CHAPTER I

INTRODUCTION

In recent years, there has been a growing interest in evolving suitable methods for achieving robust communication under highly noisy, interferring and jamming conditions. Such methods primarily concentrate on using error control codes on the one hand to minimize the bit error rate (BER) and implementing efficient detection schemes on the other hand to maximize the predetection signal to noise ratio (SNR). Error correcting codes have been developed as powerful tools to combat random and burst noise for practical systems. Interference and jammig are overcome by employing spread spectrum techniques.

Our objectives have been to study,develope and evaluate the performance of (i) adaptively decodable burst error correcting coders and decoders to take care of error bursts of various lengths, (ii) efficient detection and decoding techniques for direct sequence spread spectrum communication systems to combat jamming and interference and (iii) a hybrid system comprising of burst error correction and spread spectrum techniques to achieve greater reliability and robustness. Some details of the subject matter concerning the problems attacked and the approach to their solutions are outlined in the following paragraphs.

BURST ERRORS:

In general, there are two types of classifications of error correcting codes. The first type of classification as the block and the convolutional codes was made on the basis of the code construction. According to another type of classification, the codes are called either as the random error correcting (REC) or the burst error correcting (BEC) codes depending on the kind of channel errors to be corrected. However, there is no sharp demarcation line between the BEC and the REC codes. In fact, based on the studies made by many research workers on the statistics of the different channels, it is now realized that most of the real channels do suffer from both random as well as burst errors. The interest in finding codes for correcting burst errors has resulted in the development of such block codes as Fire codes, Reed-Solomon, interleaved and other codes and the convolutional codes such as recurrent codes, Berlekamp-Preparata-Massey code and Iwadare codes. Similarly, a lot of work has been reported in the literature on the development of random error correcting block as well as convolutional codes. Since, the real channels exhibit both random as well as burst errors, it is necessary to develop such codes which have the ability to correct both the types of errors. The convolutional codes based on adaptive decoding techniques, as suggested by Gallager , have been found to outperform the non-adaptive counterparts in correcting the random and the burst errors in real channels. But one disadvantage of the Gallager's code is that it performs very badly when there are unexpected errors in the guard space between the bursts. Sullivan modified the Gallager's adaptive error control scheme, though with a modest sacrifice in rate, to enable the decoder to correct burst errors even under such conditions. This is accomplished by using two convolutional codes, say C and C^*, where C^* contains C. Quite often, it will be useful and convenient to classify the burst errors into short and long bursts. This classification enables the use of a single BEC code to correct burst errors very efficiently.

In this thesis, we have developed a novel decoding shceme for correcting long as well as short bursts of errors. Decoders have also been developed to detect and correct the short and long error bursts by choosing suitable alogorithms. Thus, at the receiver the scheme consists of three basic operations viz., (i) adaptation to switch the mode of operation of the decoder between short and long bursts, (ii) correction of short bursts and (iii) correction of long bursts of errors. A code selection procedure to suit a given error statistics of the channel has been formulated and discussed in detail. In general, a rate half BEC code can be represented by the sub-generator polynomial

$$g(D) = D^{d}\,(g_0 + g_1 D + g_2 D^2 + \ldots + g_m D^m)$$

where D is the Huffman delay operator and g_i are the subgenerators which determine the error correcting capability of the code which also depends on the parameters m and d. Some of the special cases of these codes with different values of m and d have been evaluated. In order to enable the generation of codes which can correct both long as well as short bursts, the generator polynomial is suitably modified as

$$g(D) = D^{d}(1 + D^{m} + D^{m+M})$$

where d,m and M are selected in accordance with channel conditions. The design equations and the bounds for burst length and guard space requirements of the code have been derived in terms of the parameters d,m and M.

Experimental transmitter and receiver using the adaptive decoding techniques developed for correction of burst errors have been designed and implemented. Thorough practical tests have been carried out to evaluate the efficacy of the adaptive decoder by subjecting it to fixed and random bursts of errors having variable lengths. The decoder performance has also been evaluated with randomly varying guard space between error bursts. Though, the adaptive decoder is meant for correcting burst errors, its performance under the random errors have also been studied and the results are found to be quite interesting.

SS TECHNIQUES AND OPTIMUM DETECTION

To make a communication system immune to interference due to jamming, spread spectrum techniques are employed. The spectrum of the message signal is spread over a large bandwidth at the transmitter by employing some codes such as PN sequences etc., to make the power spectral density of the transmitted signal very low and hide the signal. The optimum techniques for detecting the signal in such coded digital communication systems involve matched filters for the binary pulses and correlators for the code. Due to the advent of LSI and VLSI technology, the digital implementation of the matched filters have become very attractive. The digital matached filters (DMF) are also preferred over the analog ones when jamming and real-time signal programmability are the issues . But, the necessary amplitude quantization of the incoming signal at the DMF receiver introduces a degradation in the output SNR depending on the statistics of the noise and the interference. In a zero-mean gaussian channel, the SNR loss has been shown to be approximately 2 dB with hard limiting. We have studied some of these aspects in detail and developed some novel methods of signal detection to compensate for this loss of SNR.

The receiver of a spread spectrum system essentially computes the correlation between the received signal and the locally generated code which is the replica of the desired transmitted code. Obviously, if the received code and the locally generated code match, the output is large and the signal is detected, otherwise not. Such correlation receivers are generally classified as active or passive depending on whether the code is locally generated or stored at the receiver. In this work, the passive correlation receivers which employ tapped delay lines in the form of matched filters have been studied. For optimum detection, a chip matched prefilter is used to maximize the SNR of the individual chips and then a matched filter correlator is used for decoding the desired signal. The realization of chip matched prefilter by using discrete components is more tedious than by using transversal filters with appropriate tap coefficients. Similarly, matched filter correlators for decoding can also be conveniently realized using transversal filters. Now, in order to realize the chip and the code MFs, a multiple sampling technique is suggested in this work where the received signal is oversampled (M samples per chip). The multiple samples are then processed either parallely by employing M code MFs or serially by summing them over the chip duration and using a single code matched filter correlator. Both the parallel and serial processings have been found analytically to be exactly equivalent. Of course, the complexities of their realizations in hardware are different. The serial matched filter structure has been studied in detail. For a correlated white Gaussian channel, the analytical expressions for the output SNR in terms of the code length N, the sample size M and the correlation coefficients ρ_{ij} are obtained.

As pointed out earlier, the hard limiter (quantizer) at the DMF receiver input results in a degradation in the SNR which in turn introduces errors in the detection process. These detection errors at the input will reflect ultimately at the output as the decoded errors. Therefore, methods for minimizing the detection errors are to be devised. In this thesis, two detection schemes have been described. In the first one we suggest the use of a multisampled chip MF before the hard limiter for improving the SNR whereas in the second scheme a decision based on majority voting among the hard limited samples is suggested for reducing the detection errors. In the first scheme, the chip MF is realized by sampling the received signal at M times the chip rate and passing it through tapped delay lines (TDL). The outputs of the TDL are summed hard limited and then used for decoding by a DMF. The overall system has been analysed by considering the correlation between the noise samples and an expression has been obtained for the output SNR relating clearly the gains due to the chip MF and the code DMF, and the loss in the hard limiter.

In the second scheme of the majority voting, the received signal is first hard limited and then sampled M times the code chip rate. The decision on the chip state is made on the basis of the majority voting over M samples. The code chips thus detected are then passed through a single DMF to detect the code word which in turn gives the data bits. The performance of the proposed majority voting method has been analysed in the thesis. An analytical expression for the chip detection error probability p, in terms of the input SNR and the number of samples n has been derived for correlated Gaussian channel. This shows how the performance depends on the number of samples. An experimental system has also been set up to evaluate the performance of the multisampling technique practically. Experimental results which are found to be in close agreement with the theory, are reported here.

The performance of the multisample majority decision scheme including the DMF in presence of non-gaussian interference such as square wave jamming or incoherent constant amplitude jamming is also very important in our study. It is pointed out by Cahn that this kind of jamming suppresses the signal and destroys the code structure completely. In order to avoid the inherent suppression effect, dithering is introduced to randomize the quantization process. We have carried out experiments to evaluate the performance of our scheme for non-gaussian interference and to study the effect of dither. Results of our experimental investigations are included in the thesis.

HYBRID SYSTEM:

In many practical situations both jamming as well as noise are present and the communication system is required to work against them. The deliberate jamming makes the SNR in the channel very low preventing the signal from detection whereas the noise causes errors making the communication unreliable. Therefore, a much more effective strategy for attaining high performance of low or moderate SNR on wide band channels is to combine error correcting codes with spread spectrum techniques employing codes of small length. This type of hybrid systems are expected to make the communication systems more reliable and robust. Since the errors caused by jamming and interference are normally bursty in nature, burst error correcting codes are the proper choice for the efficient decoding of digital signals. In this work, a simulation study using a rate-half convolutional BEC code and a direct swquence spread spectrum system has been carried out. The performance of this system has been evaluated under both noise and sinewave jamming.

THESIS ORGANISATION:

In all, the present thesis embodies the details and the results of our theoretical, simulation and experimental studies and investigations carried out on some aspects of coding and signal detection for robust communication in the presence of noise and interference. The whole thesis has been divided into six chapters.

This chapter introduces the subject matter of the work, basic problems addressed and our technical approach to their solutions. The second chapter presents a survey of the literature on burst error correcting block and convolutional code and spread spectrum techniques including the coded SS systems. Chapter III deals with adaptively decodable convolutional codes for burst errors. Our studies, investigations and the design aspects of the adaptive forward error correcting convolutional codec suitable for practical channels are described and the results are presented in this chapter. Chapter IV is devoted to the analysis and the performance evaluation of the multisampled digital matched filters and correlators for direct sequence spread spectrum systems. The problems of optimum detection of digital signals and their processing have been dealt with in this chapter. Analytical and experimental results are given here. A hybrid system comprising of the BEC coding and DS spread spectrum techniques has been discussed in chapter V. Results of the simulation studies on the improvements brought about by the use of BEC codes in combination with spread spectrum are reported here.

And finally, all the results of the various investigations as carried out in this work on adaptively decodable burst error correcting convolutional codes, multi-sampled digital matched filters and correlators and coded DS-SS systems have been summarised in chapter VI. The concluding remarks and the present and the future scope of the work are discussed in this chapter.