CHAPTER ONE

INTRODUCTION

Since the mid 1990's, the wireless communications industry has witnessed explosive growth. The worldwide cellular and personal communication subscriber base surpassed 600 million users in the late 2001, and the number of individual subscribers reached 2 billion at the end of 2006 [1, 2]. Most countries in the world also experience cellular subscription increases in excess of 40% per year. The rapid worldwide growth in cellular telephone subscribers has demonstrated conclusively that wireless communications is a popular voice and data transport mechanism. The widespread success of cellular communications has lead to the development of newer wireless systems and standards for many other types of communication traffic, besides mobile voice and data calls. For example, new standards and technologies such as Worldwide Interoperability for Microwave Access (WiMAX), are being implemented to allow wireless networks to replace fiber optic or copper lines between fixed points separated by several kilometres (fixed wireless access). Similarly, wireless networks have been increasingly used as a replacement for wires within homes, buildings, and offices through the deployment of Wireless Local Area Networks (WLANs). (See *Chapter 1, Section 1.6* for an in depth discussion on WLANs).

1.1 EVOLUTION OF CELLULAR NETWORKS, 2G – 2.5G

Most of today's cellular networks are currently in the phase of migrating from what is commonly known as Second Generation (2G) technologies to 3G technologies. The 2G mobile cellular systems uses digital radio transmission for traffic, as opposed to First Generation (1G) systems' analogue radio transmission. Thus, 2G networks have much higher capacity than the 1G systems. One frequency channel is simultaneously divided among several users (either by code or time division). Furthermore, hierarchical cell structures, in which the service area is covered by macrocells, microcells, and picocells, enhance the system capacity even further.

The four most popular 2G standards include three Time Division Multiple Access (TDMA) standards and one CDMA standard:

- Global System for Mobile (GSM) that is widely deployed in Europe, Asia, Australia, Africa, South America and some parts of the US,
- Interim Standard 136 (IS-136) in North America,
- Pacific Digital Cellular (PDC) that was deployed in Japan, and
- Interim Standard 95 (IS-95), a CDMA standard that was deployed in North and South America, Korea, China and Australia.

The 2G standards mentioned above represent the first set of wireless air interfaces to rely on digital modulation and sophisticated Digital Signal Processors (DSPs) in both the handset and BS.

Since 2G technologies use circuit-switched data modems that limit the data users to a single circuit-switched voice channel, data transmissions are generally limited to the data throughput rate of an individual user. This throughput rate is in the order of 10 kilo bits per second (kbps), which is too slow for rapid email and Internet browsing applications. In an effort to comply with increasing data throughput demands, new data-centric standards as well as packet switching techniques have been deployed that can be overlaid on existing 2G technologies. These new standards represent 2.5G technologies and allow 2G equipment to be upgraded to support the higher data rate transmissions for web-browsing, email traffic, mobile commerce etc.

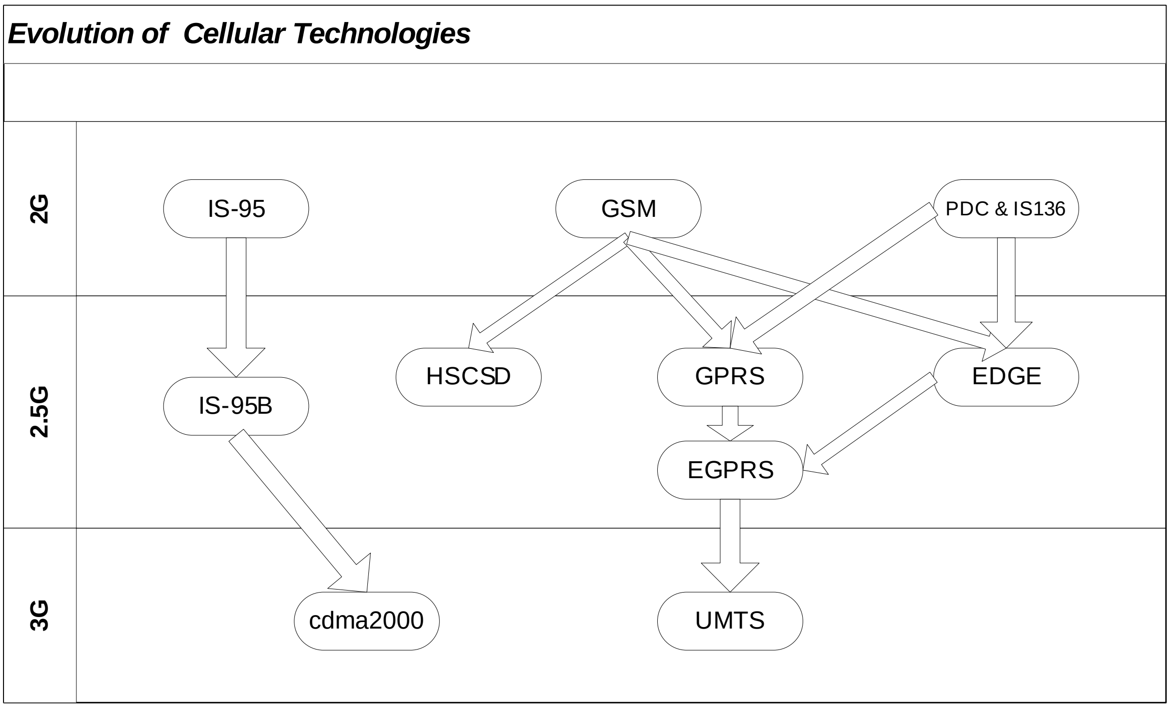

Figure 1.1. Various upgrade paths for 2G technologies [1].

The appropriate 2.5G upgrade path for a particular wireless carrier must match the original 2G technology choice. For example, the 2.5G upgrade solution for GSM must be based on the original 2G GSM technology since it would otherwise be incompatible and require complete equipment changes at the BS. *Figure 1.1* illustrates the various 2.5G and 3G upgrade paths for the major 2G technologies [1]. Also note that 3G technologies are discussed in *Section 1.2*.

The four most popular 2.5G standards include three TDMA standards, based on the GSM, IS-136 and PDC technologies, as well as one CDMA standard based on IS-95 technology, viz.

- 2.5G High Speed Circuit Switched Data (HSCSD) for GSM,
- 2.5G General Packet Radio Services (GPRS) for GSM and IS-136,
- 2.5G Enhanced Data Rates for GSM Evolution (EDGE) for GSM and IS-136, and
- The 2.5G IS-95B evolution of IS-95.

The main problem with GSM was its low air interface data rates that could originally provide only a 9.6 kbps user data rate, although a 14.4 kbps data rate specification was included later. HSCSD provided an easy way to speed things up by theoretically enabling the MS to use several time slots for a data connection, instead of only one time slot. Thus, the total rate is simply the number of time slots times the data rate of one slot. Although circuit switching is expensive in terms of resources, HSCSD was a relatively inexpensive way to upgrade the data capabilities, as it requires only software upgrades to the network (plus, of course, new HSCSD-capable phones). However, in current commercial implementations, the maximum number of time slots used is usually two, where one time slot may either use 9.6 kbps or 14.4 kbps data rates [4].

With GPRS technology, theoretical data rates of up to 171-kbps [4] with eight 21.5 kbps downlink timeslots can be obtained, or even higher if error correction is neglected. However, realistically, throughput rates of 40–60 kbps is currently being obtained [2]. Besides increased throughput rates, GPRS is packet switched, and thus does not occupy radio resources continuously. Bursty data is also well handled with GPRS, as it can adjust the assigned resources according to current needs. A drawback with GPRS is that it is not well suited for real-time applications, because the resource allocation in GPRS is contention based. Although the implementation of a GPRS system is much more expensive than that of an HSCSD system, it is seen as a necessary step toward improved data capabilities and an important step toward a 3G system, as 3G Partnership Project (3GPP) core networks are based on combined GSM and GPRS core networks.

With EDGE, Eight-Phase Shift Keying (8PSK) was adopted that increased the data rates of standard GSM up to three times [4]. EDGE is an attractive upgrade for GSM networks, as it only requires a software upgrade of BSs if the Radio Frequency (RF) amplifiers can handle the non-constant envelope modulation with EDGE’s relatively high peak-to-average power ratio. It does not replace, but rather coexists with the old Gaussian Minimum Shift Keying (GMSK) modulation, so mobile users can continue using their old phones. Thus, it provides a trade-off for quality of service, i.e. data rate vs. Bit Error Rate (BER). Another reason for retaining the old GMSK is that 8PSK can only be used effectively over a short distance; thus GMSK is still needed for wide area coverage.

Combining EDGE with GPRS results in the combination known as Enhanced GPRS (EGPRS). The maximum data rate of EGPRS using eight time slots (and adequate error protection) is 384 kbps. Note that the much advertised 384 kbps is only achieved by using all radio resources per allocated frequency carrier, and even then only when the MS is close to the BS. EDGE Circuit Switched Data (ECSD) is the combination of EDGE and HSCSD and it also provides data rates three times that offered by standard HSCSD.

The evolutionary path from an IS-95 system into a full cdma2000 system can take many forms [4]. The first step would be IS-95B, which increases the data rate from 14.4 kbps to 64 kbps. The next step is the cdma2000 1xRTT system that consists of four levels: The first level is known as 1xRTT release 0, or simply 1xRTT. This release can provide a 144-kbps peak data rate. The next level is the 1xRTT release A, which can give 384 kbps rates. Level 3, i.e. Single Carrier Evolved Data Only (1xEV-DO) can be regarded as a 3G system according to the International Telecommunication Union (ITU). This level, which only comprises of a data channel, can provide data rates up to 2.4 Mega bits per second (Mbps). The final level supporting data and voice is known as Single Carrier Evolved Data and Voice (1xEV-DV) and provides data rates of up to 3 Mbps.

Note that only new coding schemes and small changes in how data are slotted into the time frames improved the effective throughput rate of the data in all of the above mentioned 2.5G technologies. However, this was not the case for further improvement to 3G. New research had to be done in order to further improve data throughput rates beyond 1.8 Mbps, for example, High Speed Downlink Packet Access (HSDPA) that can provide theoretical data rates of up to 14.4 Mbps. However, effective throughput rates of 800 kbps is currently being obtained.

1.2 3G WIRELESS NETWORKS

With the increased data throughput rates of current 3G systems, wireless access is provided in ways that have never been possible before, e.g. TV broadcasting to MSs, etc. Multi-megabit Internet access, communication using VoIP, unparalleled network capacity and “always on” access are just some of the advantages delivered by current 3G systems.

Furthermore, with the increased data throughput rates provided by 3G, users have the ability to receive live music, conduct interactive web sessions and have simultaneous voice and data access with multiple parties at the same time using a single mobile handset, whether driving, walking or standing still.

The ITU formulated a plan to implement a global frequency band in the 2000 MHz range that would support a single wireless communication standard for all countries throughout the world. This plan, called International Mobile Telephone 2000 (IMT-2000), has been successful in helping to cultivate active debate and technical analysis for new high-speed mobile telephone solutions, compared to 2G. However, as seen in *Figure 1.1*, the hope for a single worldwide standard has not materialized, as the worldwide user community remains split between two camps, namely: GSM, IS-136, PDC and the IS-95B SS technologies, respectively.

The eventual 3G evolution of the IS-95B system lead to cdma2000. Several variants of cdma2000 are currently being developed, but they are all based on the fundamentals of IS-95 and IS-95B technologies, i.e. SS technology. The eventual 3G evolution for GSM, IS-136, and PDC systems lead to the Universal Mobile Telecommunications Service (UMTS). UMTS is based on the network fundamentals of GSM, as well as the merged versions of GSM and IS-136 through EDGE. The ITU IMT-2000 standards organizations are currently separated into two major organizations reflecting the two 3G camps, namely: 3G Partnership Project (3GPP) for UMTS standards (Europe and Asia) and 3G Partnership Project Two (3GPP2) for cdma2000 standards (USA). It is fair to say that these two major 3G technology camps, i.e. cdma2000 and UMTS, will remain popular throughout the early part of the 21st century. It should also be stated here that Wideband CDMA (WCDMA) terminology is used for both UMTS and cdma2000. By definition, the bandwidth of a WCDMA system is 5 MHz or more. Thus, the nominal bandwidth was also chosen to be 5 MHz for all 3G WCDMA proposals. This particular bandwidth was chosen because it is sufficient to provide data rates of 144 and 384 kbps (i.e. 3G targets), and even 2 Mbps under favourable conditions, while providing sufficient temporal resolution to exploit the multipath structures of typical mobile channels, called multipath diversity by employing RAKE combining techniques [11]. This concept is briefly introduced in the next section.

1.3 DIVERSITY TECHNIQUES FOR MULTIPATH CHANNELS

Transmit and/or receive diversity are a powerful communication technique that provides wireless link improvement at relatively low cost. Diversity techniques are based on the notion that errors occur at reception when the channel attenuation is large, e.g., when the channel is in a deep fade. If the receiver can be supplied with several replicas of the same information signal, transmitted over independent fading channels, the probability that all the signal components will fade simultaneously is reduced considerably. That is, if p is the probability that any one signal will fade below some critical value, then p^L is the probability that all L independent fading replicas of the same signal will simultaneously fade below the critical value. Several ways exist in which the receiver can be provided with L independent fading replicas of the same information bearing signal. The most widely used are:

- **Frequency diversity [1, 5, 6, 7].** The same information bearing signal is transmitted on L carriers, where the separation between successive carriers equals or exceeds the coherence bandwidth of the channel (see Appendix C, *Sections C.4*).
- **Time diversity [1, 5, 6, 7].** A second method for achieving L independently fading versions of the same information bearing signal is to transmit the signal in L different time slots, where the separation between successive time slots equals or exceeds the coherence time of the channel (see Appendix C, *Sections C.4*). Note that the fading channel fits the model of a bursty error channel.
- **Antenna diversity [6, 8, 9, 10].** This method is also known as space diversity. It achieves diversity by employing multiple antennas, where the antenna correlation properties depend on angular spread. For example, a single transmit antenna may be employed with multiple receive antennas. The latter must be spaced sufficiently far apart so that the multipath components in the signal have significantly different propagation paths at the receive antennas. Usually a separation of a few wavelengths is required between two antennas in order to obtain signals that fade independently. For example, GSM uses an antenna separation distance of ten wavelengths at their BSs.

Furthermore, the transmission of the same information may either be viewed at different frequencies or in different time slots (or both) as a simple form of repetition coding. The separation of the diversity transmissions in time (the coherence time of the channel) or in frequency (the coherence bandwidth of the channel), is basically a form of block-interleaving of the bits in a repetition code in an attempt to break up the error bursts, thereby obtaining independent errors that facilitates the incorporation of powerful coding mechanisms traditionally designed for AWGN channels.

A more sophisticated method for obtaining diversity is based on the use of a signal having a bandwidth much greater than the coherence bandwidth of the channel. Such signals, typically known as wideband signals, will resolve the multipath components (see *Chapter 3*), providing the receiver with several independently fading signal paths. The optimum receiver for processing the wideband signal is a RAKE correlator capturing all multipath paths and was invented by Price and Green in 1958 [11].

Other notable diversity techniques are angle-of-arrival diversity and polarization diversity [1]. However, these techniques have not been used as widely as the techniques described above.

1.4 CLASSIC ERROR CORRECTION CODING

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

A channel coder operates on the digital message (source) data by encoding the source information into a code sequence for transmission through the channel. The most widely used error detection and correction codes are:

- Block codes,
- Convolutional codes, and
- Coded Modulation.

Block codes encode blocks of data at a time and the encoding process only considers current data. Convolutional codes also encode blocks of data, but the encoding algorithm also considers previous data. Coded modulation is a type of coding that maximises the bandwidth efficiency of a modulation technique, for example Trellis Coded Modulation [6]. Other forms of coding also exist, for example concatenated codes are used to combine the best features of coding schemes that may be made up of block codes and convolution codes. The first concatenated code was proposed by Forney in 1966 [12] where the output of a RS code was then convolutionally encoded. Iteratively decoded codes is a type of code that are decoded in an iterative way, for example Turbo codes [13] and codes employing belief propagation (Bayesian Interference) [6]. The first two mentioned codes are briefly described below, merely for completeness. For a general overview of error correction codes, see Proakis [6]. The aim of this section is simply to provide an example to show the performance increase of a communication system when error correction coding is used (See *Figure 1.4*).

1.4.1 Block codes

A block code consists of a set of fixed-length vectors, called code words. The length of a code word is the number of elements in the vector, denoted by χ. Block codes, as shown in *Figure 1.2*, use an encoding scheme that takes Ω-data bits to produce χ encoded bits, where $\Omega < \chi$. The resulting block code is referred to as a (χ, k) code, and the ratio $\Omega/\chi = R_c$ is defined to be the rate of the code. Besides the code rate parameter R_c, an important parameter of block codes is the minimum Hamming distance of the code set. This is a measure of the error detection and correction capabilities of the code word. Some of the block coding algorithms used in practice include: Hamming codes, Hadamard codes, Golay codes, cyclic Golay codes and Bose-Chaudhuri-Hocquenghem (BCH) codes [6]. An example of a powerful non-binary BCH code is Reed-Solomon (RS) codes [6], which have good distance properties and allow protection against burst errors. Hard decision decoding algorithms, such as the Berlekamp-Massey algorithm [14], have also been developed for RS codes, making these codes applicable for real time implementation in many applications, such as Compact Disk Read Only Memories (CD-ROMs).

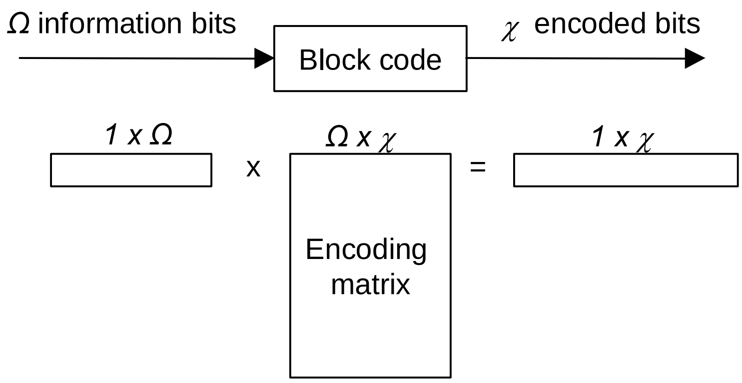

Figure 1.2. Block coding.

1.4.2 Convolutional codes

Convolutional codes are fundamentally different from block codes in that information sequences are not grouped into distinct blocks and then encoded. Instead, a continuous sequence of information bits is mapped into a continuous sequence of encoder output bits. Information bits are encoded by passing Ω-information bits at a time through a linear *Q*-stage shift register, as shown in *Figure 1.3*. A sequence of χ-encoded bits are obtained after Ω-information bits have been passed serially through an *Q*-stage shift register. The code rate $R_c = \Omega / \chi$. The parameter κ, called the constraint length, indicates the number of input data bits that the current output is dependent on and determines how powerful and

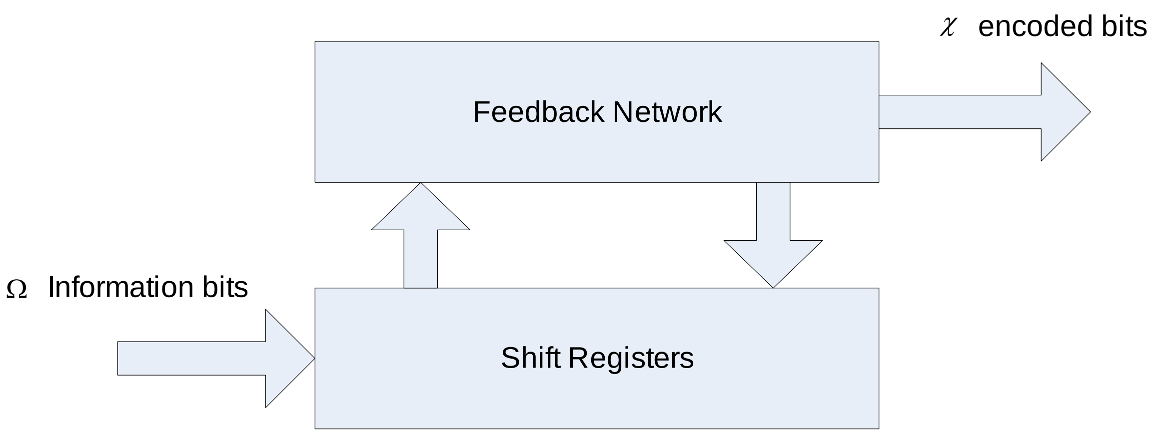

Figure 1.3. Convolutional coding consisting of a shift register and feedback network.

complex the code will be. The two popular algorithms used to decode convolutional codes are the Viterbi Algorithm (VA) [6] and the Maximum a-Posteriori (MAP) algorithm. The MAP algorithm is numerically more complex than the VA, but it offers a better BER performance [6], whereas the VA offers a better frame error rate (FER).

1.5 PERFORMANCE GAIN USING ERROR CORRECTION CODING

As an example of how error correction coding can improve the performance of a communication system, see *Figure 1.4* (taken from [15]). In *Figure 1.4* the performance of a Viterbi decoded (7,5,3) RS block code, is shown in flat fading channel conditions. Notice that the coded performance has a much lower bit error probability than the uncoded performance. Furthermore, it can also be noted that soft decision decoding has a 2dB asymptotic performance advantage over hard decision decoding.

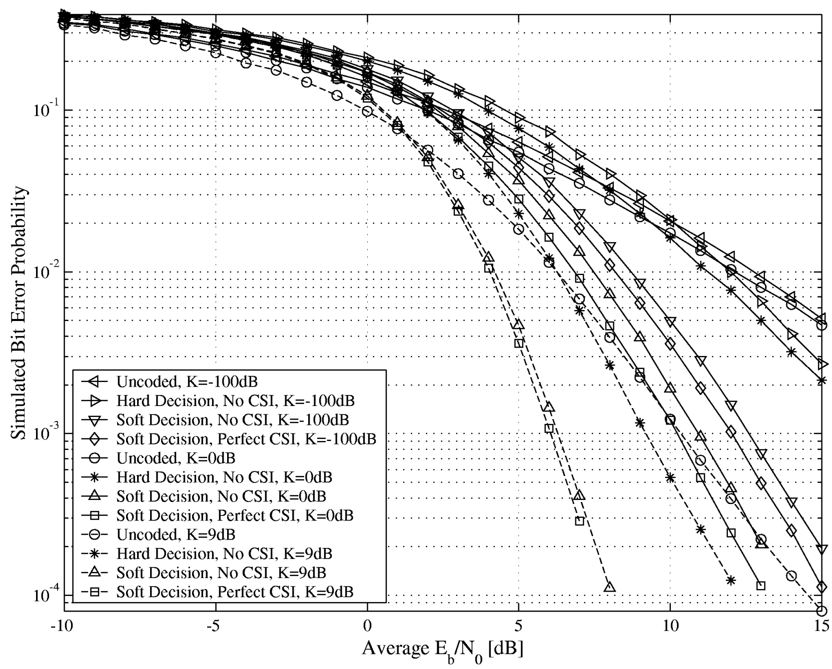

Figure 1.4. BER performance of a (7,5,3) RS block code in flat fading channel conditions, taken from [15]. K = -100dB denotes Rayleigh fading, whereas K = 0dB and K = 9dB denotes Rician fading.

1.6 RELEVANCE OF THIS DISSERTATION

As stated above, wireless communication systems, such as 3G communication systems, are constantly in demand of improving performance, such as improved voice quality, higher bit rate data services (greater than 2 Mbps), power efficiency, etc. At the same time, the remote units (mobile cellular phones) are required to be small, compact and lightweight. Further demands require that the mobile units operate reliably in various types of environments, for example, micro -, macro -, and picocellular, indoor, outdoor and urban, as well as suburban environments. In other words, the communication systems are required to have:

- increased throughput rates,
- increased capacity,
- better speech quality,
- better coverage,
- more power efficiency,
- better bandwidth efficiency, and
- to be deployed in various environments.

All of the above mentioned are constrained to services that must remain affordable for widespread market acceptance.

Time-varying multipath fading [1, 5, 6, 16] is a phenomenon which makes reliable wireless transmission very difficult. It is known to be extremely difficult to increase signal quality or reduce the BER in multipath fading channels compared to Additive White Gaussian Noise (AWGN) channels. In AWGN channels, the BER may be improved from 10^{-2} to 10^{-3} with a power sacrifice of only 1 or 2dB higher Signal to Noise Ratio (SNR) in the case of using Binary Phase Shift Keying (BPSK) (see Appendix C, *Figure C.19*). However, to achieve the same with fading channel conditions, it may require up to 10 dB more SNR (see Appendix C, *Figure C.19*). Thus, improvements in SNR may not be achievable by simply increasing transmit power or additional bandwidth, because of the increasing demand of power and bandwidth efficiency. A partial solution to this problem is to find methods to effectively reduce the effect of multipath fading at both the mobile unit and the BS, without a sacrifice in bandwidth or the need for additional power availability.

Transmitter power control is theoretically an effective technique to mitigate multipath fading in a wireless channel [1,5]. In addition, if the channel multipath conditions, as experienced by the receiver, are known at the transmitter, the transmitter can predistort the signal in order to compensate for the effect of the distortion introduced by the channel [1].

Other effective techniques to mitigate multipath fading are time-, frequency- and antenna diversity [6]. When possible, wireless communication systems should be designed to encompass all forms of available diversity to ensure adequate performance. From a practical point of view, space diversity reception in the uplink (MS to BS) is one of the most effective and, hence, widely applied techniques for mitigating the effects of multipath fading. The classical approach is to use multiple antennas at the receiver and perform combining or selection and switching in order to improve the received signal quality. The major problems with the receive diversity approach at the MS are cost, size and power. As a result, antenna diversity techniques have almost exclusively been applied to the BS. A BS often serves hundreds to thousands of MSs. From an economical perspective it will thus be more efficient to add equipment to the BS rather than the MSs. For this reason a motivation arose for developing transmit diversity schemes. Alamouti [3] proposed a transmit diversity technique using two transmit antennas and one receive antenna, providing the same diversity order than using one transmit antenna and two receive antennas. This technique is capable of effectively mitigating the effects of multipath fading in the downlink (BS to MS) without adding any additional antennas at the MS. Thus, by using multiple antennas the effective throughput as well as system capacity of the system can be improved. Presently, communication systems like 3G and WLANs make use of two transmit antennas in order to improve the data throughput rate and system capacity. However, this dissertation strives to extend the number of transmit antennas at the BS to more than 2 antennas with full rate and full diversity in order to further increase data throughput rates and system capacity. Together with the multiple antennas diversity approach, techniques such as time interleaving used with error correction coding, can provide further performance increases.

1.7 CONTRIBUTIONS AND PUBLICATIONS

1.7.1 Contributions

This study's main focus is directed towards ST coding and modulation techniques within the field of digital communications. Minor contributions include a communication channel simulator, which had to be reproduced from Staphorst [17], in order to simulate the Direct Sequence Space-Time Spreading (DSSTS) in realistic environments. This baseband equivalent communication channel simulator was implemented and verified by comparing the simulation results with theoretical results. One advantage of using this channel is that Doppler and fading effects can be simulated without the need to simulate an RF carrier frequency.

1. Major contributions from this study on ST coding and modulation are as follows:
   a. Increasing the number of transmit and receive antennas in wireless communications has been researched over the past few years because of advantages such as increased data throughput rate as well as increased system capacity. However, inherent limitations with ST coding theory discussed in *Section 2.3*, only 2 and 4 transmit antennas can obtain both full rate and full diversity. As a result, this study was motivated by the quest for having more than 4 transmit antennas that can achieve both full rate and full diversity in order to exploit diversity gain as well as increase system capacity. Thus, a scheme called DSSTS was proposed in this study that achieves full rate and full diversity for any number of transmit antennas in multiples of two. It extends conventional ST coding theory by combining ST coding with SS modulation. An advantage of DSSTS over current Space-Time Spreading (STS) techniques is that it uses 50% less spreading codes.

   b. A novel idea to relax the Alamouti ST coding transmit diversity scheme's dependency on the channel being quasi-static. The condition of the channel being quasi-static over two consecutive time intervals in low throughput rate applications with a high Doppler component has been countered by replacing the concept of a two interval approach with two conventional simple complex-spreaded RAKE receiver-based Direct Sequence Spread Spectrum Multiple Access (DS/SSMA) communication systems operating in parallel in one combined time interval. This new approach will henceforth be referred to as the Space-Sequence Transmit Diversity (SSTD) alternative to Alamouti's two-symbol ST coding transmit diversity technique.

2. Major contributions from this study in terms of modelling and simulation of communication systems are as follows:
   a. Flexible DSSTS transmitter and receiver structures are presented in *Chapter 5*. These structures are flexible in the sense that it allows for any type of spreading sequence to be used in conjunction with the ST coding.

   b. The novel performance evaluation platform used to create the DSSTS simulation results presented in *Chapter 6*, are presented in *Chapter 5*. This performance evaluation platform allows the user to simulate results for any number of transmit antennas at different Doppler frequencies under Rayleigh or Rician fading channel conditions.

   c. The novel SSTD transmitter and receiver structures are presented in *Chapter 5*.

3. Important contributions from this study's simulation results are as follows:
   a. It was shown that the use of Walsh spreading sequences in a Rayleigh fading channel was insufficient to exploit the diversity gain over self-noise, created by adding additional antennas to the DSSTS scheme. However, with a Rayleigh fading-channel the number of transmit antennas effects the performance of the DSSTS scheme differently in low and high mobility environments: At low velocities, i.e. at low Doppler frequencies, the DSSTS's 10 transmit antennas outperformed the DSSTS's 10 transmit antennas scenario for high mobile velocities.

b. It was also shown that the use of Walsh spreading sequences in a Rician fading channel did not produce any diversity gain due to self-noise created by adding additional antennas to the DSSTS scheme. The effect of Doppler spread in a Rician fading environment has no effect on the DSSTS scheme's BER performance at a Rician factor of 9dB and higher.

c. From the results presented in *Chapter 6*, it is evident that there exists an optimal trade-off between diversity and self-noise. Considering DSSTS in a Rician or Rayleigh fading channel, the 2 transmit antenna scenario yields the best diversity gains in both channels.

d. From the capacity simulation results presented in *Chapter 6*, it is evident that the DSSTS scheme achieves equal capacity performance gain than an open loop capacity system. It was also shown that the DSSTS 4 Tx antenna scenario outperformed other 4 Tx diversity schemes as presented in Papadias *et al.* [18].

e. The proposed SSTD scheme is successful in combating the effects of multipath fading for small Code Division Multiple Access (CDMA) user loads. However, as a rule of thumb, a user load equal to the square root of the spreading sequence length, *N*, divided by two constitutes the user load at which the SSTD scheme was not capable of overcoming the combined effects of Multi-User Interference (MUI) and multipath fading.

1.7.2 Publications

During this study, the author researched and co-authored two local conference papers and submitted one paper for possible publication in an accredited journal. These papers are listed below, as well as their relevance to this dissertation:

1. *Simulation Study of a Space-Sequence Transmit Diversity Scheme for DS/SSMA Systems (Part I)* [19], co-authored by L. Staphorts and L.P. Linde, presented at ICT 2005 in Cape Town South Africa, described the SSTD scheme proposed in this

dissertation. The 1st of the two papers elaborated on the Complex Spreading Sequence (CSS) families employed, as well as the complex DS/SSMA Quadrature Phase Shift Keying (QPSK) transmitter and RAKE receiver simulator structures.

2. The 2nd of the two paper series, *Simulation Study of a Space-Sequence Transmit Diversity Scheme for DS/SSMA Systems (Part II)* [20], presented the encoder and decoder structures of the proposed SSTD scheme, the complete multi-user multipath fading simulation platform, built around the proposed SSTD scheme and several simulation results, comparing the performances of simple uncoded and SSTD encoded wideband systems in multi-user multipath fading channel conditions for the different CSS families considered.

3. *A Full Rate, Full Diversity Space-Time Block Code for an Arbitrary number of Transmit Antennas* [21], presented the encoder and decoder structures of the proposed DSSTS scheme, several BER performance simulation results as well as the DSSTS capacity analysis.

1.8 OVERVIEW OF CHAPTERS

This dissertation is structured as follows: In *Chapter 2*, ST block coding is presented as a transmit diversity scheme. The chapter includes ST encoding, decoding, ST theory and ST capacity. Lastly, an overview of current ST literature is presented. In *Chapter 3* SS modulation techniques are presented. The chapter includes theory on Direct Sequence Spread Spectrum (DS-SS) and CSSs, as well as theory on the techniques of combining ST coding and SS modulation techniques. Lastly, an overview of current ST CDMA literature is presented. In *Chapter 4* the flat fading channel simulation platform used in this dissertation is presented, as well as a mathematical analysis of the channel. In *Chapter 5* the encoding, decoding and capacity of the new DSSTS scheme are presented, as well as how to use SSTD to relieve the Alamouti ST block-decoding scheme from the necessity for the channel to be quasi-static [3, 19, 20]. In *Chapter 6,* BER simulation results, as well as capacity plots for the new DSSTS scheme are presented. The BER simulation results were obtained from the simulation platform described in *Chapter 4*. Lastly, conclusions on the performance of the new DSSTS scheme, as well as possible future work, are presented.

CHAPTER TWO

TRANSMIT DIVERSITY TECHNIQUES

As stated in the introduction, the fundamental phenomenon that makes reliable wireless communications difficult is time varying multipath fading. Improving the information quality or reducing the effective BER in a multipath fading channel is known to be a daunting task. Besides improving the BER through channel coding, an effective technique to combat multipath fading is through transmitter power control [1, 5]. By predistorting the signal at the transmitter, the effects of the channel can be overcome at the receiver if the transmitter knows the channel conditions experienced at the receiver. Problems encountered with this approach are as follows:

- Increased power levels. Increasing the transmitter power is costly due to the high cost involved in developing power amplifiers with large dynamic output power ranges, while at the same time having the ability to respond to fast power fluctuations.
- Lack of knowledge of the channel conditions. The transmitter normally does not have knowledge of the channel conditions. The received information at the receiver can be fed back to the transmitter in order to obtain the channel effects, but this causes a delay in the system. By using feedback, the effective throughput is also reduced and complexity is added to the circuitry at the transmitter, as well as the receiver.

Antenna diversity [6, 8, 22, 23] is a practical and effective method to combat time-varying multipath fading and improve the wireless link quality. In contrast to the classical approach to using multiple receive antennas at the receiver, a new approach of using multiple transmit antennas at the transmitter has recently been developed [3]. This scheme enables wireless cellular networks to use multiple antennas exclusively at the BS and obtain diversity in the uplink as well as downlink, i.e. receive diversity in the uplink, and transmit diversity in the downlink. The use of multiple receive antennas as a receive diversity technique and applying Maximal Ratio Receiver Combining (MRRC) principles are discussed in detail in Appendix A.

This chapter is structured as follows: In *Section 2.1* a general overview of transmit diversity is presented. In *Section 2.2* the transmission model, as well as encoder and decoder structures of ST block coding are presented, followed by ST coding theory in *Section 2.3*. An analysis of ST coding probability of error is presented in *Section 2.4*, and ST capacity in *Section 2.5*. Lastly, the focus of *Section 2.6* and *Section 2.7* is on ST coding literature.

2.1 TRANSMIT DIVERSITY

Transmit diversity is a method of using multiple transmit antennas to achieve the same diversity as would be gained by using multiple receive antennas. The work done by Alamouti [3], followed by Tarokh *et al.* [22], laid the foundation for research in the field of ST coding, i.e. a method of obtaining transmit diversity. Results obtained from the transmit diversity scheme presented by Alamouti [3], showed that it performed within 3dB of the MRRC scheme employing multiple receive antennas. This was also theoretically derived and verified by Gao *et al.* [24]. It was also shown in [25] that by using four transmit antennas and m receive antennas, performances similar to an $4m$ level MRRC scheme can be achieved.

The following topics are discussed in detail in the Appendix A:

- A mathematical formulation of the encoding, transmission and decoding models.

- Performance comparison between MRRC and transmit diversity employing 2 transmit antennas.
- Implementation issues concerned with the differences observed between the transmit diversity scheme and the MRRC scheme.

2.1.1 Throughput increase of multiple antenna systems

Three mechanisms of multiple antenna systems that can improve the throughput of current communication systems are [25, 26, 27]:

- Beamforming
- Diversity combining
- Spatial multiplexing

Beamforming involves the weighing and summation of signals at each individual antenna to control the angular response of the array. Thus, angular isolation between wanted signals and interference signals is an important advantage of beamforming. This technique can be applied to both transmit as well as receive diversity if knowledge of the channel is available. Another advantage beamforming offers, is an increased SNR at the receiver, thus improving link reliability, and increasing the communications range.

Diversity combining is a well established technique for combating fading, thus improving the mean throughput rate. To obtain diversity by means of multiple antenna systems, the same information is transmitted from different antennas at the same time. Provided that the paths the individual information signals traverses on their way to the receiver antenna array are uncorrelated, reception is improved because if one channel experience a deep fade, the information may still be conveyed through the other channel. Thus, for each additional diversity branch added by adding a new antenna, the probability of the combined received signal being badly attenuated decreases. However, because of the inherent limitations with ST coding theory discussed in *Section 2.3*, only 2 and 4 transmit antennas can obtain both full rate and full diversity. Thus, this study was motivated by the quest for having more than 4 transmit antennas that can achieve both full rate and full diversity in order to exploit diversity gain.

Spatial multiplexing involves the use of *m*-receive antennas to separate the interfering substreams of data that were transmitted from *n*-antennas at the transmitter. In theory, separating the interfering substreams directly, translates to improved BER performance and a corresponding improvement of the data throughput rate.

2.1.2 Capacity and spectral efficiency

The effect of adding multiple antennas does not only increase the throughput rate of a communication system, but also has an effect on the capacity and spectral efficiency of the communications system. Shannon's research [29] on the channel capacity determined the upper bound on the rate of error-free communication in the case of a single band-limited AWGN channel. Extensions to Multiple Input Multiple Out (MIMO) systems that make use of multiple antennas, promise dramatic improvements in spectral efficiency [26, 27]. Recent research [26, 27, 30, 31] has shown that the capacity of a system, employing *n*-transmit and *m*-receive antennas, grows linearly with min(*n*,*m*), but only logarithmically with the SNR. Thus, multiple antennas are a realistic solution of improving the spectral efficiency for communications systems.

2.2 ST CODING AS A TRANSMIT DIVERSITY TECHNIQUE

Tarokh *et al*. [22] generalized Alamouti's transmit diversity scheme of two transmit antennas and proved that it is based on orthogonal designs. This laid the foundation for ST block code designs for both real and complex signal constellations. These ST block codes achieve the maximum possible transmission rate for any number of transmit antennas using real signal constellations, such as Pulse Amplitude Modulation (PAM). For complex signal constellations, such as Phase Shift Keying (PSK) and Quadrature Amplitude Modulation (QAM), ST block codes can theoretically only achieve full rate and full diversity for 2 transmit antenna, whereas ½ of the maximum possible transmission rate for any other number of transmit antennas [14] can be obtained. For specifically 3 and 4 transmit antennas, Tarokh *et al*. [25] showed that ¾ of the maximum transmission rate can be achieved using the orthogonal designs presented in [22].

2.2.1 Transmission model

Suppose the downlink in a cellular environment consists of a BS with *n* antennas and a mobile with *m* antennas. In each time slot *t*, signals c_t^i, $i = 1, 2, 3, \ldots, n$ are transmitted from *n* transmit antennas. Assuming a flat fading channel, described in *Chapter 4, Section 4.1*, with path gains defined as $h = \mathrm{Re}\{h\} + j \cdot \mathrm{Im}\{h\}$, constant over a frame length *l*, the received signal at antenna *j* at time *t* is

$$r_t^j = \sum_{i=1}^{n} h_{i,j}\, c_t^i + \eta_t^j \qquad (2.1)$$

where $h_{i,j}$ is defined as the path gain from transmit antenna *i* to receive antenna *j*. Here η_t is the AWGN defined in Appendix C, *Section C.5*. The average energy of the symbol transmitted from each antenna is scaled by the factor 1/*n*, thus producing unit transmit power and unit receive power at each receive antenna if the channel power is also normalised to unity.

Assuming the availability of perfect Channel State Information (CSI), the receiver computes the decision metric [7]

$$DM = \sum_{t=1}^{l} \sum_{j=1}^{m} \left| r_t^j - \sum_{i=1}^{n} h_{i,j} c_t^i \right|^2 \qquad (2.2)$$

over all signals

$$c_1^1, c_1^2, \ldots c_1^n, c_2^1, c_2^2, \ldots c_2^n, \ldots c_l^1, c_l^2, \ldots, c_l^n \qquad (2.3)$$

and decides in favour of the signal that minimizes the sum of (2.2).

2.2.2 Encoding algorithm

To elaborate on the transmit diversity scheme, a ST block code is defined by an *l x n* transmission matrix *C*. The matrix *C* consists of linear combinations of *k* variables, denoted as x_1, x_2, x_3,…,x_k, as well as their conjugates. These variables and conjugates are grouped in such a manner that the matrix is orthogonal [22]. The topic of orthogonality is discussed in *Section 2.3*. For example, C_2^{1rate} represents a code that utilizes two transmit antennas and obtains full rate. C_2^{1rate} can be defined by

$$C_2^{1rate} = \begin{bmatrix} x_1 & x_2 \\ -x_2^* & x_1^* \end{bmatrix} \quad (2.4)$$

i.e., the well known Alamouti matrix proposed in [3]. It is evident from C_2^{1rate} that n = 2, indicating that two transmission antennas are used (seen from the number of columns in C_2^{1rate}) and l = 2, which means that the frame length is equal to 2 symbol periods (seen as the number of rows in C_2^{1rate}).

Assume that transmission occurs in baseband and a signal constellation ν is employed with 2^q signal points, where q denotes the number of bits in a signal point. At time slot t = 1, kq bits from bitstream $b(t)$ arrive at the encoder and are mapped to constellation signals a_1, a_2, a_3,…,a_k of the signal constellation ν. An additional matrix A is then formed by setting x_i = a_i for i = 1, 2, 3, …, k from C. Thus, similar to C, A consists of linear combinations of constellation points and their conjugates, as shown in *Equation (2.5)*.

$$A_2^{1rate} = \begin{bmatrix} a_1 & a_2 \\ -a_2^* & a_1^* \end{bmatrix} \quad (2.5)$$

These constellation signals are transmitted from the n transmit antennas. So, the i'th column of A represents the symbols transmitted from the i'th antenna and the t'th row represents the transmitted symbols at time t. Because of C's orthogonality [22], A is also orthogonal and thus allows a simple decoding process at the receiver. Since l time slots are used to transmit k symbols, $R = k/l$ can be defined as the code rate. For example, in C_2^{1rate}, R = 1.

Other examples are the following ½ rate ST block codes [25]. These codes are orthogonal designs presented in [22].

$$C_3^{1/2\,rate} = \begin{bmatrix} x_1 & x_2 & x_3 \\ -x_2 & x_1 & -x_4 \\ -x_3 & x_4 & x_1 \\ -x_4 & -x_3 & x_2 \\ x_1^* & x_2^* & x_3^* \\ -x_2^* & x_1^* & -x_4^* \\ -x_3^* & x_4^* & x_1^* \\ -x_4^* & -x_3^* & x_2^* \end{bmatrix} \tag{2.6}$$

$$C_4^{1/2\,rate} = \begin{bmatrix} x_1 & x_2 & x_3 & x_4 \\ -x_2 & x_1 & -x_4 & x_3 \\ -x_3 & x_4 & x_1 & -x_2 \\ -x_4 & -x_3 & x_2 & x_1 \\ x_1^* & x_2^* & x_3^* & x_4^* \\ -x_2^* & x_1^* & -x_4^* & x_3^* \\ -x_3^* & x_4^* & x_1^* & -x_2^* \\ -x_4^* & -x_3^* & x_2^* & x_1^* \end{bmatrix} \tag{2.7}$$

Examples of ¾ rate ST block codes are given in *Equations (2.8)* and *(2.9)* [22].

$$C_3^{3/4\,rate} = \begin{bmatrix} x_1 & x_2 & \frac{x_3}{\sqrt{2}} \\ -x_2^* & x_1^* & \frac{x_3}{\sqrt{2}} \\ \frac{x_3^*}{\sqrt{2}} & \frac{x_3^*}{\sqrt{2}} & \frac{-x_1 - x_1^* + x_2 - x_2^*}{2} \\ \frac{x_3^*}{\sqrt{2}} & -\frac{x_3^*}{\sqrt{2}} & \frac{x_2 + x_2^* + x_1 - x_1^*}{2} \end{bmatrix} \tag{2.8}$$

$$C_4^{3/4\,rate} = \begin{bmatrix} x_1 & x_2 & \frac{x_3}{\sqrt{2}} & \frac{x_3}{\sqrt{2}} \\ -x_2^* & x_1^* & \frac{x_3}{\sqrt{2}} & -\frac{x_3}{\sqrt{2}} \\ \frac{x_3^*}{\sqrt{2}} & \frac{x_3^*}{\sqrt{2}} & \frac{\left(-x_1 - x_1^* + x_2 - x_2^*\right)}{2} & \frac{\left(-x_2 - x_2^* + x_1 - x_1^*\right)}{2} \\ \frac{x_3^*}{\sqrt{2}} & -\frac{x_3^*}{\sqrt{2}} & \frac{\left(x_2 + x_2^* + x_1 - x_1^*\right)}{2} & -\frac{\left(x_1 + x_1^* + x_2 - x_2^*\right)}{2} \end{bmatrix} \quad (2.9)$$

2.2.3 Decoding algorithm

Tarokh *et al*. [25] has shown that Maximum Likelihood (ML) decoding of any ST block code can be achieved by using only linear processing at the receiver. The decoding algorithm is explained by means of an example. Consider the ST block code C_2^{1rate} defined in *Equation (2.4)*. Assume that 2^q bits from bitstream *b(t)* arrive at the encoder and select two complex symbols a_1 and a_2, as described in the previous section. These two symbols are simultaneously transmitted from antennas 1 and 2, respectively. At the second time slot, signals $-a_2^*$ and a_1^* are simultaneously transmitted from antennas 1 and 2, respectively.

Then, from *Equation (2.2)*, the ML detection amounts to minimizing the decision metric

$$DM = \sum_{j=1}^{m} \left(\left| r_1^j - h_{1,j} a_1 - h_{2,j} a_2 \right|^2 + \left| r_2^j - h_{1,j} a_2^* - h_{2,j} a_1^* \right|^2 \right) \quad (2.10)$$

over all possible values of a_1 and a_2.

Expanding *Equation (2.10)* and deleting terms that are independent of the codewords, the following equation is obtained [25].

$$DM = -\sum_{j=1}^{m} \left[r_1^j h_{1,j}^* a_1^* + (r_1^j)^* h_{1,j} a_1 + r_1^j h_{2,j}^* a_2^* + (r_1^j)^* h_{2,j} a_2 - r_2^j h_{1,j}^* a_2 - (r_2^j)^* h_{1,j} a_2^* \right.$$

$$+ r_2^j h_{2,j}^* a_1 + (r_2^j)^* h_{2,j} a_1^* \Big] + \left(|a_1|^2 + |a_2|^2 \right) \sum_{j=1}^{m} \sum_{i=1}^{2} |h_{i,j}|^2 \qquad (2.11)$$

which decomposes into two parts, one that is only a function of a_1,

$$DM_{a_1} = -\sum_{j=1}^{m} \Big[r_1^j h_{1,j}^* a_1^* + (r_1^j)^* h_{1,j} a_1 + r_2^j h_{2,j}^* a_1 + (r_2^j)^* h_{2,j} a_1^* \Big] + |a_1|^2 \sum_{j=1}^{m} \sum_{i=1}^{2} |h_{i,j}|^2 \qquad (2.12)$$

and the other part that is only a function of a_2:

$$DM_{a_2} = -\sum_{j=1}^{m} \Big[r_1^j h_{2,j}^* a_2^* + (r_1^j)^* h_{2,j} a_2 - r_2^j h_{1,j}^* a_2 - (r_2^j)^* h_{1,j} a_2^* \Big] + |a_2|^2 \sum_{j=1}^{m} \sum_{i=1}^{2} |h_{i,j}|^2 \qquad (2.13)$$

which, for both cases a_1 and a_2, are equivalent to

$$DM_{a_1} = \left| \left[\sum_{j=1}^{m} \left(r_1^j h_{1,j}^* + (r_2^j)^* h_{2,j} \right) \right] - a_1 \right|^2 + \left(-1 + \sum_{j=1}^{m} \sum_{i=1}^{2} |h_{i,j}|^2 \right) |a_1|^2 \qquad (2.14)$$

and

$$DM_{a_2} = \left| \left[\sum_{j=1}^{m} \left(r_1^j h_{2,j}^* - (r_2^j)^* h_{1,j} \right) \right] - a_2 \right|^2 + \left(-1 + \sum_{j=1}^{m} \sum_{i=1}^{2} |h_{i,j}|^2 \right) |a_2|^2 \qquad (2.15)$$

respectively. Thus, minimization of *Equation (2.10)* is equivalent to minimizing *Equation (2.14)* for a_1 and *Equation (2.15)* for a_2.

Note that *Equations (2.14)* and *(2.15)* are similar to Alamouti's scheme shown in *Appendix B, Section B.3*, and thus proves the generalization of his scheme to an arbitrary number of transmit antennas. Also note that *Equation (2.14)* can be written as

$$DM_{a_1} = | \tilde{a}_1 - a_1 |^2 + \left(-1 + \sum_{j=1}^{m} \sum_{i=1}^{2} |h_{i,j}|^2 \right) |a_1|^2 \qquad (2.16)$$

where $\tilde{a}_1 = \sum_{j=1}^{m}\left(r_1^j h_{1,j}^* + (r_2^j)^* h_{2,j}\right)$. *Equation (2.15)* can be written as

$$DM_{a_2} = |\tilde{a}_2 - a_2|^2 + \left(-1 + \sum_{j=1}^{m}\sum_{i=1}^{2}\left|h_{i,j}\right|^2\right)\left|a_2\right|^2 \tag{2.17}$$

where $\tilde{a}_2 = \sum_{j=1}^{m}\left(r_1^j h_{2,j}^* - (r_2^j)^* h_{1,j}\right)$.

2.3 ST BLOCK CODING THEORY

In [22], code designs were presented in order to create orthogonal block codes. These design criteria states that a matrix X must obey

$$X^H X = \left(\left|x_1\right|^2 + \cdots + \left|x_n\right|^2\right)\cdot I \tag{2.18}$$

where I is the identity matrix, X is a square matrix with n columns, X^H is the conjugate transpose of matrix X and x_1, x_2, …, x_n is the entries of the first row of matrix X. Also note that n is the number of transmit antennas used.

The concept of using *Equation (2.18)* in designing a square ST code can best be described by means of an example: Suppose a ST code X, using 2 transmit antennas, is to be designed. This code's general form is given by

$$X = \begin{bmatrix} x_{11} & x_{12} \\ x_{21} & x_{22} \end{bmatrix} \tag{2.19}$$

where x_{ij} are complex symbols. Thus $X^H X$ is given by

$$\begin{aligned} X^H X &= \begin{bmatrix} x_{11}^* & x_{21}^* \\ x_{12}^* & x_{22}^* \end{bmatrix} \cdot \begin{bmatrix} x_{11} & x_{12} \\ x_{21} & x_{22} \end{bmatrix} \\ &= \begin{bmatrix} \left|x_{11}\right|^2 + \left|x_{21}\right|^2 & x_{11}^* x_{12} + x_{21}^* x_{22} \\ x_{12}^* x_{11} + x_{22}^* x_{21} & \left|x_{12}\right|^2 + \left|x_{22}\right|^2 \end{bmatrix} \end{aligned} \tag{2.20}$$

By using the design criteria presented in *Equation (2.18)*, the code is only orthogonal if $X^H X$ is diagonal, i.e. when

$$x_{11}^* x_{12} + x_{21}^* x_{22} = 0 \tag{2.21a}$$

and

$$x_{12}^* x_{11} + x_{22}^* x_{21} = 0 \tag{2.21b}$$

By using and rearranging *Equation (2.21a)*, $x_{22} = -x_{11}^* x_{12} / x_{21}^*$. Substituting x_{22} into *Equation (2.19)*, yields X as

$$X = \begin{bmatrix} x_{11} & x_{12} \\ x_{21} & -\dfrac{x_{11}^* x_{12}}{x_{21}^*} \end{bmatrix} \tag{2.22}$$

Also note that a similar result may be obtained by using *Equation (2.21b)*. By setting $x_{21} = -x_{12}^*$ we obtain

$$X = \begin{bmatrix} x_{11} & x_{12} \\ -x_{12}^* & x_{11}^* \end{bmatrix} \tag{2.23}$$

which is well known as the Alamouti matrix (see *Equation (2.4)*). This completes the proof that the Alamouti matrix is an orthogonal block code.

2.4 ST CODING ERROR PROBABILITY

In a flat fading Rayleigh channel, the probability of error for a ST block code based on Alamouti's scheme and utilising M-ary Phase Shift Keying (MPSK) modulation was derived by Gao *et al*. [24]. This probability of error was derived from the Probability Density Function (PDF) of the phase of the received signal. It should also be stated here that, although the procedure given in [24] applies to any M-ary modulation scheme, only the BPSK and QPSK modulation schemes are given in *Equations (2.24)* and *(2.25)* respectively, as only these apply to this dissertation.

The probability of error for a BPSK ST block code, is defined as [24]

$$P_{e,BPSK\ ST} = \frac{1}{2}\left[1 - \mu - \frac{1}{2}\mu\left(1 - \mu^2\right)\right] \tag{2.24}$$

and in the case of a QPSK modulation, the probability of error is [24]

$$P_{e,QPSK\ ST} = \frac{1}{2}\left[1 - \frac{\mu}{\sqrt{2 - \mu^2}} - \frac{\mu\left(1 - \mu^2\right)}{\left(2 - \mu^2\right)\sqrt{2 - \mu^2}}\right] \tag{2.25}$$

where μ is defined as the normalised cross-correlation between two random variables, one that is a function of channel 1, h_1, and the other a function of channel 2, h_2 [24]. The normalised cross-correlation is given by

$$\mu = \sqrt{\frac{\Gamma}{(\Gamma + 2)}} \tag{2.26}$$

where $\Gamma = E_b/N_0$ is the SNR per bit.

Equations (2.24) and *(2.25)* were simulated and are plotted in Appendix C, *Section C.8, Figure C.18* in order to compare the two error probability graphs. The probability of error for a flat fading Rayleigh channel (see Appendix C, *Section C.6.2, Equation (C.11)*), is also included as a reference graph. Also note that a 3dB difference exists between the BPSK and QPSK modulation techniques.

2.5 ST CODING CAPACITY

Digital communications using MIMO has recently emerged as one of the most significant technical breakthroughs in modern day communications. In [26] it has been shown that ST codes can achieve phenomenal capacity compared to a traditional single transmit and receive antenna system [29]. In order to define the concept of channel capacity, *Equation (2.1)* is rewritten in matrix form

$$\bar{r} = A\bar{h} + \bar{\eta} \tag{2.27}$$

where A is defined as a ST encoding matrix of transmitted symbols, $\bar{h}$ is the complex channel conditions and $\bar{\eta}$ is the received noise. $\bar{h}$ and $\bar{\eta}$ are defined as

$$\bar{h} = \begin{bmatrix} h_1 \\ \vdots \\ h_n \end{bmatrix}, \qquad \bar{\eta} = \begin{bmatrix} \eta_1 \\ \vdots \\ \eta_n \end{bmatrix} \tag{2.28}$$

However, the standard model used in [26] to derive MIMO channel capacity is

$$\bar{r} = H\bar{a} + \bar{\eta} \tag{2.29}$$

where

- $\bar{a} = [a_1 \cdots a_n]^T$ is the n x 1 vector of the transmitted symbols over n transmit antennas, each assumed to be of equal variance;
- H is the m x n channel matrix;
- $\bar{r}$ is the m x 1 vector of the received signals;
- $\bar{\eta}$ is defined in *Equation (2.28)* and represents a m x 1 vector of AWGN variables.

Thus, in order to use the MIMO channel capacity, *Equation (2.27)*, for ST coding schemes, the following condition must be satisfied:

$$A\bar{h} = H\bar{a} \tag{2.30}$$

By using the MIMO signal model in *Equation (2.29)*, the channel capacity bound of a Single-Input Single-Output system, i.e. the Shannon bound [29], is given by

$$CAP_{SISO}^{n=1,m=1} = \log_2\left(1 + SNR\,|h|^2\right) \tag{2.31}$$

where h is the normalised complex gain of a fixed wireless channel, i.e. $H = h$ in *Equation (2.29)*. In Foschini [26], *Equation (2.31)* was extended to accommodate the use of multiple transmit and receive antennas and hence the MIMO open-loop Shannon capacity of *Equation (2.29)* was derived. This capacity, known as MIMO channel capacity bound, is defined as:

$$CAP_{MIMO}^{n,m} = \log_2 \det\left(I_m + \frac{SNR}{n}\left(HH^H\right)\right) \tag{2.32}$$

where H is a (m x n) fading channel matrix. Note that *Equation (2.31)* and *Equation (2.32)* represents channel capacity bounds. In order to obtain the capacity, one needs to average over all fading states h. In *Chapter 6*, *Figure 6.22*, a plot of the capacity Complementary Cumulative Distribution Functions (CCDFs) for various systems with $n = m$ receive antennas is shown. Notice, as the number of antennas is increased, the capacity increases significantly. In fact, both Foschini [26] and Teletar [27] demonstrated that the capacity in *Equation (2.32)* grows linearly with min(m, n).

With the capacity defined as a random variable by *Equation (2.32)*, the issue of how to best characterise it, arises. Two commonly used methods are the mean capacity [27], and capacity outage [18, 26]. The mean capacity simply takes the expectancy, i.e. the mean of the capacity equations defined in *Equations (2.31)* and *(2.32)*, whereas capacity outage measures (by means of simulation) the probability that the capacity is above a certain capacity threshold. These are often denoted as $CAP_{0.1}$ or $CAP_{0.01}$, meaning that those capacity values are respectively supported 90% or 99% of the time and obtained from the CCDFs. Thus, this method also has an indication of the system reliability. In this study, the method of CCDFs will be used, as most authors tend to make use of this method.

2.6 OVERVIEW OF CURRENT ST CODING LITERATURE

Delay diversity [6] and other related schemes were among the first techniques presented to exploit transmit diversity. Delay diversity is a special case of Space-Time Trellis Coding (STTC) that was later developed by Tarokh *et al*. [32]. The generalized approach combines Trellis Coded Modulation (TCM) with transmit diversity techniques that perform very well in slow fading environments. The only drawback of these codes is that the decoding complexity grows exponentially with the number of antennas. A major contribution that emerged from this work was the rank and determinant criteria that became a benchmark in ST code design. A more structured method of STTC construction, ensuring full diversity, was later presented in [33, 34]. Due to receiver complexity, as stated above, alternative methods to employ effective ST coding was researched. The Alamouti code, given in *Equation (2.4)*, is remarkable for having an elegant and simple linear receiver and became a paradigm in ST block coding. Alamouti's scheme with two transmit antennas was later generalized by Tarokh *et al*. [22, 25], proving that the Alamouti matrix is based on orthogonal designs, which have full diversity and linear ML detectors that decouple the transmitted symbols. Unfortunately, the Hurwitz-Radon theorem showed that square complex linear processing orthogonal designs cannot achieve full diversity and full rate simultaneously for constant symbol constellations, except for the two transmit antenna case [22]. Using this work, a formula for the maximum achievable data rate for square code matrices was derived by Tirkkonen and Hottinen [35]. Several orthogonal codes have been discovered with full diversity for 3 or 4 transmit antennas, but these codes [22, 25] are only ¾ rate codes. As shown in [36, 37], it is possible to design orthogonal, full rate and full diversity complex codes for more than two transmit antennas for specific symbol constellations. For example, 4 transmit antennas obtaining full rate and full diversity, are presented in [36, 37] using constellation phase rotation for specialized PSK and PAM symbol constellations.

Another class of ST codes is the unitary ST modulation [38] and differential unitary modulation [39] that is almost similar to orthogonal codes. These codes use a set of unitary code matrices to represent data. The only drawback of the unitary modulation code compared to orthogonal codes is that the optimal receiver is more complex than the orthogonal code's receiver. This is due to the fact that the code matrix is not structured by symbols that can be decoupled for detection. These codes are typically non-square and designed for systems where CSI is unknown at the receiver.

The ABBA[1] code presented in [40] and similar codes [18, 41, 42] have full rate, but are quasi-orthogonal and offer a diversity order of only 2. The ST Transmit Diversity-Orthogonal Transmit Diversity (STTD-OTD) code [43] provides some diversity gain by grouping symbols into Alamouti blocks and transforming them using a Walsh-Hadamard matrix. For the 4 transmit antenna case, this orthogonal code has full rate and diversity order 2. Recently, an orthogonal full diversity, full rate ST block code for 4 transmit antennas was presented in [44]. However, perfect knowledge of the channel at the transmitter and receiver is required to cancel Inter-Symbol Interference (ISI) and ensure orthogonality.

It has been shown that full diversity and full rate can be achieved with Generalised Algebraic ST (GAST) codes, which use rotated constellations with a Hadamard transform [45]. In addition, these non-orthogonal codes offer a coding gain over comparable orthogonal codes, especially for large constellations and many transmit antennas. Another code in the literature utilizes ST diversity with unitary constellation rotating precoders [46, 47]. Constellation rotating codes essentially transmit a linear combination of the phase-rotated symbols through one antenna at a time, while leaving the other antennas silent. These codes are capable of achieving full rate and full diversity, but are not orthogonal.

In El Gamal *et al.* [48], an algebraic approach was used to construct new trellis and block codes for BPSK and QPSK modulated systems with an arbitrary number of transmit antennas, which guarantees that the ST code achieves full spatial diversity. These codes are actually convolutional codes, and for that reason do not achieve full rate.

[1] The ABBA code received it's name from the transmission matrix structure, i.e $Matrix = \begin{bmatrix} A & B \\ B & A \end{bmatrix}$

Other recently proposed schemes and research to achieve transmit diversity are:

- A ***Differential Detection Scheme***, proposed by Tarokh *et al*. [49]. This scheme exploits diversity by two transmit antennas when neither the transmitter nor the receiver has access to CSI. A further advantage is that encoding is simple and the receiver can be implemented with low decoding complexity.

- ***Iterative Maximum-likelihood Sequence Estimation for Space-Time Coded Systems***, proposed by Li *et al*. [50], is another scheme to decode ST codes without CSI. This scheme is based on ML sequence estimation and can be used for both quasi-static and non-static fading channels.

- ***Spherical Space-Time Codes***, proposed by Terry *et al*. [51], extends the trellis ST code concept to include signal mappings drawn from a N-dimensional sphere. These signal points are designed to increase the minimum squared distance between points in the constellation, without increasing the average transmit energy.

- ***ST Codes Based on Number Theory***, proposed by Damen *et al*. [52], is a scheme that achieves full data rate over two transmit antennas and 2 symbol periods. A further advantage is that the coding gain outperforms Alamouti's scheme [3] at low and high SNR when the number of transmit antennas is greater than one.

- ***ML Detection and Decoding for ST Codes.*** All the other schemes mentioned concentrate on the optimal code-design for ST coding systems. This method, proposed by Larsson *et al*. [53] takes a unified approach to interference-resistant detection of symbols transmitted over a MIMO channel and optimal (in a ML sense) information transfer from the ST detector to the channel decoder. That is, using soft decision decoding rather than hard decision decoding.

2.7 MIMO SYSTEMS EMPLOYING ST CODES

The advantages, such as mobility of WLAN and portable devices, promise to revolutionise the people's way of living, working and playing. In order to achieve throughputs similar to current Ethernet standards and long range operations, improvements need to be incorporated into WLAN. One such solution is to incorporate multiple antennas.

The use of multiple antennas is most recently introduced to the third generation cellular systems as well as the IEEE 802.16d/e, i.e. WiMAX standards, where the 802.16d standard is intended for static applications and the 802.16e standard for mobile applications. The mayor advantage of using multiple antennas at either side of the communication link is an improvement in link reliability and throughput. Until recently, antenna arrays were considered to be an obstacle, however in the microwave band (> 5GHz), the use of dual polar antennas solves the problem of compact ergonomic packaging of MSs. These improvements will enable WLAN to be able to compete with current wired Ethernet standards, with the added advantage of a cable free environment. However, besides the improvements in throughput and link reliability, security, affordability and energy efficiency are also areas that have to be addressed in order to compete with wired technologies.

Current WLANs, such as the IEEE 802.11a or IEEE 802.11g standards, offer a peak rate of 108 Mbps at the physical layer. However, at higher order layers, such as the Medium Access Control (MAC) layer, the average throughput is typically less than 20Mbps.

As stated in *Section 2.1.2*, extensions to MIMO systems that make use of multiple antenna systems, promise dramatic improvements in spectral efficiency. Thus, multiple antennas are a realistic solution for improving the spectral efficiency for future WLANs.