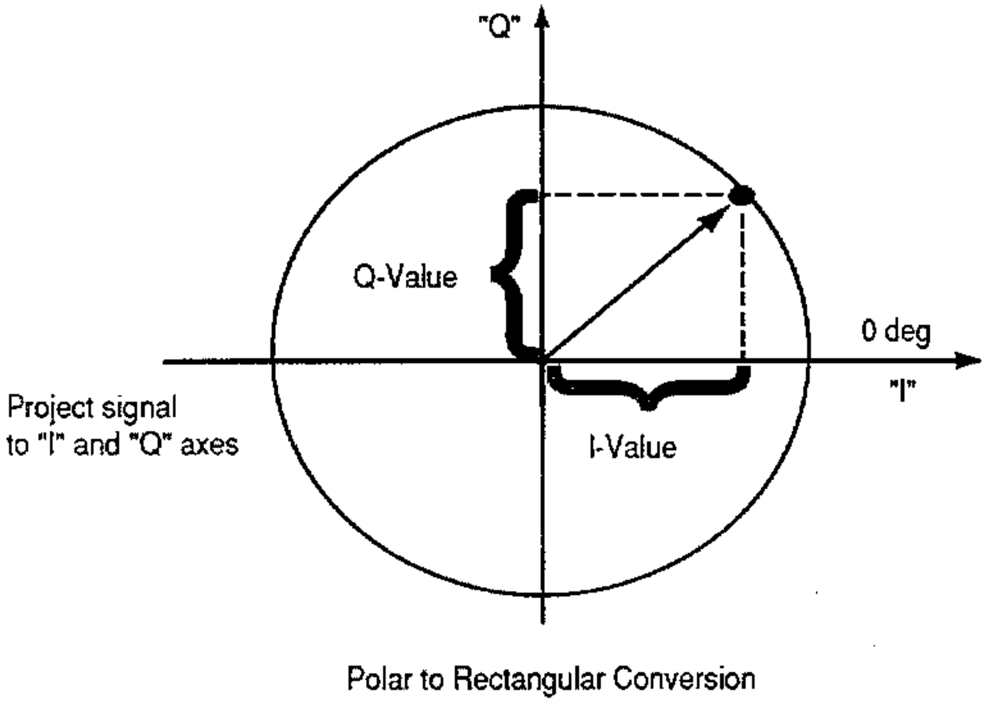

Fig. 5.3 "*I-Q*" Format

Digital modulation is easy to accomplish with *I/Q* modulators. Most digital modulation maps the data to a number of discrete points on the *I/Q* plane. These are known as constellation points. As the signal moves from one point to another, simultaneous amplitude and phase modulation usually results. To accomplish this with an amplitude modulator and a phase modulator is difficult and complex. It is also impossible with a conventional phase modulator.

The signal may, in principle, circle the origin in one direction forever, necessitating infinite phase shifting capability. Alternatively, simultaneous AM and Phase Modulation is easy with an *I/Q* modulator. The *I* and *Q* control signals are bounded, but infinite phase wrap is possible by properly phasing the *I* and *Q* signals.

5.5 PHASE SHIFT KEYING

One of the simplest forms of digital modulation is binary or Bi-Phase Shift Keying (BPSK). One application where this is used is for deep space telemetry. The phase of a constant amplitude carrier signal moves between zero and 180 degrees. On an *I* and *Q* diagram, the *I* state has two different values. There are two possible locations in the state diagram, so a binary one or zero can be sent. The symbol rate is one bit per symbol.

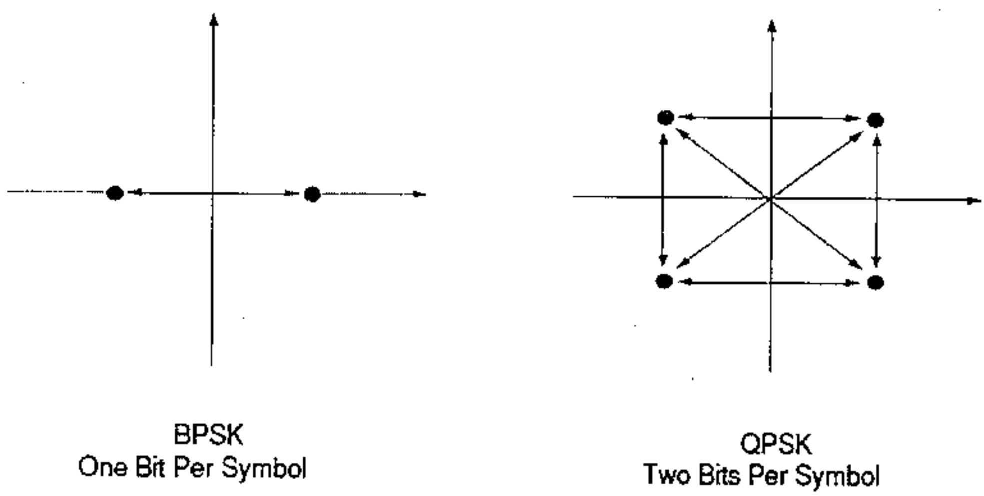

Fig. 5.4 Phase Shift Keying

A more common type of phase modulation is Quadrature Phase Shift Keying (QPSK). It is used extensively in applications including CDMA (Code Division Multiple Access) cellular service, wireless local loop, Iridium (a voice/data satellite system) and DVB-S (Digital Video Broadcasting — Satellite). Quadrature means that the signal shifts between phase states which are separated by 90 degrees.

The signal shifts in increments of 90 degrees from 45 to 135, –45, or –135 degrees. These points are chosen as they can be easily implemented using an *I/Q* modulator. Only two *I* values and two *Q* values are needed and this gives two bits per symbol. There are four states because $2^2 = 4$. It is therefore a more bandwidth-efficient type of modulation than BPSK, potentially twice as efficient.

5.6 BINARY PSK

Binary data are represented by two signals with different phases in BPSK. Typically these two phases are 0 and *T*, the signals are

$S_1(t) = A\cos 2\pi f_c t, \quad 0 \leq t \leq T, \quad \text{for } 1$

$S_2(t) = -A\cos 2\pi f_c t, \quad 0 \leq t \leq T, \quad \text{for } 0$

These signals are called antipodal. The reason that they are chosen is that they have a correlation coefficient of -1, which leads to the minimum error probability for the same E_b/N_0, as we will see shortly. These two signals have the same frequency and energy.

All PSK signals can be graphically represented by a ***signal constellation*** in a two-dimensional coordinate system with

$\Phi_1(t) = \sqrt{(2/T)}\, 2\cos 2\pi f_c t, \quad 0 \leq t \leq T$

$\Phi_2(t) = -\sqrt{(2/T)}\, 2\sin 2\pi f_c t, \quad 0 \leq t \leq T$

As it's horizontal and vertical axis, respectively.

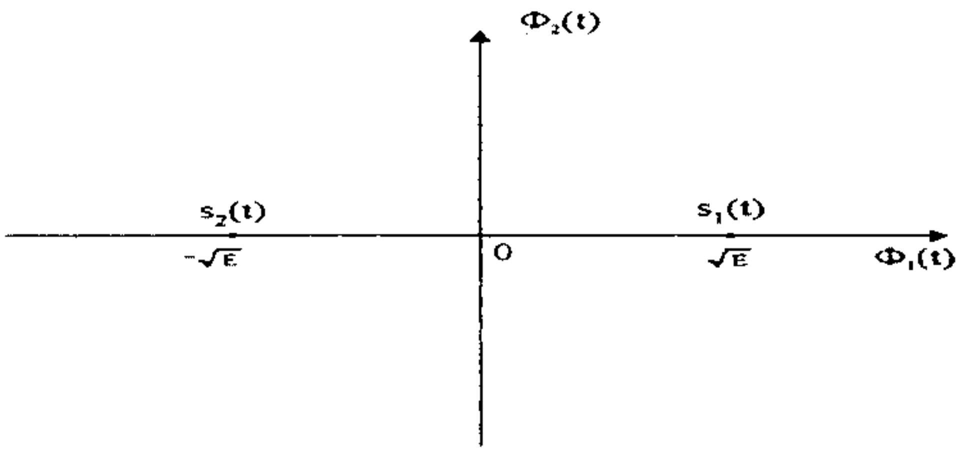

Fig. 5.5 BPSK signal constellation

CHAPTER 6

TURBO CODES

6.1 INTRODUCTION TO TURBO CODES

Turbo codes which are one of the most powerful error correcting codes presently available, were introduced in the early 90's by Berrou, Glavieux and Thitimajshima [1]. Turbo codes were reported to yield extremely impressive results. Berrou *et al.* showed that turbo codes could approach Shannon limit [1] to about O.7dB with reasonable complexity at BER of 10^{-5}. They have become a popular area of communications research as well as being implemented into standardized systems such as Third Generation (3G) systems.

A conventional turbo encoder consists of two parallel concatenated recursive systematic convolutional (RSC) encoders with their inputs separated by an interleaver or permuter. The turbo decoder comprises two Maximum a Priori (MAP) decoders connected in series through interleavers and deinterleavers. The output of the second decoder is fed back to the input of the first decoder. In the remainder of this section, we investigate turbo codes in detail by describing the structure of turbo encoders and decoders and by showing their performance with various parameters.

6.2 TURBO CODE ENCODER

The turbo code is also called parallel concatenated coding. This is because both encoders separated by an interleaver act on the same set of input bits, unlike a serial concatenated scheme in which the output of one encoder is encoded by the other encoder. For this reason, turbo codes are also referred to as parallel concatenated convolutional codes (PCCC). Fig. 6.1 shows a block diagram of a standard turbo code encoder.

Both convolutional encoders are recursive systematic and are assumed to be identical. The choice of recursive convolutional encoders over non-recursive systematic encoders will be discussed later. In the remainder of this sub-section, we describe the building blocks that form a standard turbo code encoder: constituent encoders, interleaver and puncturer. In Figure 2.6, x_k^s, x_{1k}^p, x_{2k}^p represent the systematic, first parity and second parity bits respectively.

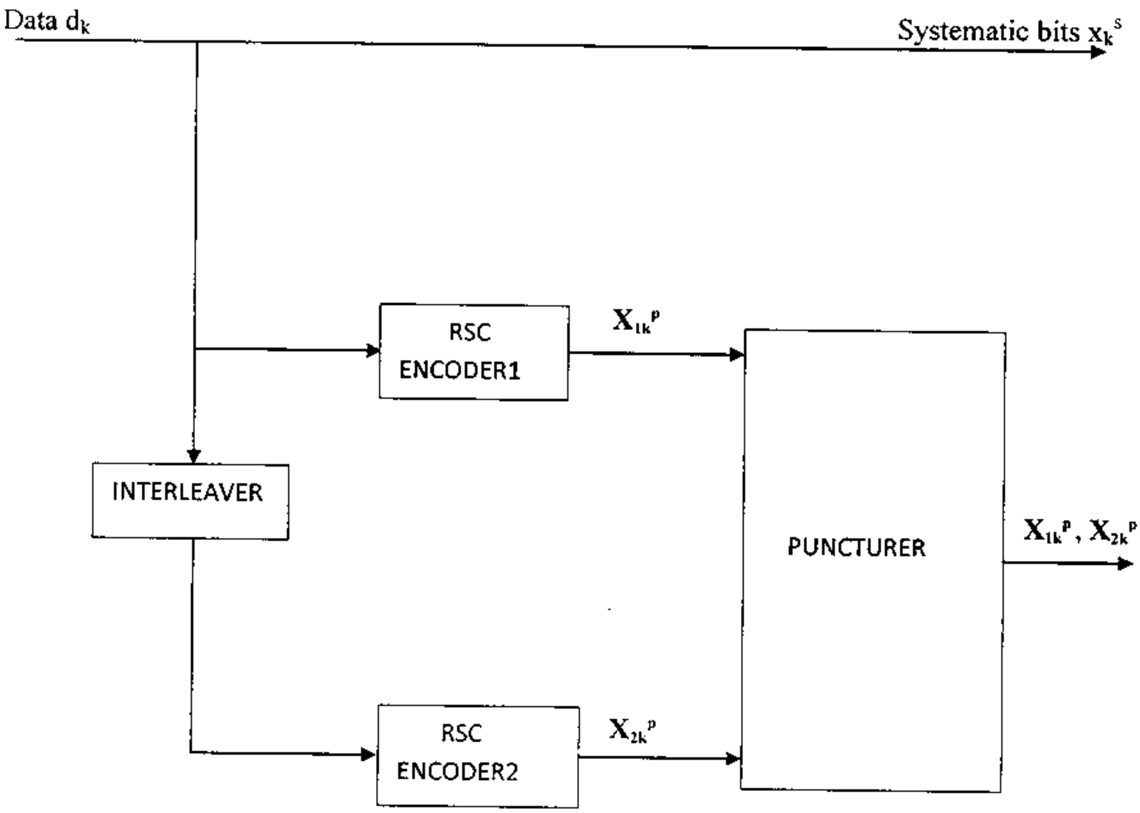

Fig. 6.1 Standard Turbo code encoder

6.2.1 RECURSIVE SYSTEMATIC ENCODERS

Most convolutional codes have been used in their non-recursive or feed-forward form and are represented by a set of generator polynomials $G_{NR} = (g_1(D), g_2(D))_g$ Turbo codes use recursive systematic convolutional (RSC) encoders which are convolutional codes with feedback ("recursive or feedback") and in which the uncoded data sequence appears in the transmitted data sequence ("systematic"). A feedback or recursive systematic encoder can easily be obtained from a feed-forward convolutional code without changing its distance property, in other words feed-forward and feedback convolutional encoders generate the same set of encoded sequence. This is done by feeding back the generator polynomials $g_1(D)$, $G_R = (1, g2(D)/g_1(D))_g$ where $g_1(D)$ is the feedback polynomial and $g_2(D)$ is the feed-forward polynomial, both in octal notation.

The term $g_2(D)/ g_1(D)$ is not an actual division but rather a notation allowing the feed-forward and feedback taps of the code to be distinctly deformed. Fig. 6.2 gives a description of a recursive systematic encoder, where $g_1(D) = 1+ D^2$, $g_2(D) = 1+ D + D^2$ (D is the delay operator and D^m represent a delay of m symbol times).

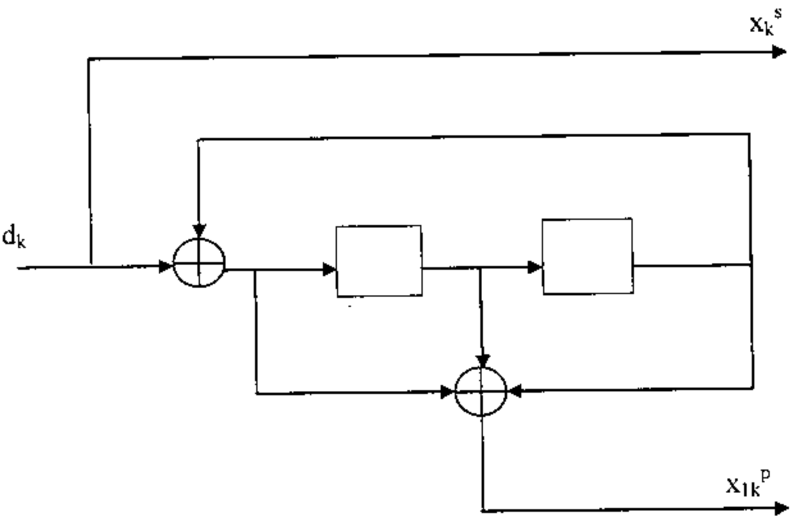

Fig. 6.2 Recursive systematic convolutional encoder

Unlike feed-forward convolutional encoders, recursive systematic encoders prevent the encoders from going back to the all-zero state by zero symbols. For a data sequence $d_k = (0, \ldots ,0,1,0, \ldots ,0)$ and assuming that the RSC codes terminate in the all-zero state, the data sequence d_k will end up with at least two non-zero bits or symbols.

It is probable that one of the RSC codes have high inputs, since the data sequence d_k is permuted by an interleaver before going through the second RSC encoder. This is not the case for a non-RSC encoder, because a data sequence $d_k = (0, \ldots ,0,1,0, \ldots ,0)$ and its permuted version (despite the permutation) will always generate a low output codeword.

It is known that codes with low minimum Hamming distance have poor error correction capability. This is why feedback convolutional or RSC encoders are suitable in turbo codes.

6.2.2 THE INTERLEAVER

An interleaver is a device for permuting or reordering bits or symbols in an information sequence. The primary role of the interleaver in communications is to protect data transmission against burst errors. This is done by spreading the errors apart in a systematic pattern. Another role of the interleaver is to decrease the number of codewords with small Hamming distance by breaking low weight input sequences that can generate low weight outputs. The final function of the interleaver is to have the inputs to the two decoders uncorrelated, so that an iterative decoding algorithm based on information exchange between the two decoders can be employed.

There is a high probability that after correction of some errors in the first component decoder, some of the remaining errors can be corrected in the second decoder provided that the inputs sequence to both encoder are uncorrelated. In turbo coding, structured codes do not perform as good as random codes. Codes with structure can be obtained not only with block interleaver which is a rectangular matrix that permutes the input sequence in a systematic fashion, but also with random interleavers. But some structured codes can permit decoding with reasonable complexity. Berrou *et al.* introduced a pseudo-random interleaver to solve this coding dilemma. Also the size of the interleaver or the number of elements to permute N should be large, since large block-length random codes approach the capacity limit.

6.2.3 THE PUNCTURING UNIT

The code obtained from a turbo code encoder shown in Fig. 6.1 without the puncture is a $(3N, N)$ linear block code. Puncturing is the process of periodically deleting some selected bits from the codeword. The main role of the puncture is to increase or vary the rate of the overall system. In turbo coding, puncturing is only applied on both parity bits or symbols of both encoders and can lead to rates of 1/2, 3/4... Depending on the puncturing matrix is used. Rates 1/2 and 3/4 can be obtained from the unpunctured $(3N, N)$ code by using the respective puncture matrices:

$$P_{1/2}=\begin{bmatrix} 1 & 1 \\ 1 & 0 \\ 0 & 1 \end{bmatrix} \quad \text{and} \quad P_{3/4}=\begin{bmatrix} 1 & 1 & 1 & 1 & 1 & 1 \\ 1 & 0 & 0 & 0 & 0 & 0 \\ 0 & 0 & 0 & 1 & 0 & 0 \end{bmatrix}$$

6.2.4 TRELLIS TERMINATION

The term *trellis termination* refers to a process in which the encoder is driven back to the all-zero state. As discussed above, in turbo coding it is impossible to terminate the trellis of any of the encoders by transmitting the all-zeroes tail bits. This is because the encoders are recursive. It is very difficult to terminate both encoders simultaneously with the use a pseudo-random interleaver, because the terminating sequence of the first encoder is interleaved and may not by itself temllnate the second encoder. Interleaver designs have been devised which can drive both encoders back to all-zero state. It was shown in that the performance degradation produced by terminating both encoders is negligible for large interleaver size. We assume in the sequel of the dissertation that only the first encoder is terminated leaving the second encoder in an unknown state.

6.3 TURBO CODE DECODER

A block diagram of the turbo code decoder structure is shown in Fig. 6.3. The two constituent decoders are *soft-in soft-out* (SISO) and are linked by interleavers in a manner reminiscent of the turbo code encoder structure. Each of the SISO decoders takes three inputs:

- The corrupted version of the transmitted systematic symbols y_k^s from the output of the demodulator.
- The faded version of the transmitted parities associated with the SISO, that is, y_{1k}^p for decoder 1 and y_{2k}^p for decoder 2, from the output of the demodulator.
- The information referred to as *a-priori*, from the other decoder about the likely values of the bits concerned. The *a-priori* information is also termed intrinsic information.

In other words, the component decoder I is used to decode the sequences from the encoder 1 and the same applies for decoder 2. Each decoder provides two soft outputs:

- An extrinsic information which is to be exchanged between the two SISO decoders. It is the additional information provided by the decoder based on the *a-priori* information and the received sequence.
- A posteriori information which is the information given by the decoder considering all variable sources of information about a particular bit. We assume BPSK modulation is used in the following discussion. The outputs of the two decoders are expressed in terms of the Log

Likelihood Ratios (LLRs), which are the logarithm of the ratio of the two probabilities (for single-binary turbo codes). For example the LLR for the value of a decoded bit d_k *(a-priori* information) is given by

$$L(d_k)=\ln\left(\frac{p(dk=+1)}{p(dk=-1)}\right)$$

Where $p\ (d_k = +1/-1)$ is the probability that the decoded bit $d_k = +1$ or $d_k = -1$ respectively.
The LLRs give two valuable information, the amplitude of the probability for the correct decision and its magnitude which determines the sign of the bit.

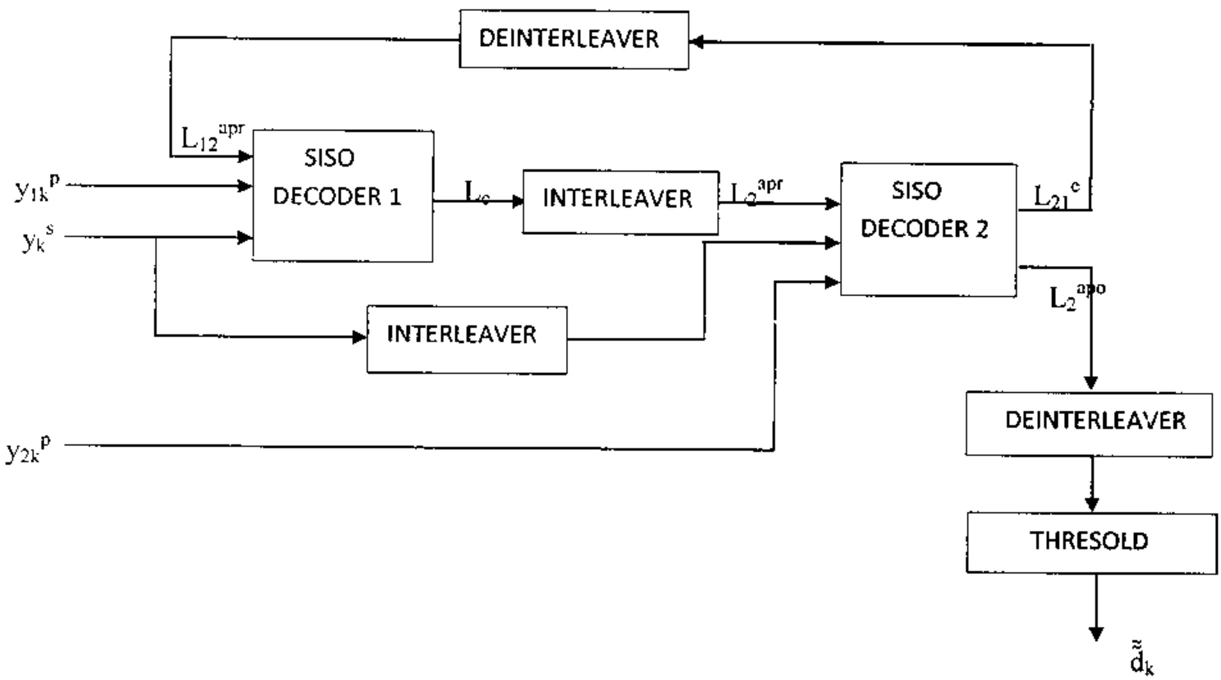

Fig. 6.3 Structure of an iterative (turbo code) decoder

There are two types of SISO decoding algorithms which are appropriate for turbo codes: MAP algorithm also known as BCJR algorithm introduced by Bahl, Cocke, Jelinek and Raviv and the Soft Output Viterbi Algorithm (SOVA) proposed by Hagenauer and Hoeher. In this dissertation, we will only describe the MAP algorithm and its modifications, since the performance of the MAP algorithm is better than SOVA.

6.3.1 ITERATIVE DECODING

The decoder shown in Fig. 6.3 works in an iterative manner. In the first iteration, the first MAP decoder takes as inputs the received sequences of the systematic y_k^s and first parity symbols y_{1k}^p from the output of the demodulator and calculates a soft output as its estimate of the information sequence. This soft output is the extrinsic information L_{12}^e. L_{12}^e is then interleaved, using an interleaver similar to the one used in the turbo code encoder, and then used as a priori information L_2^{apr} to the second MAP decoder along with the other channels outputs, which are in this case, an interleaved version of the systematic bits and the second parity bits y_{2k}.

The second MAP decoder yields the extrinsic information L_{21}^e which is deinterleaved before being fed back to the first MAP decoder as its *a-priori* information before the second iteration starts. This cycle is repeated and after reaching a number of iterations, the second decoder yields *a posteriori* information L_2^{apo} which is deinterleaved and then passed through a threshold to determine the decoded bit $\tilde{d}_k$ The BER performance of the turbo codes improves as the number of iterations used increases. In the low SNR region, the error performance of turbo codes is very poor as it is the case for other error correcting codes.

However the error rate can be lowered by increasing the number of iterations, but is not appropriate for most communication systems. In the waterfall region, the error performance drops quickly and can be further lowered by increasing the number of iterations. In the error floor region, the error rate is almost flat and does not improve much after a certain number of iterations.

CHAPTER 7

DUAL REPEAT PUNCTURED TURBO CODES

7.1 INTRODUCTION

The DRPTC is an extension of the repeat-punctured turbo code scheme. Repeat-Punctured Turbo Codes shows an improvement in the performance of turbo codes for moderate to high SNRs. The exceptional performance achieved with turbo codes is primarily due to the interleaver, which is responsible for spectral-thinning of the generated codewords. In the conventional turbo codes, the frame length is set identical to the interleaver size. It has been shown that increasing the frame length leads to an increase in the performance.

However, this would not be tolerable in many communication systems, e.g. real-time voice communications, since increasing frame length would result in large encoding/decoding delays. Increasing frame length is tantamount to increasing interleaver size in conventional turbo codes.

With a pseudo-random interleaver the weight-two input sequence is known to produce low-weight codewords for the conventional turbo code. However, with RPTC, a weight-two input sequence becomes a weight-$2T$ input sequence, where T is the repetition factor. This significantly improves the performance of the turbo code since the probability of the second constituent encoder producing a low-weight codeword is reduced due to the recursive nature of the component encoders.

7.2 DRPTC ENCODER

The structure of the DRPTC encoder is shown in Fig. 7.1. In each messaging interval an input message sequence of length N is produced at the input. The information frame is sent directly to the output to form the systematic output branch. Prior to processing for the parity sequences, the input frame is repeated T times. This allows for the use of larger interleavers prior to the constituent encoders processing their respective inputs. Note that there are two interleavers, i.e. two interleaver mappings π_1 and π_2, utilized in this scheme, or more generally, as many interleavers as parity branches implemented.

After interleaving, the enlarged message sequences are encoded by recursive systematic convolutional coders. The output codewords are again of length *TN* or depending on the code rate of the component encoder if not a single-rate one. The resultant parity sequences $y_1^{p1}, y_2^{p1}, \ldots, y_{TN}^{p1}$ and $y_1^{p2}, y_2^{p2}, \ldots, y_{TN}^{p2}$ are now responsible for the decreased code rate, i.e. $\frac{N}{N+TN+TN} = \frac{1}{1+2T}$ to counteract this, puncturing is employed to recover this loss in the code rate.

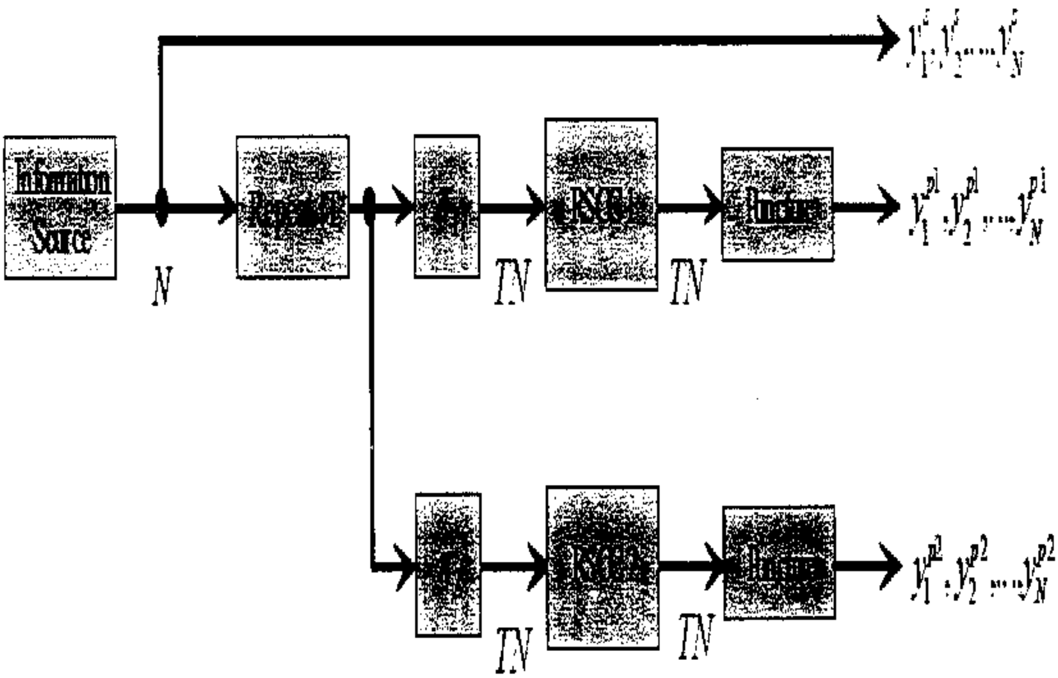

Fig. 7.1 Structure of the proposed DRPTC encoder

Fig. 7.2 illustrates how the puncturing is carried out for DRPTC. For a repetition factor *T=2*, every second bit from every *T* bits is punctured in the parity sequences. The memory of the code is, $m=2$. Puncturing of the systematic output need not be done, since we don't employ repetition here, which is unnecessary at this point.

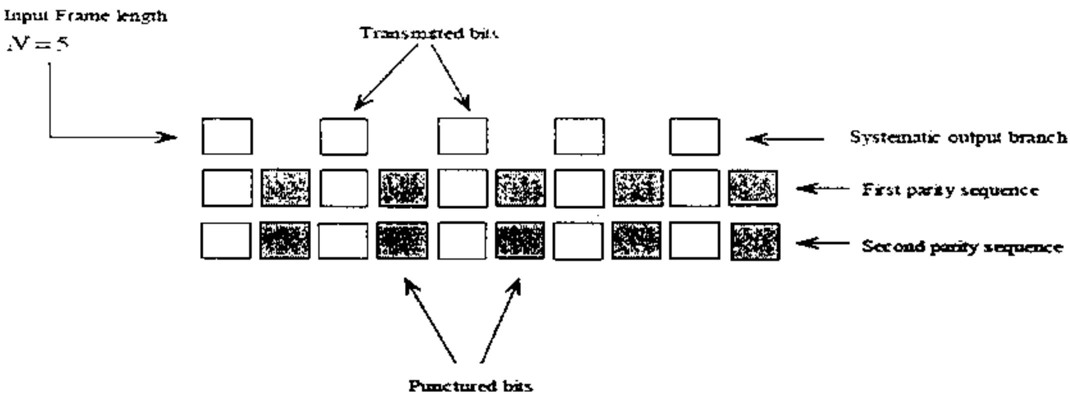

Fig.7.2 Puncturing at transmitter, T=2, m=2

However, these systematic bits need to be repeated again at the receiver prior to decoding, which contributes to an increase in performance exhibited by the code. In this paper, a repetition factor, $T=2$ was used, however, T can be increased to lower the code rate even further or to generate a series of rate-compatible codes.

The distance spectrum shows that a RPTC with $T=3$ has slightly more codewords at the low-weight end of the spectrum than when $T=2$ was employed. In addition, it is important to note that as T is increased the computational complexity also increases. The manner in which the trellis is terminated has effect on the performance of the DRPTC, similar to classical turbo codes.

7.3 DRPTC DECODER

The decoding of dual-repeat-punctured turbo codes is of a higher complexity than that of classical turbo codes and is based on the log-MAP & SOVA algorithm [3], [9]. The structure of the decoder is presented in Fig. 7.3. Prior to feeding inputs to respective component decoders, the sequences at the receiver front-end need to be reconstructed (due to puncturing at the transmitter) in a manner so as to satisfy the processing needs of the constituent decoders. Figure 4 illustrates how these sequences are prepared for decoding.

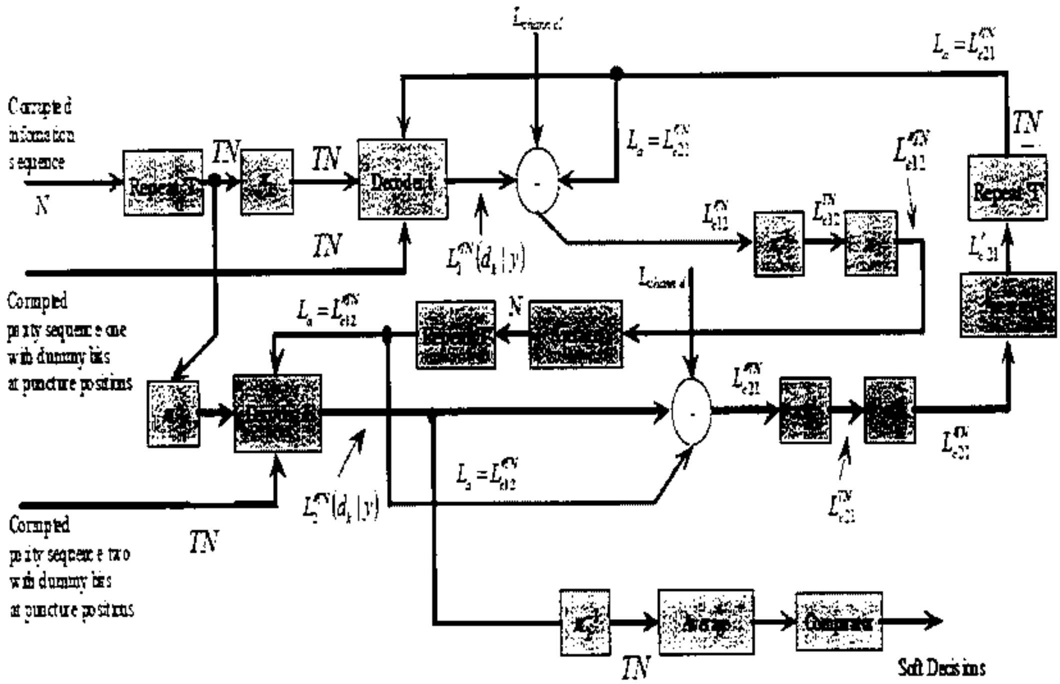

Fig. 7.3 Structure of DRPTC Decoder

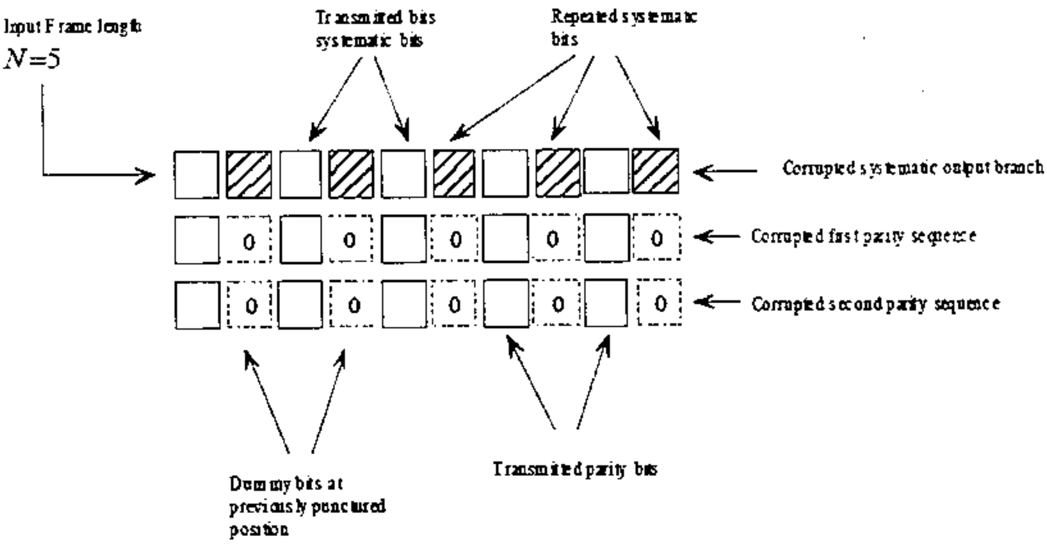

Fig. 7.4 Puncturing prior to decoding, T=2, m=2

The corrupted systematic output sequence is of length N. This sequence is first repeated T times. The hatched blocks in the diagram show the repeated systematic bits. For the parity sequences, dummy bits need to be introduced into previously punctured positions. In the lower part of Fig. 7.4 the parity sequences are depicted, the previously punctured positions are now occupied by dummy bits '0', which represent erasures.

These positions are represented by shaded dash-outlined blocks. Also, since two interleavers are being used here it is useful to have a special naming convention. Figure 5 shows the type of naming convention that will be used. The input sequence ϑ_1 if interleaved by the first interleaver π_1 will result in an output sequence ϑ_1' and an input sequence ϑ_1 if interleaved by the second interleaver mapping π_2 will result in a sequence ϑ_1'' Reverting back to the structure of the decoder in Fig. 7.3,

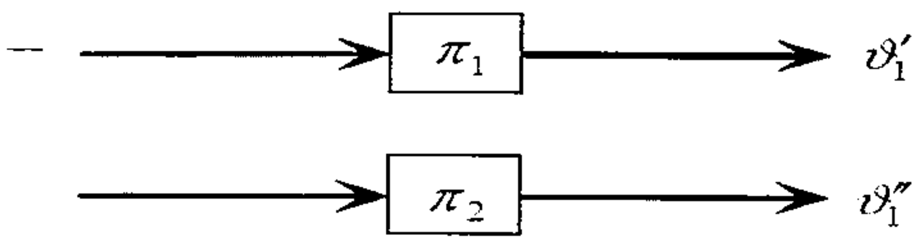

Fig. 7.5 Naming convention used when two interleavers areutilized

Decoder one accepts two inputs in the first iteration, viz. the repeated corrupted systematic sequence and the corrupted first parity sequence. The repeated systematic sequence$y_1^s, y_2^s, \ldots, y_{TN}^s$ has to be interleaved by the first interleaver mapping $\pi_1$before being sent to Decoder 1. Decoder one then uses these two sequences to generate soft log-likelihood decisions$L_1^{TN}(d_k \mid y)$. These soft log-likelihood ratios (LLRs) are next converted into soft extrinsic decisions L_{e12}^{TN}. Prior to supplying this as a-priori information to the second decoder, the extrinsic information is first deinterleaved π_1^{-1} (L_{e12}^{TN}then interleaved by interleaver mapping two to yield the extrinsic sequence L_{e12}^{TN} . Averaging and repetition of the extrinsic decisions can be utilized to improve the reliability of the assist information $L_a = L_{e12}^{TN}$ for Decoder 2.

The second component decoder takes three sequences as input, viz. the assist or a-priori information supplied from Decoder 1, the repeated corrupted systematic sequence, which has to be first interleaved by the second interleaver mapping, and the corrupted second parity sequence with dummy bits in punctured positions. Decoder two now uses these inputs to produce soft LLRs $L_2^{TN}(d_k \mid y)$. These soft LLRs are routinely converted into soft extrinsic information L_{e21}^{TN}, which has to be deinterleaved and then interleaved in the opposite manner to the extrinsic information from Decoder 1.

The extrinsic information L_{e21}^{TN} is deinterleaved by the second interleaver mapping to yield L_{e12}^{TN} . This new sequence is next interleaved by the first interleaver matrix, i.e. $\pi_1(L_{e12}^{TN})$ to produce L_{e21}^{TN}. Next this new extrinsic information sequence is averaged and repeated to improve the reliability of the a priori information $L_a = L_{e21}^{TN}$ supplied to Decoder one in the second iteration.

This exchange of extrinsic information between the decoders is iterated several times after which the TN soft LLRs from Decoder 2, $L_2^{TN}(d_k \mid y)$ are deinterleaved by π_2^{-1}, then averaged and converted into hard decisions with a comparator to yield the message sequence estimate, $\tilde{d}_1$, $\tilde{d}_2$,..., $\tilde{d}_N$. The averaging and repetition involved in the mutual exchange of information between the decoders is quite crucial to this scheme and serves to improve the performance by a substantial amount. In addition it is needed for scaling the soft decisions passed between the decoders.

CHAPTER 8

RESULTS AND DISCUSSION

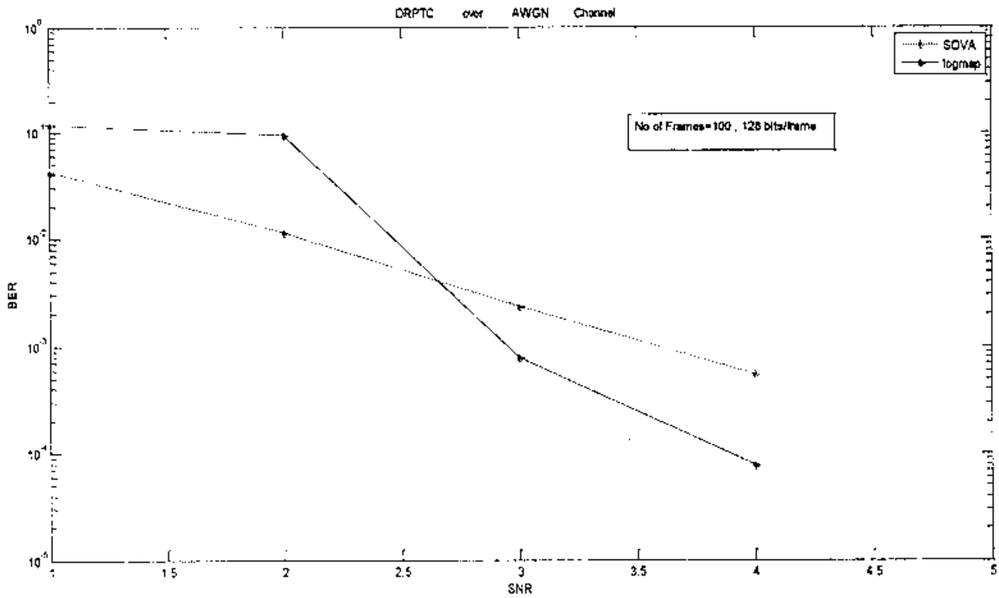

Fig. 8.1 DRPTC over AWGN channel, 128 bits/frames

Bit error rate is defined by the ratio between number received error bits to total transmitted bits. From Fig. 8.1, 128 bits/frames were transmitted over AWGN channel using Dual repeat punctured turbo codes. Compared to Log-MAP SOVA algorithm is fast and not accurate.

Similarly, in Fig. 8.2, 128 bits/frames were transmitted over Rayleigh fading channel using Dual repeat punctured turbo codes. Here SOVA gives good performance compared Log-MAP algorithm. In Fig. 8.3 and Fig. 8.4 256 bits/frames were transmitted over both AWGN and Rayleigh fading channels using Dual repeat punctured turbo codes. Here also Log-Map gives good performance compared to SOVA. In Fig. 8.5 and Fig. 8.6 512 bits/frames were transmitted over both AWGN and Rayleigh fading channels using Dual repeat punctured turbo codes. Here also SOVA gives good performance compared to Log-MAP

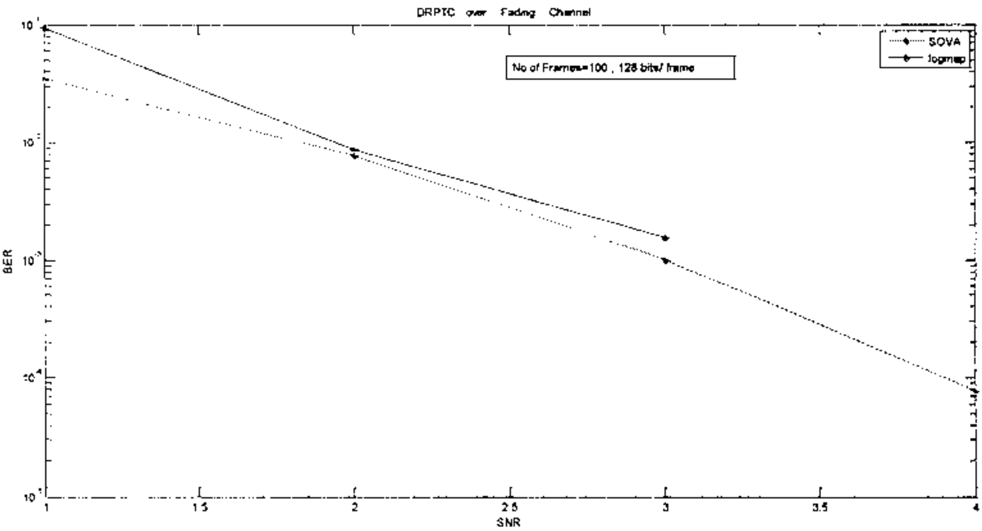

Fig. 8.2 DRPTC over Fading channel, 128 bits/frames

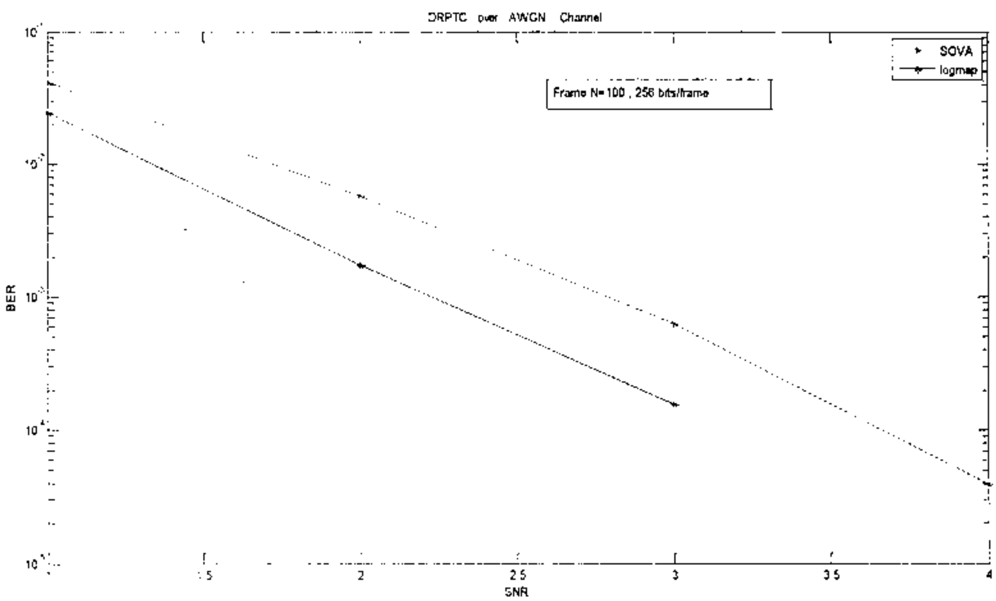

Fig. 8.3 DRPTC over AWGN channel, 256 bits/frames

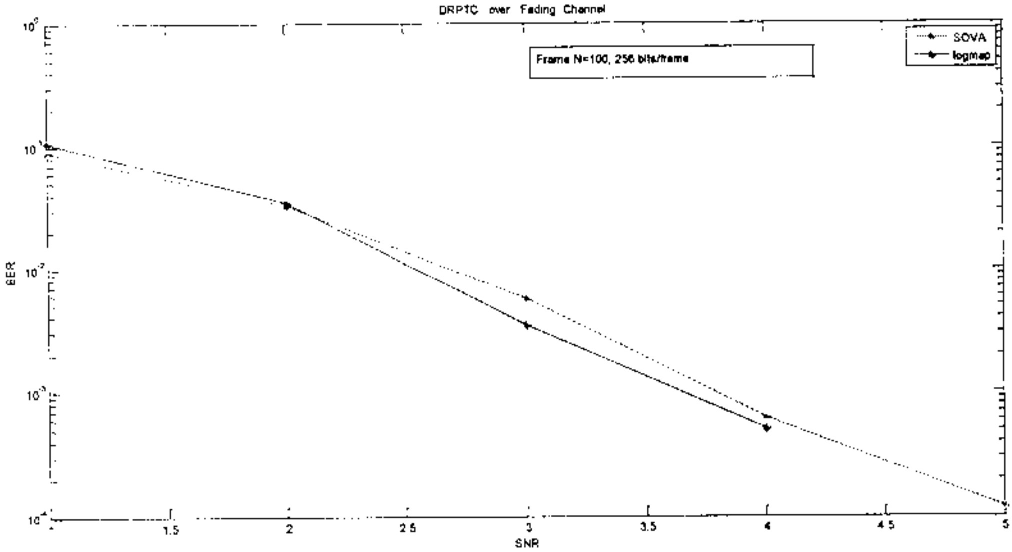

Fig. 8.4 DRPTC over Fading channel, 256 bits/frames

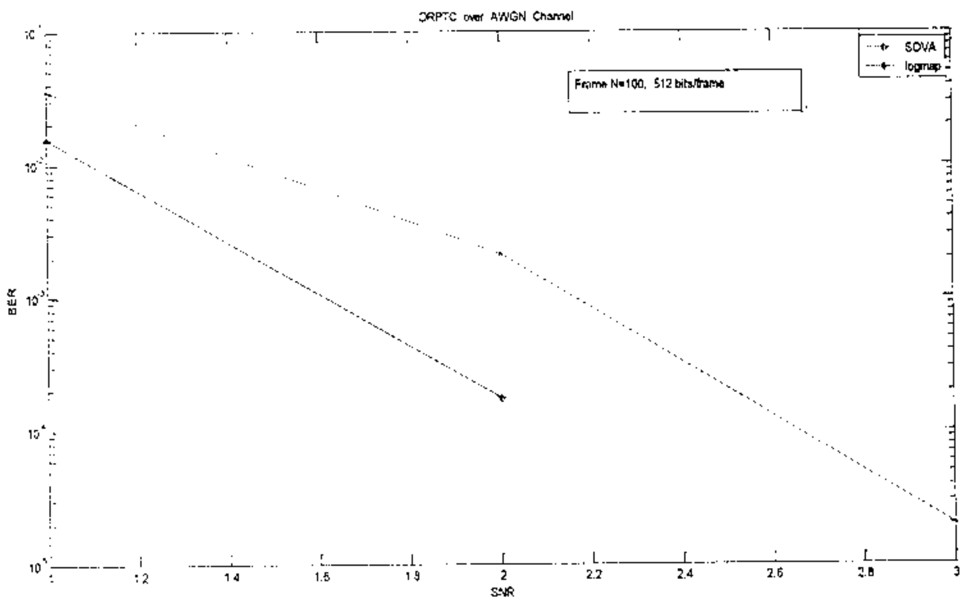

Fig. 8.5 DRPTC over AWGN channel, 512 bits/frames

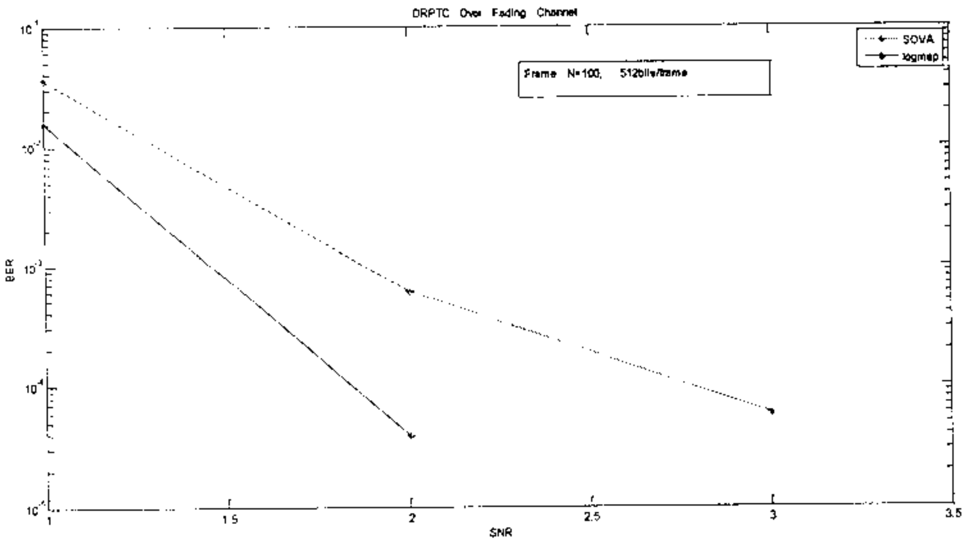

Fig. 8.6 DRPTC over Fading channel, 512 bits/frames

INFERENCE FROM RESULTS:

1. For a given SNR, the probability of error decreases with increasing number of iterations.
2. After ten iterations there is no significant improvement decoding performance.
3. For a fixed number of iterations, the probability of error of error decreases with increasing SNR.

CONCLUSION AND FUTURE SCOPE

Dual-repeat-puncture turbo codes have certain advantages in power and bandwidth efficiency when compared with widely used decoding algorithm such as Viterbi algorithm. This allows for less power to be used while getting exactly the same bit-error-rate performance. The performance of the Dual-Repeat-Punctured Turbo Codes has been shown to be greater compared to Turbo codes. Simulations were undertaken in both the AWGN and Rayleigh fading channel for fairly small frame lengths and the effect of using larger interleavers for turbo coding has shown its advantages. Future will research to improve the performance of encoder and decoder then incorporating in practical speech, video systems and in differential QoS.

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