The Design of an HF Band Direct Sequence Point – to – Point Link for Rural Telecommunications

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ABSTRACT

The following work documents the design of an HF band direct sequence point-to-point link as used in a rural environment. The dissertation begins with a description of the overall document layout. An introduction into the problems associated with providing rural access is then given, with special emphasis on wireless technologies. It is argued that the attributes of HF band radio make it a good candidate for providing wireless communications links for under serviced rural areas in South Africa. The pitfalls and disadvantages of using an ionospheric-based medium in which to propagate an electromagnetic wave are discussed and several solutions are put forward. One of these solutions is Direct Sequence Code Division Multiple Access (DS-CDMA). A thorough analysis of the principles of DS-CDMA is given with emphasis on its abilities to combat the negating effects of the HF channel. A CDMA HF system is then proposed, outlining the various practical and theoretical aspects. Next, an HF channel model is designed and simulated, first with no spreading or coding, then with just spreading and finally with spreading and coding. It is found that although the extra diversity of the spreading and coding aid in reducing the bit error probability, more detailed local measurements and refinement in the design of the channel model and simulated system are needed to increase performance. Various aspects of a practical system that was built are then discussed, highlighting issues such as hardware interfacing and the software design of a man-machine-interface. Field measurements are also given with estimates on an upper limit on expected performance of a practical system. Finally conclusions are given, detailing the achievements and shortcomings of the research.

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LIST OF SYMBOLS

D	Distance of radio link
H	Height of antenna
F_1	First Fresnel Zone
f	Frequency in GHz
η	Throughput efficiency
n_w	Additive White Gaussian Noise
$S(t)$	Transmitted signal
$J(t)$	Interference signal
$R(t)$	Received signal
Gp, L	Process gain
Pe	Probability of error
η_{oJ}	Interferer power density

1. Introduction

1.1 Rural Telecommunications in South Africa

The advent of the information age has seen the rapid development of a variety of communications devices. As the impact of multimedia and the Internet becomes more prevalent, the ever-increasing need for bandwidth seems to show no signs of abatement. Real time video, push technology and seamless global connectivity have pushed an already overcrowded spectrum to its limits. Add to this the fact that personal communications devices are becoming smaller (partly for practical reasons but also for competitive and status ones), we have seen operation in the ever higher regions of the frequency spectrum. It seems that this insatiable quest for bandwidth has spelt the end of operation in the lower regions of the spectrum. Or has it? This research takes an alternative look at using the latest technology coupled with seemingly forgotten communication techniques, in the light of enhancing the lives of rural dwellers.

The task of providing communications infrastructure for rural populations poses several problems. The first is that generally (but especially in South Africa), the distance over which the communications occurs is very large. This makes for very expensive wireless infrastructure as either satellite or long haul repeater systems need to be employed. The problem is aggravated by the fact that rural communities are in no position to pay for such facilities. Another problem is the low population densities. Generally, building such infrastructure for such low densities is just not economically viable. A solution would thus require low cost equipment connecting widely separated communities together and to the outside world. Recently, there has been explosive growth in the area of telecommunications, at various levels. Rapid advances in technology have enabled people to come close to the telecommunications industries' "holy grail" : Ubiquitous availability or universally available dial tone.

These solutions are ideal for countries with developed infrastructure, customers who have a need for and are able to pay for such services and whose basic telecommunications needs are already fulfilled. This is not the case in Africa. Those fortunate enough to have service have become used to telecommunications infrastructure that is both costly and low quality. The facts are that 12% of the worlds population resides in Africa with a tele-density ratio of 1.74%. 75% of all Africans have never placed or received a phone call [10]. There are more phone lines in New York or Tokyo than there are in the whole of Africa [11]. Studies by the International Telecommunications Union and World Bank show that telecommunications are a vital driver of economical growth and socio-economic upliftment as well as showing a direct positive correlation between telephone penetration and per capita growth [12].

In South Africa the situation is marginally better with tele-density at approximately 9.5% due to the well-developed urban infrastructure. Rural areas however, still remain vastly under or non-serviced. As part of its exclusivity agreement, Telkom (South Africa's parastatal telco) is required to role out an additional 2.8 million lines at a cost of over R40 Billion.

It is clear then that the high-speed Mbps offerings of vendors are not what is needed in a rural environment. Basic communications, be it voice, text or data, is. Various methods, non conventional or otherwise, will need to be employed to overcome the enormous task of providing universal service to all South Africans.

1.2 Research Objectives

This research effort concentrates on both the theoretical and practical aspects of designing and implementing a relatively low cost communications system that would enable real time and non real time communication. The investigation concentrates on technologies that overcome limitations of both economies of scale and environmental obstacles.

The broad research objectives were to:

- Investigate current South Africa Telecommunications needs
- Identify obstacles to implementing rural access technologies
- Devise solutions which are innovative, readily available and relatively low cost to implement

The specific research objectives were to:

- Investigate wideband modulation techniques and HF radio
- Construct and implement an HF channel model
- Investigate various error correction schemes
- Simulate and evaluate a proposed system
- Investigate practical aspects of a real system
- Provide recommendations for future implementation

The proposed solution involves the utilisation of a combination of two technologies. HF band (2-30 MHz) communication provides the long distance repeater-less links and Direct Sequence Spread Spectrum (DS-SS) provides:

- Suppression of multipath components and other artifacts introduced by the ionosphere and other narrowband interference,
- Simultaneous operation with other narrowband users in the HF Band and
- Multiple access.

Practically, this involves low cost PC's as user interfaces connected to readily available radio equipment. These stations would enable communities, who live tens to hundreds of kilometres from nearby telecommunications infrastructure, to communicate with a central peering point or directly with each other. The peering point would then connect them to other services such as the PSTN or the Internet.

1.3 Research Methodology

Research was conducted in three major areas:

1. Modelling and simulation: This area focuses on designing a model of ionospheric propagation by directly mimicking its effect on radio signals. An investigation is then made into the effects of spreading in an HF environment. Also simulated are various error correction techniques.

2. Software and Graphical User Interface: This area focuses on developing a user interface for terminals used in the CDMA-HF radio link. This consists of an application, which allows various communications methods to be used, as well as providing some functionality of maintaining the link quality.

3. Hardware and field measurements: This area investigates what hardware is currently available to practically implement the proposed system. Research is conducted by taking various field measurements and inferring possible link quality and performance levels by using the results obtained from the simulations and data from the literature survey.

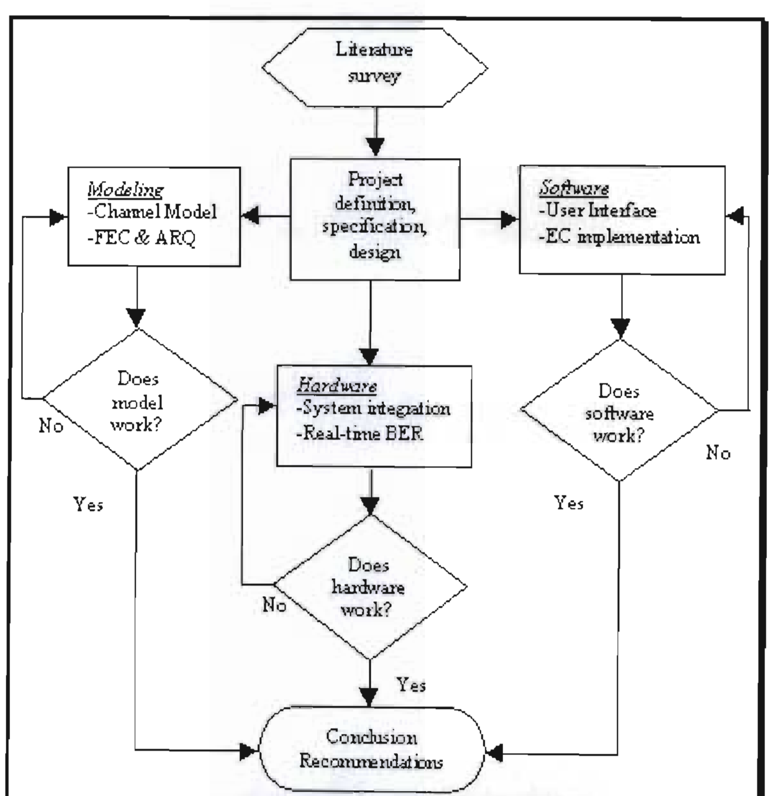


Figure 1.0 Research Methodology

1.4 Dissertation Structure

The dissertation is structured as follows:

Flexible Radio Channels

This chapter discusses the various radio channels that are available when considering a system for rural access networks. Their various strengths and weaknesses are weighed up leading to a proposal for a CDMA-HF radio link.

Multiple Access techniques

This chapter discuses and compares various multiple access techniques such as FDMA, TDMA and CDMA and compares them both qualitatively and quantitatively. These are then compared in a rural environment as opposed to an urban one.

The CDMA-HF Radio link

This chapter introduces the concept of a CDMA-HF radio link. Topics that are discussed include error correction, system performance and finally, the notion of an HF channel model is introduced.

Simulation of the CDMA-HF Radio System

This chapter deals with the construction, simulation and validation of an HF channel model. The model is then used in a system that is first unspread and uncoded, to a fully spread and coded system.

Implementation of the CDMA-HF Radio System

This chapter deals with the practical aspects of implementing a CDMA-HF radio rural network. Topics discussed include hardware design an implementation as well as the development of a software package that serves as the GUI for users.

2. Flexible Radio Channels

Radio wave transmission between two terrestrial locations can be classified into two main categories, Line-Of-Sight (LOS) and HF Radio. The LOS mode propagates by direct electromagnetic wave expansion from the transmitting antenna. Radio waves propagated via HF radio however, consist of a combination of a ground wave component and a sky wave component and where the ground wave can be further classified into a direct and a surface wave. The other method of communicating between two points is via satellite, where the satellite acts as a reflector, relaying the signal from one point to the other. Attention now turns to these various forms of propagation methods, highlighting the advantages and disadvantages in the light of their use in a rural setting.

2.1 Line-of-Sight UHF/VHF Radio Links

Radio waves in the upper regions of the frequency spectrum behave like light. Thus, wireless data links that operate in the UHF\VHF and higher part of the radio spectrum rely on Line-Of-Sight (LOS) for operation. RF radio links are used in diverse applications such as television and radio broadcast, to mobile telephony. In the past, most terrestrial links have used analogue FM for its excellent noise suppression capabilities [15]. Nowadays, many still use FM but the baseband signal has increasingly taken on a digital format. Digital technology has made it possible for designers of communications systems, to utilise developments in digital signal processing. This has enabled the use of tools such as powerful error correction codes, mass data storage, multiplexing, and other information rate-change techniques to provide efficient communications infrastructure. Transmission over terrestrial distances requires the use of modulated carriers and mostly M-ary PSK and QAM modulation techniques are used [15]. There are several other factors to consider when designing an RF link.

To transmit a carrier signal between two locations in a point-to-point link, involves two antennas. The height, type and efficiency of the antennas will determine the distance between transmitting and receiving stations. As already mentioned, UHF propagation and light are alike in many ways. They each tend to travel in LOS but because the RF signal is at a lower frequency, it bends slightly and will go a bit farther than light. This increase at approximately 900 MHz, a frequency commonly used in RF links, is about 1.18 times the line-of-sight distance. Also, due to the curvature of the earth, the line-of-sight distance will vary with the height of the antenna Above Ground Level (AGL) and frequency, as RF signals 'bend' less at higher frequencies. At 1.83 m AGL, line-of-sight is about 4.8 km. At 4 m AGL, it is about 6.3 km. At 7.6 m, the line-of-sight is about 9.65 km but the UHF wireless distance is around 11.3 km. Assuming a constant frequency, the relationship between antenna height and distance is

$$D = 1.61\sqrt{0.61H} \qquad (2.00)$$

where H is in metres

D is in kilometres.

Another consideration with RF links is in keeping the lower 0.6 Fresnel Zone clear of any obstructions to prevent echoes or multipath from reducing the received signal. The lower part of the 0.6 Fresnel Zone is like a "sag" or widening of the radio beam at the middle of the path. Its width (at 900 MHz) is about 13.4 m at 4 km for an 8 km link and about 22 m at 11.5 km for a 23 km link. The lower 0.6 Fresnel Zone, as well as the radio centre line between the antennas, must be clear of all obstacles for optimum performance. The antenna AGL is about 30 m at each end of a 23 km unobstructed path over flat ground. The lower 1st Fresnel Zone distance is given by the expression

$$F_1 = 72.1\sqrt{\frac{d^2}{2fd}} \qquad (2.01)$$

where F_1 is the 1st Fresnel Zone radius in feet,

d is the midway distance in miles, and

f is the frequency in GHz.

Wireless signals get weaker as the distance increases. In the 900 MHz region, the attenuation is -96 dB for the first 1.6 km and increases by -6 dB each time the distance doubles. At 3.2 km it would equal approximately -102 dB, 6.4 km it would equal -108 dB, 12.8 km it would equal about -114 dBm, and 25.7 km would equal -120 dBm [18]. This is important in determining how strong the received signal will be and if a proposed link is practical.

Many antennas exist for use in the UHV/VHF bands and a very common external array is a 50-ohm, Yagi-Uda. The Yagi is a directional antenna that has a relatively wide transmitting and receiving angle. It can be mounted for either vertical or horizontal polarisation and is offered with several gain figures. The higher the gain, the narrower the angle. Another popular antenna is a panel type used mostly with cellular systems. Regulatory authorities, such as the Federal Communications Commission (FCC) and the South African Telecommunications Regulation Authority (SATRA), impose a gain limit to reduce Electromagnetic Interference (EMI). In addition, a power limit termed Effective Isotropic Radiated Power (EIRP) is also specified, with the value being +36 dBm or about 4 watts. It means the total power level in dBm from the transmitter (+30 dBm) plus the antenna gain (+6 dBm) must not exceed +36 dBm. Therefore, the antenna gain figure is important in meeting the EIRP technical requirement.

Link Budget example:

Suppose that it is necessary to send data 8 km over "flat" terrain. Assume that the transmitter's RF power is +30 dBm and that the receive and transmit antennas each have a gain of +6 dBi. The product of the transmit power and antenna gains are therefore +42 dB. The losses are calculated as follows: 8 km of space loss is approximately -110 dB with 2.7 dB of coaxial cable losses (18.3 m) at each of the transit and received ends, with the total loss being approximately -116 dB. Therefore, -116 dB and +42 dB combined will equal -74 dB which is the calculated received signal strength. Assuming the receiver has a receiver sensitivity of –80 dBm, there is therefore 6 dB of fade margin. The calculation is as follows

$$[f_d] = [P_T] + [G_T] + [G_R] + [L_d] + [L_o] + [P_r] \tag{2.02}$$

$$6 = 30+6+6-110-2.7+76.7$$

where : $[f_d]$ is the fade margin

$[P_T]$ is the transmitter power

$[G_T]$ is the transmitter antenna gain

$[G_R]$ is the receiver antenna gain

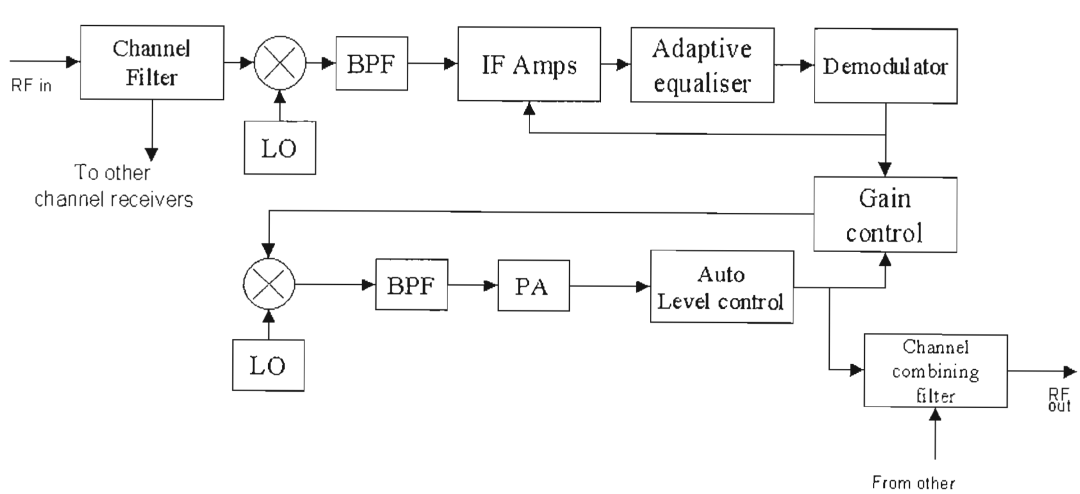
$[L_d]$ is the free space loss

$[L_o]$ is the cable loss

$[P_r]$ is the received power

Case example: AT&T DR 6-30

A representative terrestrial digital radio system is the AT&T DR 6-30 repeater. It operates within a 6 GHz common carrier frequency allocation and 30 MHz 3dB bandwidth, with 16 QAM modulation. Receiver sensitivity is –78 dBm for a BER of 10^{-3} but in practice this is normally kept at a level which gives a performance of less than 10^{-6} by operating the link at a higher receive power. Other parameters include a transmitter amplifier with a 30 dB gain, which is 2.5 dB backed off from saturation. Linearity throughout the whole system is carefully controlled due to the QAM modulation. Shown in Figure 2.0 is a block diagram of a DR 6-30.



Paul H Young, '*Electronic Communication Techniques*', Maxwell Macmillan, 1991

Figure 2.0 DR 6-30 Repeater System

UHV\VHF and microwave terrestrial links are increasingly playing a larger part in modern day communication networks. Problems associated with traditional wire line solutions such as theft and speed of deployment have forced Telco's to use a wireless approach in many areas. Applications such as last mile fixed cellular systems and mobile telephony are all popular examples of RF use in this band. There are however, problems associated with this approach for a true rural environment. The main stumbling blocks are the LOS requirement and economies of scale. Rural areas are generally not flat in South Africa.

Therefore, to cover the great distances between communities, a great number of repeaters would be needed, increasing costs on both capital and maintenance levels. Also, low population densities would make such a system economically unviable.

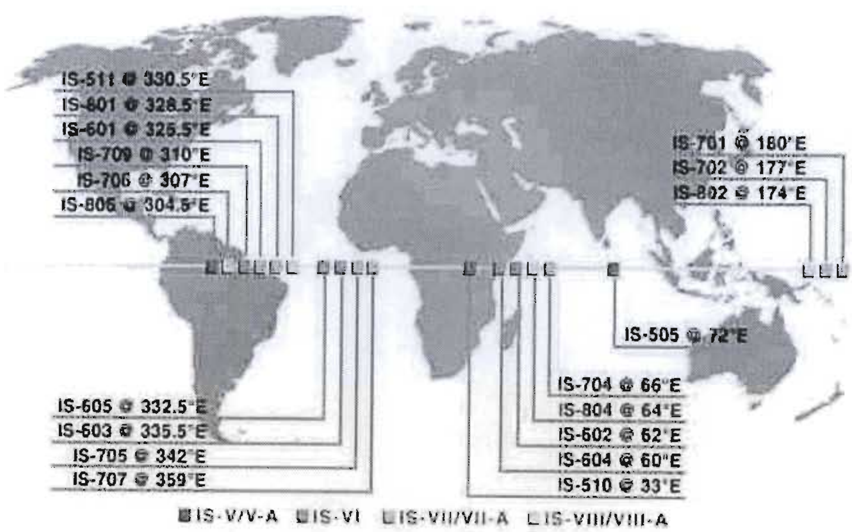
2.2 Satellite Links

Traditionally satellites have been used extensively to cover areas of the earth, which had little or no telecommunications infrastructure. Target markets were initially geologists, healthcare workers, maritime and disaster relief organisations. Examples of these are the INTELSAT, INMARSAT , PanamSAT and other VSAT systems. These networks enable a person to make a telephone or data call almost anywhere on earth. They use suitcase size terminals or small parabolic dishes for call origination and termination. Their large form factor is mainly due to the fact that the satellites that they are communicating with are in geostationary orbit.

Case example: INTELSAT

INTELSAT is one of the older generations of satellite systems. Using 19 geostationary satellites, INTELSAT is able to offer services, which include voice, fax and data up 9600 bps. Terminals are priced at about $3000 to $5000 dollars with call charges in the range of $5 to $6 per minute. Operational frequencies are C band as well as Ku band. Antenna sizes for earth stations range from up to 9m for the C band frequencies down to about 1.8m for the Ku band. INTELSAT also targets companies and organisations with high bandwidth requirements as clients and can provide other services ranging from multimedia to video conferencing.

Other typical users include: Providers of basic long distance telephone services, airlines for transcontinental booking arrangements, international banks for credit verification and authorisation, multinational manufacturers, petroleum companies, news and financial information services to facilitate their global operations, international newspaper distributors for simultaneous remote printing of daily editions on several continents, disaster relief and health care agencies and organisations, regional economic organisations, national governments, and the United Nations missions. Shown in Figure 2.1 is a coverage map giving INTELSAT satellites and their various orbits.



Intelsat website: http://www.intelsat.com/coveragemaps/

Figure 2.1 Intelsat Orbit Map

It is clear that although, geostationary satellites could be used as temporary relief in critical areas, they are not suitable for small population, rural access. The main factor against their implementation is the large capital cost of earth station equipment and the high ongoing call costs.

Lately, there has been rapid development in the area of global mobile personal communications. Currently, several systems are set to be rolled out and in operation within the next few years, with at least two systems (Iridium and Global), being in operation already. These systems aim to use various techniques to enable the notion of universal dial tone, to become reality. Although differing slightly amongst themselves, most are based on Low Earth Orbit (LEO) satellites to provide connectivity.

The bankruptcy of both Iridium and ICO, another LEO based Global Mobile Personal Communications System, prove this a difficult business in which to be profitable.

As LEO's are not visible all of the time, sophisticated hand-over techniques are being employed to enable seamless communications. Some systems pass calls between satellites, while others pass them to local earth stations called gateways, and allow use of terrestrial based networks to enhance capacity. A brief overview of two of these new generation satellite networks follows [13]:

Orbcomm: System uses 36 little LEO's for 2-way data communications market i.e.- global pager. Frequency spectrum 137-138 MHz for the downlink and 148-150 MHz for the uplink. Terminals are small subscriber communicator units, which transmit under the control of a frequency-hopping scheme.

Globalstar: System uses 48 LEO satellites for global coverage. Services include single line residential, payphone 9600 bps data group 3 fax. Satellites orbit in 8 planes with six satellites in each plane as shown in Figure 2.2.

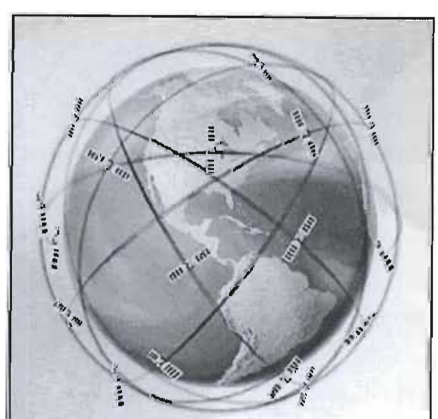


Figure 2.2 Globalstar Satellite System

CDMA is the modulation technique used. Initial cost estimates include $1200 for a fixed installation and $0.65 to $1 call charges per minute.

Iridium:

Iridium was one of two commercially operating systems. Its features included 2400 bps vocoder and data rate and group 3 fax. The system was aimed mainly at roaming mobile users who could also have taken advantage of local terrestrial networks such as GSM and cdmaOne, due to the dual nature of the handsets. Prices where approximately $1300 for handsets and $3 per minute for calls.

Other systems which shall come on line between 1999 and 2004 include Euro-African Satellite Telecommunications (EAST) and ICO. Among the broadband offerings are Teledesic (which has since merged with ICO) and Skybridge. It is clear that these systems shall be efficient and cost effective in providing global coverage for those who can afford the service. However, with the overall system capacities for each satellite network and the relatively large equipment and running costs, they are also not a viable solution for low population density rural access. Although technology is available to overcome many of the problems associated with rural telephony and limited infrastructure, affordability is an issue that is not.

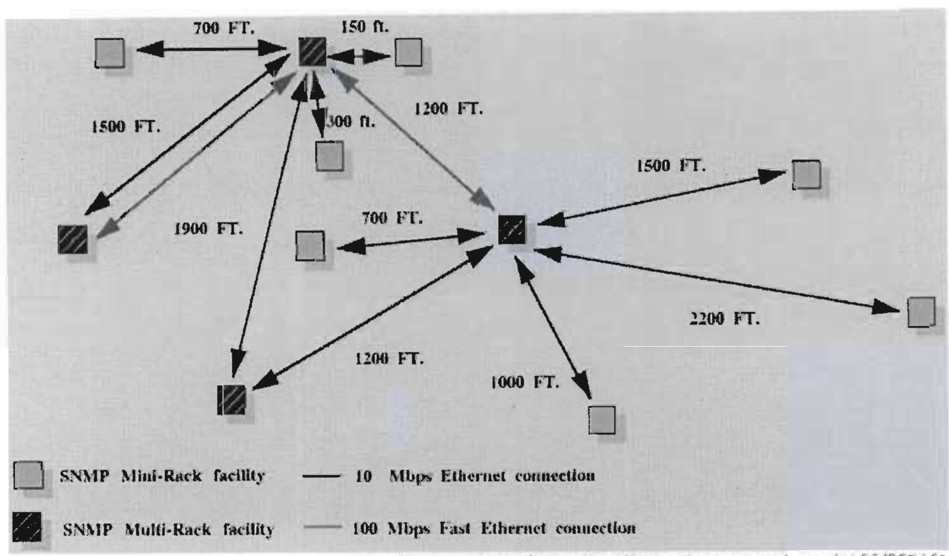
2.3 Optical links

Although optical communications are not new, recent advances in laser optics have resulted in significant advances in this field. Previous disadvantages included low quality links due to fog and rain fade and the short distances achievable. However, new higher powered lasers and improved optics have enabled the viability of optical links that can transmit over several kilometres. The two main advantages in using optics are that infrared lasers allow massive bandwidths to be achieved and also the fact that there are no interference problems normally associated with traditional RF links. These and other factors have made optical links a realistic choice for a local loop type situations. Below is a case example of one such system:

Case example: Optical network

A prison system in New York needed a network to connect the main institution to its medical facility. The bandwidth needed was high due to the fact that medical imaging records were transported over the network. A wire line solution was ruled out as this would have comprised security.

Traditional RF microwave networking was also prohibited as the facility was near an airport and local regulatory authorities denied permission on the grounds of interference. The solution used was an optical network that consisted of both 10 Mbps Ethernet as well as 100 Mbps fast Ethernet. Figure 2.3 shows the achieved network architecture.



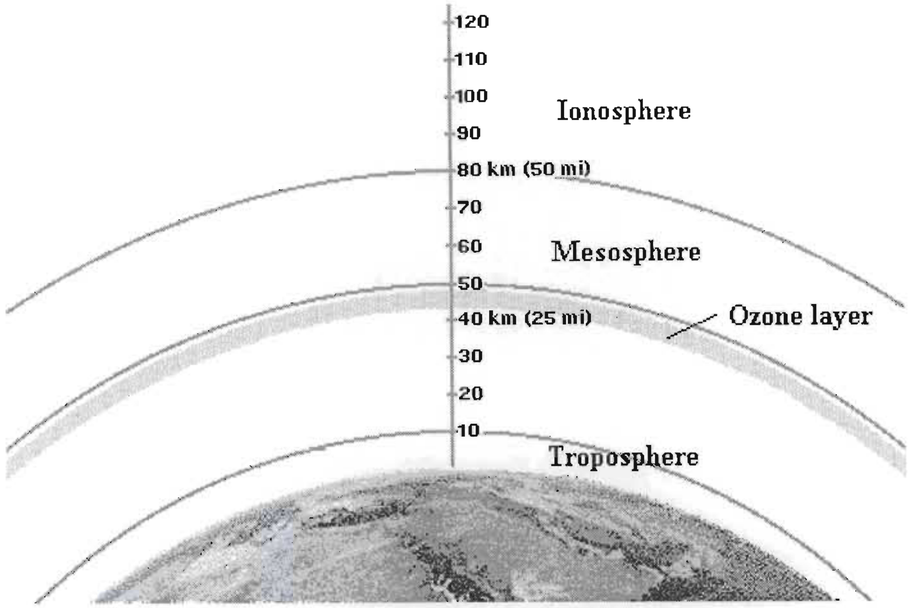
Lucent press release: http://www.lucent.com/press/1198/9821039.bla.html

Figure 2.3 Optical Network Topology

As is shown the longest link was approximately 2200 ft. although this kind of range would be acceptable for a last mile problem, it is clearly not suitable for the vast distances needed to be covered in a rural setting. Although the advantages of high bandwidth and zero interference would be welcome, the application of optical links as a backbone is not viable in a low population density rural environment.

2.4 HF Radio Links

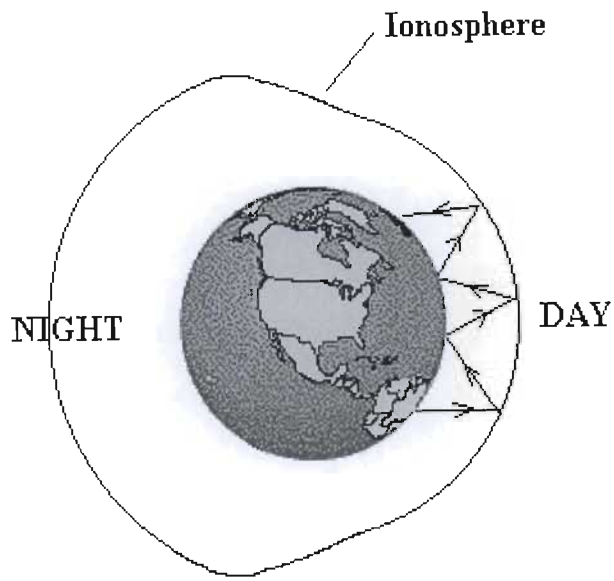
HF radio is considered to be radio communication in the frequency spectrum of 2MHz to 30 MHz. This frequency spectrum is also currently assigned to short-wave radio broadcasting. However this should not rule out investigations to determine better ways of using this spectrum. Communications in this band allows one to communicate over very large distances due to the fact that propagation is via the ionosphere [4]. The problem however, is that the ionosphere acts as a dynamic, imperfect mirror, continually distorting the transmitted signal. Thus to fully understand HF radio wave propagation, knowledge of the ionosphere is required. A brief introduction is given below [7].



Pactor-II website: www.scs.com

Figure 2.4 Atmospheric Structure

The ionosphere is that region of the earth's atmosphere in which free ions and electrons exist in sufficient abundance to affect the properties of electromagnetic waves that are propagated within and through it. Ions are produced in the atmosphere partly by cosmic rays but mostly by solar radiation. The latter include ultraviolet light, X-rays, and particle radiation (during storm periods). The several gaseous constituents of the atmosphere selectively absorb this radiation, with ion-electron pairs being produced in the process. For practical purposes, the ionosphere can usually be assumed to extend from about 50 Km to approximately 2000 Km above the earth's surface. Its structure is highly variable, and this variability is imparted onto the signals that propagate through it.



Pactor-II website, www.scs.com

Figure 2.5 Ionospheric Day to Night Transition

The ionosphere is divided into three vertical regions D, E, and F, which increase in altitude and in electron density. The D region has an altitude range from 50 to 90 km. The electron density in the region has large diurnal variations highly dependent upon the solar zenith angle. The electron density is a maximum near local noon, is higher in summer than in winter, and is lowest at night.

The E region spans the altitude range from about 90 to 130 km. The maximum density occurs near 100 km, although this height varies with local time. The diurnal and seasonal variations of electron density are similar to those of the D region. Collisions between electrons and neutral particles, while important in the E region, are not as numerous as in the D region. The E region acts principally as a reflector of HF waves, particularly during daylight hours.

Embedded within the E region is the so-called sporadic-E layer. This layer is an anomalous ionisation layer that assumes different forms that are irregular and patchy in "appearance" and has little direct bearing to solar radiation. The causes of sporadic-E ionisation are not fully understood. Short-skip openings, sometimes on an otherwise dead band, are often a result of one-hop sporadic-E ionisation.

The highest ionospheric region is termed the F region. The lower part of the F region, from 130 to 200 km, is termed the F1 region, and the part above 200 km is termed the F2 region. The F2 region is the highest ionospheric region, usually having the highest electron density, and is the region of greatest value in long-distance HF ionospheric propagation. The region exhibits large variability in both time and space in response to neutral winds and electrodynamic drifts in the presence of the earth's magnetic field. The maximum electron density generally occurs well after noon, sometimes in the evening hours. The height of the maximum ranges from 250 to 350 km at mid-latitudes to 350 to 500 km at equatorial latitudes. The F1 region, like the E region, is under strong solar control. It reaches a maximum ionisation level about one hour after local noon. At night and during the winter the F1 and F2 regions merge and are termed simply "F region". Electromagnetic waves are refracted when passing through an ionised medium, the refraction increasing with increased electron density and decreasing with increase of frequency. If the refraction is large enough, a wave reaching the ionosphere is bent back toward earth as though it had been reflected, thereby permitting reception of the wave at a large distance from the transmitter. The F2 layer is the most important in this regard because of its height and its high electron density. The maximum earth distance traversed in one F2-layer "hop" is about 4000 km. Round-the-world communication can occur via multiple hops. If the frequency is too high, the wave is not refracted sufficiently to return to earth. The maximum frequency for which a wave will propagate between two points is called the Maximum Usable Frequency (MUF). Frequencies higher than the existing MUF at any given time are not supported, no matter how much power is used.

Signals on their way to or on their way back from the F2 layer must pass through the E region of the ionosphere. The E layer is also capable of "reflecting" HF signals, and if the E layer MUF is too high, the signals to or from the F2 layer are cut off by the E layer. The E layer cut-off frequency is referred to as ECOF. Signals at frequencies below the ECOF will not pass through the E layer. Signals can propagate between two points on earth via the E layer in the same manner as they do via the F2 layer, but the maximum earth distance traversed in one E-layer hop is only about 2000 km, so a significantly greater number of hops is usually required on very long distance (DX) paths.

The D region of the ionosphere must be traversed by signals on their way to and from the F2 or E layers. Electron densities in the D region are not large enough to cause HF signals to be returned to earth, but the high collision frequency between the electrons and neutral particles in the D region gives rise to absorption of signals passing through it. The reduction of signal strength can be substantial, particularly in daytime on the lower HF frequencies. Antenna installations that provide low radiation (take-off) angles can minimise the number of hops required between two stations, thereby reducing the number of passes through the D region and the amount of signal absorption.

Electron density in the ionosphere increases with increased solar activity. Therefore, MUFs and signal absorption both increase as solar activity increases. The Zurich smoothed mean sunspot number has been used extensively as an index of solar activity and the one with which propagation data has been correlated over the years. Therefore, most propagation prediction models require that the user specify the sunspot number to be used in making a prediction.

Ionospheric propagation is susceptible to several kinds of short-term disturbance that are usually associated with solar flares. Depending upon the nature of the disturbance, they are called sudden ionospheric disturbances, polar cap absorption events, or ionospheric storms. These disturbances upset the electron configuration in the ionosphere, and consequently affect propagation. Propagation is also affected by changes in the earth's magnetic field. The magnetic field is constantly fluctuating, but the fluctuation occurs over much wider limits during magnetic storms that accompany ionospheric storms. Ionospheric and magnetic storms are also often accompanied by visible aurora. Except for the tendency of these disturbances to recur in synchronism with the 27-day rotation period of the sun, they are difficult to predict and to quantify.

Other problems that are associated with HF radio propagation include the fact that not only one signal reaches the receiver but a combination which is a result of the various paths taken by the transmitted signal [5].

Three major multipath effects can be identified, time dispersion, frequency dispersion and selective fading. They are strong if the frequency used is much below the MUF, and if the distance is long. For example, a single hop path on the 20m band, normally does not suffer from severe multipath effects. However, a DX link on the 80m band at night often provides considerably strong multipath problems [6]. Thus successful operation in the HF band depends on numerous factors.

Given in table 2.0 are the approximate values for a typical HF channel [16].

Characteristic	Unit	Min	Typical	Max
Total differential path delay	ms	0.1	2	8
Fade rate	Hz	0.01	1	10
Time spread	µs	0	40	500
Frequency shift	Hz	0	0.1	1
Number of significant paths	--	1	2	5

Table 2.0 Typical HF Channel Parameters

Total differential path delay is defined as the time difference encountered in traversing two different ionospheric routes. Time spread is the value of the delay of any reflections, resulting in intersymbol interference. Current narrowband techniques using digital modes in the HF band require the use of DQPSK modulation within a 500 Hz Channel. These systems are thus limited to about 800 bps throughput [6].

Several forms of RF communications techniques have been presented here. UHF/VHF has the ability to serve customers, in a several kilometre radius. Satellite has the ability to operate over a global scale. Optical networks have the ability to supply almost unlimited bandwidth. In a perfect scenario, all these attributes could be offered from the same service, similar to what Universal Mobile Telephone Service (UMTS) hopes to deliver. However, the reality of economics and population diversity dictate that for the niche application of providing basic communication facilities to rural inhabitants, HF radio, with its free-to-air interface, is a good candidate.

3. Multiple Access Techniques

Radio spectrum has become a resource that needs to be managed, just like any other. It is thus imperative that it be used in the most efficient way possible. Although various definitions of spectrum efficiency exist, and shall be developed throughout this chapter, here it is loosely defined as the maximum number of users using the least amount of spectrum at some accepted level of quality. The way in which multiple users share this spectrum is paramount. The object of a multiple access technique therefore, is to allow as many users as possible to share a limited amount of frequency spectrum. As more than one user is using the same resource at the same time, there must exist some mechanism by which they are differentiated. There are three basic ways in which users can be separated. They are: In frequency, time and space or code. Conceptually these schemes operate as follows. In Frequency Division Multiple Access (FDMA) users are allowed to use a portion of the spectrum all of the time.

In a Time Division Multiple Access (TDMA) scheme, users are allowed to use the entire allocated spectrum, for some defined time interval. Finally, Code Division Multiple Access (CDMA) users are allowed to use the entire allocated spectrum, all at the same time. These three multiple access techniques are shown in Figure 3.0.

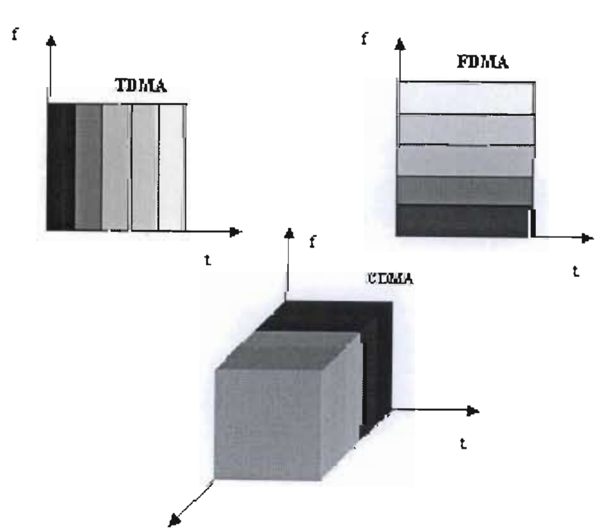


Figure 3.0 Forms of Multiple Access

3.1 Frequency Division Multiple Access (FDMA)

FDMA was one the earliest forms of multiple access techniques to be used, and it is the simplest to implement. Usually, an FDMA system can be implemented in two ways.

Multiple Channel Per Carrier (MCPC) is a form of FDMA in which a number of channels are frequency-division multiplexed at baseband and the resulting baseband signal modulates an RF carrier. Single Channel Per Carrier (SCPC) on the other hand, operates by having each channel modulate a separate carrier and the resulting modulated carriers are then summed.

Older forms of cellular systems such as Advanced Mobile Phone Service (AMPS), use FDMA as their multiple access technique. Disadvantages with using an FDMA system include the fact that in order to avoid co-channel interference from other users, gaurdbands between users are needed. Figure 3.1 shows the spectral arrangement.

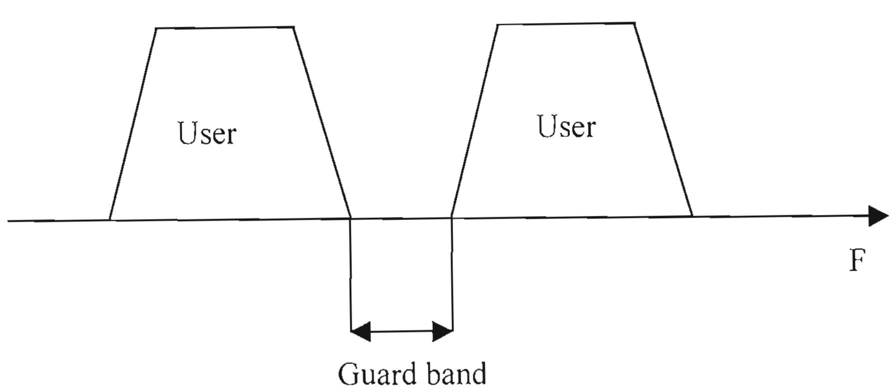


Figure 3.1 Frequency Division Multiple Access

For N users in a given frequency segment, N-1 gaurdbands are needed. This leads to a very inefficient use of the frequency spectrum. In a cellular type system where multiple users are accessing a central point ie. Point-to multipoint for the forward link and multipoint-to-point for the reverse link, the numbers of guard bands are equal to the number of users minus one. The width of the gaurdbands is dependent on, among other things, the maximum allowable co-channel interference.

For N users, the throughput of an FDMA system can be expressed as [14]

$$\eta_{FDMA} = \frac{(B_{channel} - \sum B_{guarbands}) \times \log_2(1 + SNR(N))}{B_{channel} \times \log_2(1 + SNR(1))} \tag{3.00}$$

Where : $B_{guardband}$ is the guardband bandwidth

$B_{channel}$ is the channel bandwidth

SNR [N] is the signal-to-noise ratio of the N_{th} channel

In a cellular system, capacity for an FDMA system can be given as [19]

$$Capacity\ (AMPS) = \frac{1}{21}\ Bit/Sec/Hz/Cell \tag{3.01}$$

assuming voice activity factor of $\frac{1}{2}$ and 10 Kbps voice quality.

3.2 Time Division Multiple Access (TDMA)

TDMA is a multiple access scheme that uses time as a means of separating co-users. In a TDMA system a certain portion of the spectrum is used in a time-multiplexed mode, i.e. each user gets to use the full spectrum bandwidth but only for a certain time. This is achieved by bursting each user's data, one after each other. To prevent overlapping of bursts at the receiver, guard times are introduced. These guard times reduce overall efficiency as they reduce system throughput. A typical TDMA structure is composed of a frame. Each frame consists of a reference burst and then each user's burst, separated by guard times. Each user burst consists of a preamble and the actual data as shown in figure 3.2 [15].

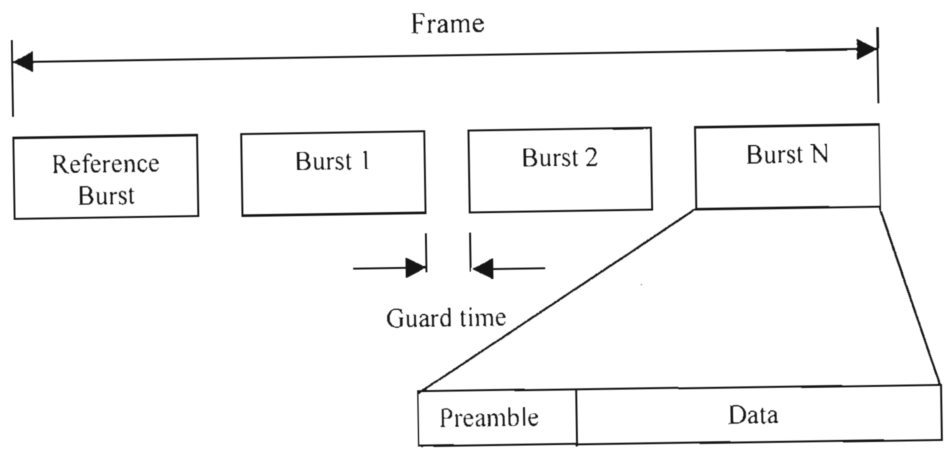


Figure 3.2 Time Division Multiple Access

The throughput of a TDMA system can be expressed as [14]:

$$\eta_{TDMA} = 1 - \sum \frac{t_{gaurd}}{t_{data}} \qquad (3.02)$$

and the capacity for a TDMA cellular system [19] is given by:

$$Capacity\ (GSM\text{-}TDMA) = \frac{1}{10}\ Bit/Sec/Hz/Cell \qquad (3.03)$$

assuming a reuse factor of $\frac{1}{4}$.

Again it can be observed that a TDMA system is limited by attempts to minimise co-channel interference. The more users using the system at any given time, the more guard times needed. Another disadvantage is that if a particular user does not use the time slot allocated to it, it becomes a wasted resource.

3.3 Code Division Multiple Access (CDMA)

The philosophy of using a multiple access technique like CDMA, which at first seems to waste resources, is sometimes confusing . The following three points attempt to summarise the reasoning behind this philosophy:

"1. Never discard information prematurely that may be useful in making a decision until after all decisions related to that information have been completed.
2. Completely separate techniques for digital source compression from those for channel transmission even though the first removes redundancy and the second inserts it.
3. In the presence of interference or jamming, intentional or otherwise, the communicator, through signal processing at both transmitter and receiver, can ensure that the performance degradation due to the interference will be no worse than that caused by Gaussian noise at equivalent power levels. This implies that the jammer's optimal strategy is to produce Gaussian noise interference. Against such interference, the communicator's best waveform should statistically appear as Gaussian noise. Thus the 'minimax' solution to the contest is that signals and interference should all appear as noise which is as wideband as possible. This is a particularly satisfying solution when, as we shall see, one user's signal is another user's interference."

Andrew J. Viterbi on Claude Shannon's channel capacity theorem

The above statement was made by Andrew J. Viterbi, one of CDMA's pioneers. It attempts to explain the seemingly absurd notion of using more spectrum then is required to form a more robust, more efficient modulation technique. CDMA is a multiple access scheme that uses spread spectrum as the choice of modulation. The origins of this wideband modulation scheme is in the military where it was used as a covert communications tool.

To introduce the concept, a definition of spread spectrum is given below:

> "Spread spectrum is a means of transmission in which the signal occupies a bandwidth in excess of the minimum necessary to send the information; the band spread is accomplished by means of a code which is independent of the data, and a synchronised reception with the code at the receiver is used for despreading and subsequent data recovery" [1]

Although this use of the frequency spectrum may seem very wasteful in the light of the fact that radio spectrum is a very scarce and valuable commodity, the benefits of spreading often outweigh it. One of the benefits are its handling of multiple access. In TDMA and FDMA systems, a user either has the whole frequency slot for some of the time (TDMA) or some of the frequency for all time (FDMA). In a CDMA system, the user has possession of the whole frequency slot all of the time. This gives some very useful practical benefits, which include:

- Antijamming,
- Anti-interference,
- Low probability of intercept,
- High resolution ranging, and
- Protection against multipath.

This research focuses mainly on a spread spectrum signal's ability to resist multipath in an HF radio environment and to cope with (and coexist with) interference caused by other narrowband users and physical phenomenon. In addition, the focus is mainly on the Direct Sequence (DS) form of spread spectrum as opposed to Frequency Hopping (FH) or hybrid variations thereof. Direct Sequence Spread Spectrum (DSSS) involves modulating an already modulated (normally phase shift keyed) signal, by a high rate pseudo random binary sequence, normally called a PN code. The following block diagram shows a basic DSSS system.

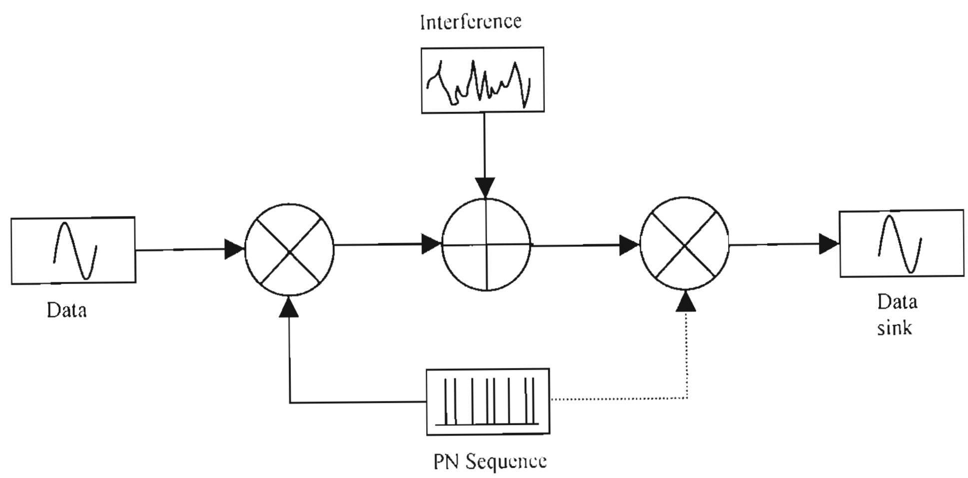


Figure 3.3 Basic Direct Sequence Spread Spectrum System

The effect of this is to spread the bandwidth of the baseband signal by an amount that is proportional to the rate of the PN sequence. The diversity achieved in spreading also enables a DS-SS system to serve as a multiple access technique. Network resources are shared by assigning each user a unique PN code. Other users and multipath signals appear as an increase in the noise floor and a subsequent increase in Bit Error Rate. CDMA is thus referred to as a soft limited multiple access scheme as there is a trade off between the number of users and the acceptable bit error performance.

3.3.1 Process Gain

It is important to keep in mind the purpose of spreading within the context of the current research topic, i.e. An attempt is made to distribute a low dimensional signal (the data) in a high dimensional environment so that an interfering signal with fixed power has to either spread that power over all parts of the spectrum occupied by the data, thereby introducing only a small amount of interference per unit of spectrum or concentrate all the power in one part of the spectrum, leaving other parts interference free [1].

Consider a transmitter and receiver pair using a set of M signalling waveforms i.e.

$$S_i(t),\ 0 \le t \le T\ ;1 \le i \le M$$

Given that the transmitter chooses one of the waveforms in the signal space every T seconds, a bitrate of

$$\frac{\log_2 M}{T} \; bps \tag{3.04}$$

is achieved.

The receiver R, will thus receive a signal

$$r(t) = s_j(t) + n_w(t) \tag{3.05}$$

over [0,T] where $n_w(t)$ is Additive White Gaussian Noise (AWGN) with a two sided PSD of $\frac{\eta_o}{2}$. It can be shown [2] that the signal can be completely described by a linear combination of $D \leq M$ orthogonal basis functions. It is said [1] that the above signal set is D dimensional if the minimum number of orthogonal basis functions required to describe all signals is D.

It can also be shown [2] that

$$D \approx 2B_D T \tag{3.06}$$

where B_D is the approximate occupied bandwidth.

It thus becomes obvious that for large D, little is achieved against the AWGN, which has infinite power and constant energy in every direction. This is because the performance of such a system is a function of only $\frac{\varepsilon_b}{\eta_o}$, where E_b is the signal power and η_o is the noise power. Therefore spreading will not increase the performance of the receiver as the AWGN is equally spread. The situation is different when one considers interference, which has finite power.

Consider now a signal $S_i(t)$ which uses D orthogonal signals imbedded in an n dimensional space i.e. spread by some amount n

$$S_i(t) = \sum_{k=1}^{n} S_{ik}\phi_k(t) \quad 1 \leq i \leq D;\ 0 \leq t \leq T \tag{3.07}$$

where

$$S_{ik} = \int_0^T S_i(t)\phi_k(t)dt \tag{3.08}$$

and $\{\phi_k(t); 1 \leq k \leq n\}$ is an orthonormal basis spanning the space. The energy of the signal can be computed by taking the integral of the mean squared value of the signal over one period, i.e.

$$\int_0^T \overline{S_i^2(t)}d(t) = \sum_{k=1}^{n} \overline{S_{ik}^2} = E_s \qquad 1 \leq i \leq D \tag{3.09}$$

It is also assumed for analysis purposes that the coefficients S_{ik} are independent i.e they have zero mean and correlation

$$\overline{S_{ik}S_{il}} = \frac{E_s}{n}\delta_{kl} \qquad 1 \leq i \leq D \tag{3.10}$$

Next consider the data signal being interfered with by a signal $J(t)$. The received signal will now be

$$r(t) = S_i(t) + J(t) + n_w(t) \tag{3.11}$$

where $J(t)$, defined as

$$J(t) = \sum_{k=1}^{n} J_k\phi_k(t) \qquad 0 \leq t \leq T \tag{3.12}$$

is an interferer with finite power, whose energy is given by

$$E_J = \int_0^T J^2(t)dt = \sum_{k=1}^{n} J_k^{\ 2} \tag{3.13}$$

The worst case scenario occurs when the components of E_J are such that the Signal to Noise Ratio (SNR) at the receiver is at a minimum. Assuming the receiver is a bank of correlators, the output at the *i*th correlators will be

$$U_i = \int_0^T r(t)S_i(t)dt = \sum_{k=1}^{n} (S_{ik}^{\ 2} + J_k S_{ik}) \tag{3.14}$$

if the received signal (ignoring thermal noise) is correlated with the known sequences. It can be shown [1] that

$$E(U_i \mid S_i) = E_s \qquad and \qquad \text{var}(U_i) = \frac{E_s}{nD} E_J \tag{3.15}$$

Using the definition of SNR we have the result

$$SNR = \frac{E^2(U)}{\text{var}(U)} = \frac{E_s}{E_J}\frac{n}{D} \tag{3.16}$$

This result shows that the SNR does not only depend on the data and interfering signal energies, but also on the ratio $\frac{n}{D}$. This ratio is known as the processing gain, G_p. It is the gain achieved directly from the spreading process, at the expense of utilising more bandwidth.

This can be rewritten as

$$G_p = \frac{n}{D} \approx \frac{2B_{ss}T}{2B_D T} = \frac{B_{ss}}{B_D} \tag{3.17}$$

i.e. process gain is the ratio of bandwidth after spreading to the bandwidth before spreading or the ratio of the chip rate of the P_n code to the baseband data rate. Thus the data signal has an advantage of G_p over the interferer.

This gain can be used to overcome various forms of interference, which shall be discussed in the following chapters. The following figure shows a normalised spread and unspread waveform. The PN sequence rate (chip rate) is ten times that of data sequence.

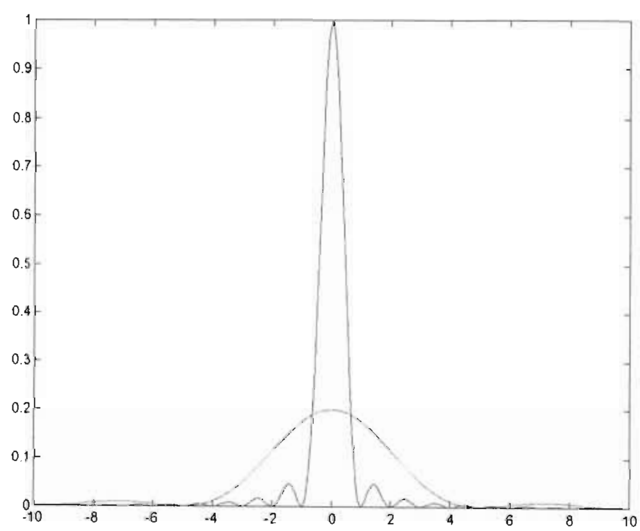


Figure 3.4 Spectrum of Direct Sequence system

3.3.2 Error Analysis

Spread spectrum systems provide no benefit against thermal noise and the probability of error (assuming BPSK modulation) is repeated here for completeness.

$$P_e = \frac{1}{2} erfc \sqrt{\frac{E_s}{\eta_o}} \qquad (3.18)$$

Shown below is a plot of the performance of the above system.

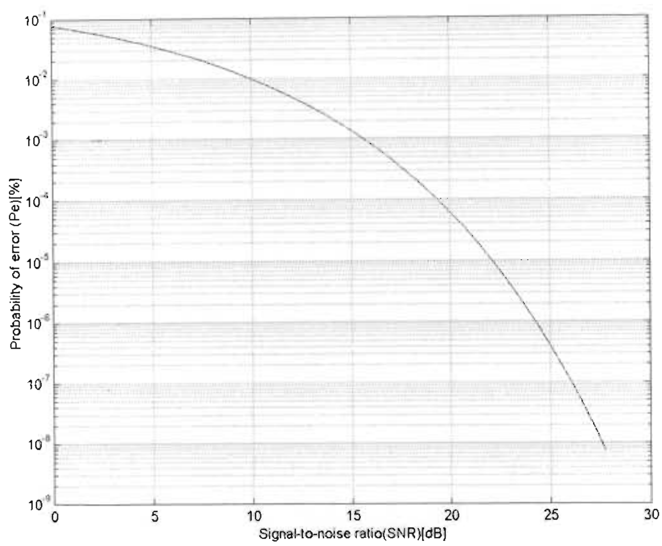


Figure 3.5 Probability of Error for BPSK

When considering error analysis for a DS-SS system, there are two main cases; that of broadband noise and that of single tone interference. The analysis for broadband noise is similar to that of the unspread signal with a slight modification. An assumption is made that the interferer occupies the entire spectrum occupied by the spread signal. Then, when the receiver despreads the signal, the interferer appears as white noise. The modified error probability expression (*Raymond L. Pickholtz* et al.) now becomes

$$P_e = \frac{1}{2} erfc\left(\sqrt{\frac{2E_s}{\eta_o + \eta_{oJ}}}\right), \qquad \eta_{oJ} = \frac{P_J}{2f_c} \qquad (3.19)$$

where the η_{oJ} is the interferer power density.

The other general case is when the interferer is a tone. It can be shown [3] that the error probability can be expressed as

$$P_e = \frac{1}{2} erfc\sqrt{2\left(\frac{P_s}{P_J}\right)G_P} \qquad (3.20)$$

All the cases shown here represent isolated forms of interference. Real world situations will dictate that the true P_e is some function of all of them.

3.3.3 PN Sequences

It is beyond the scope of this text for an in-depth analysis of PN sequences. Only important results relevant to the research topic will be briefly presented.

The ideal sequence used for spreading would be a truly random one. However, since the receiver needs an identical replica for despreading, pseudo random codes are used. There are several properties, which these codes must posses in order for them to be useful in a DS-SS system. They are and include:

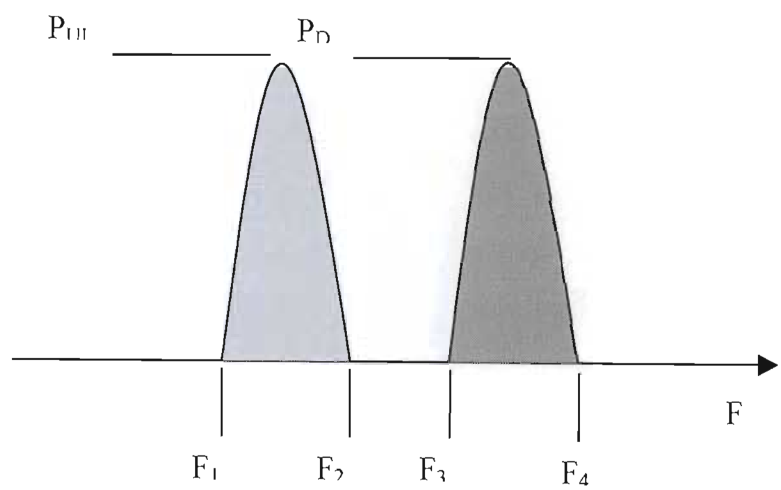
- Easily generated
- Posses randomness properties
- Have long periods
- Difficult to reconstruct from code fragments
- Have high autocorrelation and low crosscorrelation

Examples of good PN codes include gold codes and walsh codes. In practice they are often formed from linear feedback shift registers (LFSR).

3.3.4 Access Methods For Rural Wireless Radio

The various multiple access schemes outlined above all have their advantages and disadvantages. In reality none is used on its own exclusively. For instance in the GSM system, which is TDMA based, and the CDMA system which is code based, both have elements of FDMA as part of their structure. For these systems however, the channel characteristics are normally defined as those in an urban environment.
Being commercial, they are therefore optimised for the maximum number of users. The multiple access scheme is chosen for its ability to maximise the number of customers.

This is not the case in a rural environment however, as the number of users and throughput are of secondary concern. The multiple access scheme chosen needs to provide protection for the transmitted signal. In light of the fact that the HF band has such severe channel characteristics, the author suggests that CDMA, together with its high immunity to narrowband interference and multipath propagation, would be a good choice for a rural access scheme. Used in conjunction with FDMA for the forward and reverse link channels, frequency usage could be as shown in Figure 3.6 :



Where : BW_{UL} = F2-F1 is the uplink bandwidth

BW_{DL}= F4-F3 is downlink bandwidth

BW_{GUARD}= F3-F2 guard bandwidth

P_{UL} uplink power level

P_{DL} downlink power level

Figure 3.6 Frequency Plan

These parameters are discussed further in chapters 4 and 6.

Finally the throughput efficiency of a CDMA system is given as [14]:

$$\eta_{CDMA} = 1 + \frac{R_c}{R_b \Gamma \left(\frac{E_b}{N_o} \right)} \tag{3.21}$$

Where Γ is spectral efficiency.

The capacity of a cellular CDMA system is given by [19]

Capacity (CDMA) = 1 Bit/Sec/Hz/Cell

4. The CDMA-HF Radio link

The concept of how small rural communities would be connected is shown in figure 4.0. Assumptions that are made are:

- Each community has a population of between 10 to 50 people making other forms of wireless multiple access techniques non economically viable.
- There is no line-of-sight between communities and between the peering point or between each other.
- The peering point has some form of telecommunications infrastructure installed already.

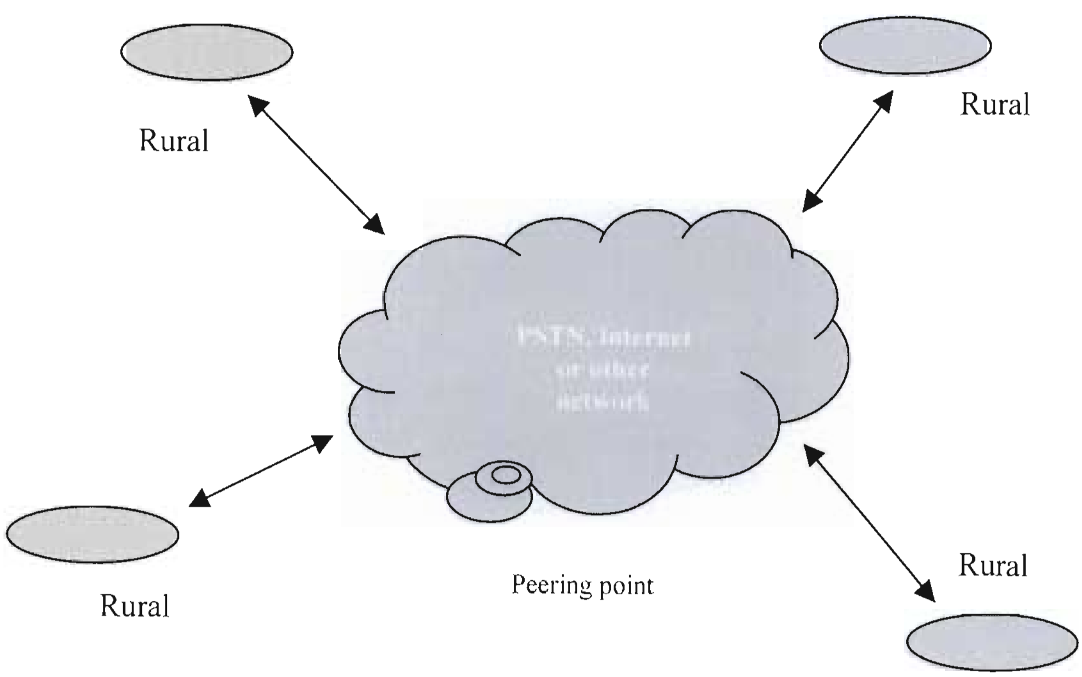


Figure 4.0 System Network Topology

In other words the system architecture would take on a fixed cellular type of network architecture.

Each rural community would posses the following equipment:

1. Low cost PC and software
2. Broadband HF antenna
3. Power amplifier module and associated electronics
4. CDMA transceiver

The system has the following configuration

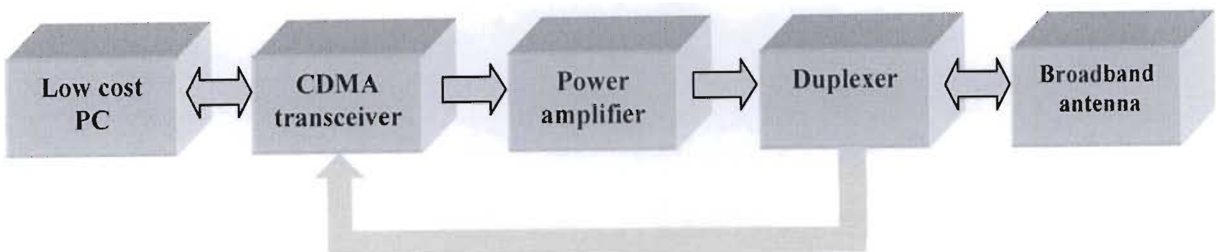


Figure 4.1 System Modules

Operation is as follows: The low cost PC acts as the Man-Machine Interface (MMI), as well as the data sink and source. It runs software, which allows various forms of communications and performs some of the error correction such as the ARQ's on the application level. Software FEC (Forward Error Correction) is limited as it adds processing delay to the system. The software also serves to control the CDMA transceiver's parameters. The CDMA transceiver performs all baseband signal processing and is responsible for pre-modulation, spreading, de-spreading and demodulation. The power amplifier module and associated hardware consists of a class A power amplifier operating in the HF band. It also contains band pass filters, which filter out-of-band noise. As only one antenna is used, a duplexer is needed to split the frequency division multiplexed forward and reverse links in order to support full duplex operation. A commercial duplexer from a supplier such as Comet can be tuned to accommodate this, although the separation of 2 MHz may need to be increased. Alternatively a duplexer can be built using bandpass and notch filters. This is difficult however, as hand built duplexers are troublesome to tune. Finally the broadband antenna covers the entire HF spectrum and is used for both transmission and reception.

The three main mode of communications that could be supported are voice, text and data.

Applications of these could are:

- Voice: Although the limited bandwidth would limit the link to one voice channel, this could be used to either communicate with neighbouring communities, or to interface with the PSTN. Fuelled by the pace of the Internet, there has been advances in signal processing and codec design resulting in bit rates for real time voice as low as 1 kbps. These codecs do not need expensive DSP hardware and can be implemented in software. An example is DSP group's True Speech™ voice codec. This codec is shipped as part of the Windows™ operating system, thus reducing costs even further. Any software application that is used on the terminals at each rural site, can take advantage of this codec by hooking it through the Audio Compression Manager (ACM).

- Text : Text based communication, although not as favourable as voice, is far less bandwidth intensive. Thus using this method over the limited bandwidth link could accommodate many more users. Applications such as distance learning, televoting, census and many others could be successfully accomplished using a simple text interface.

- Data : Although the idea of data being transmitted between rural communities may seem bizarre at first, it has great potential benefit for the upliftment of the lives of rural dwellers. For instance medical imaging for tele-medicine could be an application. Another could be remote weather forecasting and telemetry. These resources could be "bought" from the communities by interested parties to help offset the cost of the link.

4.1 Error Control Coding

It is beyond the scope of this dissertation to provide a detailed analysis of error detection/correction techniques. Only those that are pertinent to the current research topic shall be discussed. There are basically two forms of error control. Forward Error Correction (FEC) and Automatic Repeat reQuest (ARQ).

4.1.1 Automatic Repeat Request

ARQ is employed in non real time applications e.g. file transfer and text transmission where data integrity is required. Various forms, versions and standards of ARQ exist in current file transfer protocols e.g. KERMIT, XMODEM, YMODEM and ZMODEM for dial up connections and FTP for TCP\IP. Each form is designed to combat, among other things, the unique channel characteristics in which it operates. The ARQ protocol itself does not correct errors but merely detects them using block codes. It operates by sending an AcKnowledge (ACK) signal back to the receiver when an error free transmission has taken place and a Negative AcKnowledge (NAK) signal when an error is detected requiring the message to be resent.

The three basic forms of ARQ that have been investigated are: Stop and Wait (S&W) ARQ, go Back N ARQ (GBN) and Selective Repeat ARQ (SR) . The S&W method sends a message and then waits for a response and only sends the next message in the queue, once an ACK or NAK has been received. The GBN method streams message blocks without waiting. When an error is detected, it goes back to the N^{th} message where the error occurred and starts streaming again from that point in the queue, improving efficiency. Finally, the SR method just resends the message block in error when a NAK is received and continues from where it was interrupted, improving efficiency further. It can be shown [10] that the throughput efficiencies of the three methods are as follows

$$\eta_{S\&W} = \frac{k}{n}\frac{P_A}{1+\frac{T_I}{T_W}}, \quad \eta_{GBN} = \frac{1}{1+\frac{N(1-P_A)}{P_A}} \quad and \ \eta_{SR} = \frac{k}{n}P_A, \qquad (4.00)$$

where T_W is the message transmission time, T_I is the round trip time i.e. time it takes for processing by the receiver and reception of an ACK or NAK by the transmitter, (n,k) are the block code parameters and P_A is the probability that the receiver accepts the message block.

To estimate the efficiency of these ARQ algorithms, a plot of efficiency against P_A for typical HF channel values, is shown in Figure 4.2 with the corresponding parameters shown in Table 4.0. It is assumed that N is such that T_I in the S&W method is the same as the retransmission time in the GBN method.

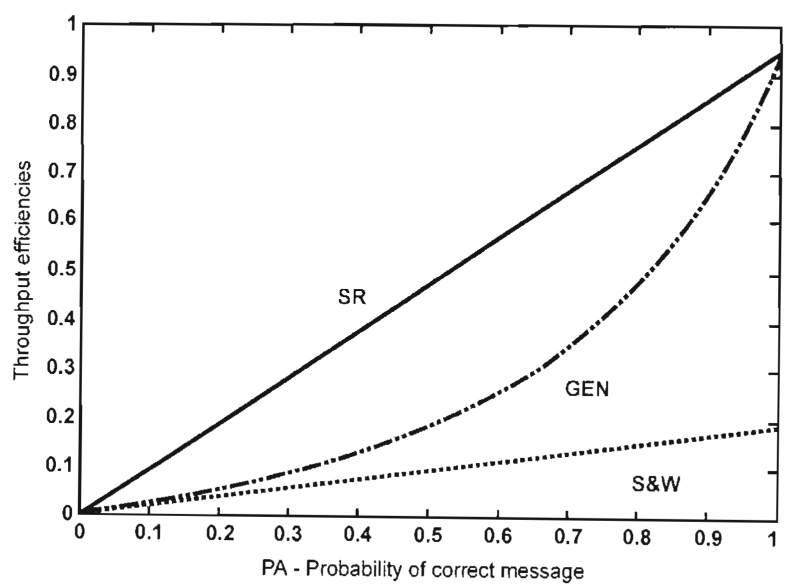


Figure 4.2 Probability of error of ARQ systems

Parameter	Value
T_W	1s
T_I	2ms
(n,k)	BCH (1023,973)

Table 4.0 ARQ Parameter Values

The S&W method shows the lowest efficiency although it is the easiest to implement. The SR method and the GBN method show comparable efficiencies for high values of P_A (>0.95). However for very low values of P_A (<0.1), the GBN and S&W provide similar results.

The SR method is most useful for midrange values (0.4 - 0.85) of P_A, where it is worth the extra implementation complexity and computational load. Its implementation in the current project is thus determined by the channel conditions at the time.

4.1.2 Forward Error Correction

The other form of error correction is *FEC* (Forward Error Correction). Unlike the ARQ method that requires a feedback channel and only detects errors, FEC detects and corrects errors. This however is at the expense of an increased bitrate (for a constant data rate) or decreased data rate (for constant bitrate). In the latter case, the effect is detrimental on the process gain by its definition. Codes that were considered are :

- The extended Golay code $X^{11}+X^9+X^7+X^6+X^5+X+1$

The extended Golay block code has been specified as a standard for Automatic Link Establishment (ALE) in High Frequency (HF) radio systems [17] . The standard specifies the use of an extended (24,12) Golay block code for forward error correction (FEC) i.e., it encodes 12 data bits to produce 24-bit code words. It is furthermore a systematic code, meaning that the 12 data bits are present in an unchanged form in the code word. The generator polynomial for this code is: $G(x) = x^{11} + x^9 + x^7 + x^6 + x^5 + x + 1$, with the corresponding generator matrix shown below. The minimum hamming distance between any two code words (the number of bits by which any pair of code words differs) is 8, giving this code the power to detect up to 7 errors in each code word (and correct none), correct up to 3 while detecting 4, or any intermediate combination.

These modes may be listed as (0,7), (1,6), (2,5), and (3,4), where the first number indicates the number of errors which may be corrected, and the second the number of errors which will be detected by each mode. In matrix formulation, the code word $\underline{u}$ corresponding to a data word $\underline{m}$, is found by the product $\underline{u} = \underline{m} \bullet G$. Note that the left half of the generator matrix G is simply a 12x12 identity matrix, I_{12}. The right half of G generates the parity check bit portion of the code word, and is commonly called the P matrix. The introduction of errors in the communication channel is typically modelled as an error vector $\underline{e}$ which is added (modulo 2) to the code word originally sent to produce a received word $\underline{r} = \underline{e} + \underline{u}$. In block codes such as this, the decoder computes a syndrome $\underline{s} = \underline{r} \bullet H^T$, where H is the parity check matrix derived from G : $H = [P^T\ I_{12}]$ and the superscript T indicating matrix transposition. This vector matrix multiplication simply computes the parity check bits for the received data bits, and adds them (modulo 2) with the received parity check bits, giving a 0 result if a valid code word is received.

$$G = \left[\begin{array}{cccccccccccc|cccccccccccc}
1&0&0&0&0&0&0&0&0&0&0&0&1&0&1&0&1&1&1&0&0&0&1&1\\
0&1&0&0&0&0&0&0&0&0&0&0&1&1&1&1&1&0&0&1&0&0&1&0\\
0&0&1&0&0&0&0&0&0&0&0&0&1&1&0&1&0&0&1&0&1&0&1&1\\
0&0&0&1&0&0&0&0&0&0&0&0&1&1&0&0&0&1&1&1&0&1&1&0\\
0&0&0&0&1&0&0&0&0&0&0&0&1&1&0&0&1&1&0&1&1&0&0&1\\
0&0&0&0&0&1&0&0&0&0&0&0&0&1&1&0&0&1&1&0&1&1&0&1\\
0&0&0&0&0&0&1&0&0&0&0&0&0&0&1&1&0&0&1&1&0&1&1&1\\
0&0&0&0&0&0&0&1&0&0&0&0&1&0&1&1&0&1&1&1&1&0&0&0\\
0&0&0&0&0&0&0&0&1&0&0&0&0&1&0&1&1&0&1&1&1&1&0&0\\
0&0&0&0&0&0&0&0&0&1&0&0&0&0&1&0&1&1&0&1&1&1&1&0\\
0&0&0&0&0&0&0&0&0&0&1&0&1&0&1&1&1&0&0&0&1&1&0&1\\
0&0&0&0&0&0&0&0&0&0&0&1&0&1&0&1&1&1&0&0&0&1&1&1
\end{array}\right]$$

Other codes include

- BCH codes, that can trade off code rate with error correction capability.
- Reed Solomon codes, which are efficient for burst errors such as those that occur during solar activity but are poor for random errors.
- Convolutional codes, that are powerful but are more complex to implement than the other codes.

4.2 System Performance

As with an communications system, there are limitations with what can and cannot be done with the above mentioned system. Physical limitations prevent high bitrates from being achieved. The main sources, as previously mentioned is the co-channel interference from other narrowband users and the ISI from the multipath distortion caused by the ionosphere. To make estimations on the system performance, the resources available need to be considered. To begin with, the HF band has approximately 28 MHz of total bandwidth available. According to the frequency plan of Figure 3.6, the following assignments could be used:

$$F1 = 2\ MHz$$
$$F2 = 15\ MHz$$
$$F3 = 17\ MHz$$
$$F4 = 30\ MHz$$
$$P_{UL} = 10\ dB\ below\ average\ narrowband\ user$$
$$P_{DL} = 10\ dB\ below\ average\ narrowband\ user$$

Which would result in the following specifications :

$$BW_{UL} = 13\ MHz$$
$$BW_{DL} = 13\ MHz$$
$$BW_{GUARD} = 2\ MHz$$

Voice performance:

The lowest real-time bitrate codec available with the Windows™ operating system is by *Lernout & Hauspie* and has the following specifications:

Type: CELP
Bitrate : 4.8kbit/s
Sampling Frequency : 8 kHz
Bits : 16 Bit, Mono

A freely available software codec has the advantage of being cheaper than dedicated DSP based hardware solutions and allows the possibility to upgrade over the link as technology in speech processing and compression advances.

Therefore, the codec requires at least 4.8 Kbps. Assuming another 2.2 Kbps for channel overhead, the total required bitrate for a single voice channel of the system is 7 Kbps. There are now two gains which can be modified to arrive at the most efficient system viz. processing gain and FEC coding gain. Considering first the processing gain without error correction

$$PG = 10\log\left(\frac{13MHz}{7Kbps}\right) = 32.7dB \qquad (4.01)$$

However, when FEC is added to the bitstream, this is reduced as the total bitrate increases due to the addition of the redundant bits. This is however, partially compensated for by the coding gain achieved.

Text :

Assuming a data rate of 5 Kbps for text and the FEC overhead combined, the process gain achievable is

$$PG = 10\log\left(\frac{13MHz}{5Kbps}\right) = 34.2dB \qquad (4.02)$$

At 5 Kbps (assuming 1 Kbps throughput) the system would be able to deliver approximately 125 characters per second, far faster than real-time typing. Thus the bitrate could be lowered even further to increase either processing gain or coding gain.

Data:

Assuming a 16 Kbps transfer rate for data including overhead,

$$PG = 10\log\left(\frac{13MHz}{16Kbps}\right) = 29.1dB \tag{4.03}$$

Assuming a throughput of 10 Kbps, a high quality 640x480 pixel VGA Joint Pictures Expert Group (JPEG) compressed 16 bit colour image, would take approximately 56 seconds to download. An average sized web page (21 KB) would take 17 seconds.

The figures presented here are only representative. They are not hard limits nor are they measured results. Although many simplifications have been made, they serve as guide as to what the system is physically capable of in terms of approximate gains and bandwidths.

4.3 Modelling the HF Channel

The main difficulties in communications over HF channels are due to the high interference levels and multipath fading. The problem is that one cannot design an optimised modem for an HF channel since the channel parameters are time varying [16]. A time and frequency sampling of an HF channel which clearly shows a second echo, is shown in Figure 4.3 [16].

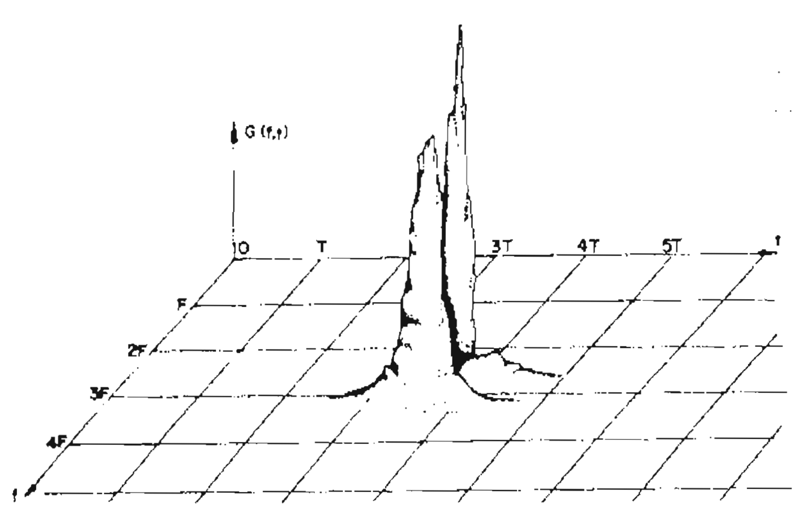


Figure 4.3 HF Channel Sounding

Attention is now turned to the modelling of co-channel interference. Often in an HF environment the performance is limited by the large amount of narrowband interfering signals from other HF users [20]. It is therefore necessary to include a statistical model of the interference when modelling the effect the HF channel on signals. Extensive measurements have bean made in Europe, of the HF occupancy. These were performed by obtaining the average power received in dBm during 1 second in steps of 1 kHz over the entire HF spectrum. On the basis of these measurements, Lacock, Gott et al. have proposed a congestion model for HF band occupancy [20]. In this model the congestion Q_k is defined as the probability that the average interference power I_k exceeds a certain threshold x. This is modelled according to the following Logit transform

$$Q_k = P_r[I_k > x] = \frac{1}{1+\exp\{-\alpha_k - B_x\}} \tag{4.04}$$

B is constant over the entire HF spectrum and shows the variance if I_k. The argument α_k is a function of the frequency, the sunspot number, time and a constant for the k^{th} ITU band. Also, $-\frac{\alpha_k}{B}$ gives the average interference power in the k^{th} ITU band. It can be shown [20] that the PDF of I_k, has the form shown in figure 4.4.

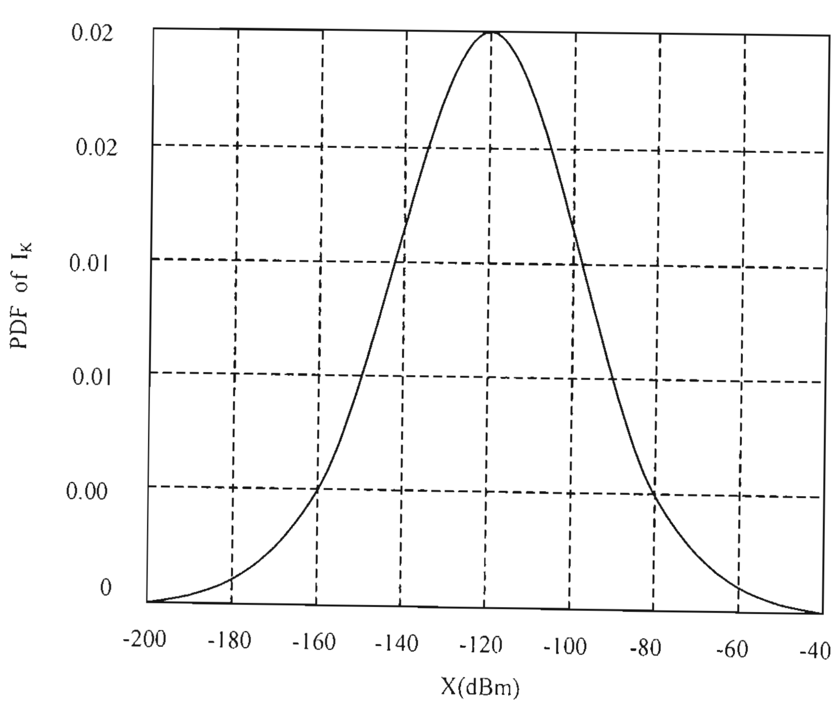


Figure 4.4 PDF of I_k

Measurements have given typical values of :

$$\alpha_k = \{-16,-9\}$$
$$B \approx -0.1$$

When considering the performance of a Direct Sequence (DS) system, the wideband Signal to Interference Ratio (SIR) needs to be taken into account. Therefore consider a stochastic variable Λ_k which represents the SIR over the whole spread bandwidth.

$$\Lambda_k = S - I_{k,tot} \tag{4.05}$$

where S is the received signal power and $I_{k,tot}$ is the total interference power received over the spread bandwidth. As $I_{k,tot}$ is computed by summing the linear powers of the L narrowband Gaussian interference processes in the spread bandwidth, the central limit theorem allows the approximation of the error term caused by the received interfering signal as being Gaussian, provided L is large. Then, given a wideband SIR Λ_k, the probability of error P_e is

$$P_e(\Lambda_k) = Q\left(\sqrt{2L \cdot 10^{\frac{\Lambda_k}{10}}}\right) \tag{4.06}$$

Where Q is the complimentary error function

$$Q(x) = \frac{1}{\sqrt{2\pi}} \int_x^{\infty} e^{-\frac{y^2}{2}} dy \tag{4.07}$$

Thus to obtain the total bit error probability, the average of P_E is taken with respect to Λ_k

$$BEP = \int_{-\infty}^{\infty} E[P_e(\Lambda_k) \mid \Lambda_k = x] \cdot f_{\Lambda_k}(x)dx$$

$$= \int_{-\infty}^{\infty} Q\left(\sqrt{2L \cdot 10^{\frac{x}{10}}}\right) \cdot f_{\Lambda_k}(x)dx \qquad (4.08)$$

Anderson et al. [20] did not obtain an explicit expression for the PDF of $f_{\Lambda k}(x)$ thus simulations were used to obtain the results of Figure 4.5

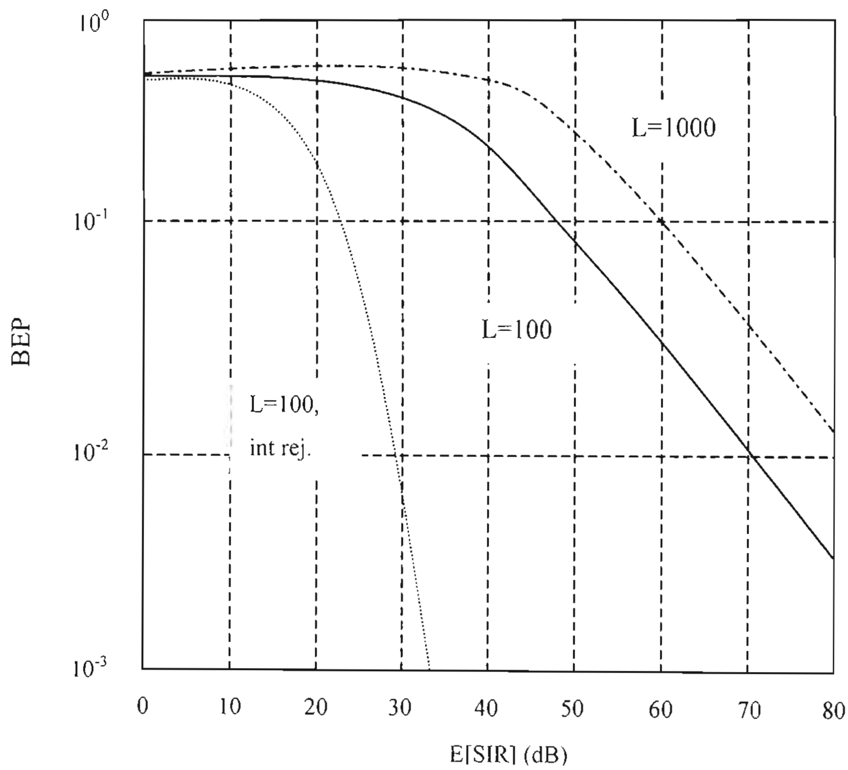


Figure 4.5 Results of Gunnar Anderson

The results in Figure 4.5 show clearly that, ignoring physical impediments to the signal such as multipath fading etc., a "straightforward" uncoded DS system, performs badly.

In fact a surprising result was that an increase in process gain resulted in poorer performance. This is due to the averaging property of DS spread spectrum. Thus, it illustrates the need to include a narrowband interference parameter in an HF model and the need for coding in a DS-HF system. Process gain is not enough.

Although multipath propagation is discussed qualitatively further on, its quantitative effects shall now be examined. Multipath propagation occurs when the electromagnetic energy carrying the modulated signal propagates along more than one path connecting the transmitter to the receiver [21]. This phenomenon affects the signal in two ways which is characterised by the delay spread and the doppler frequency spread. When an ideal impulse is transmitted through a multipath channel, the individual paths can be observed by the echoes of the original signals which have a certain delay and attenuation associated with them.

To quantify these delays with a parameter the notion of a *delay spread* is introduced which is defined as the largest of all the delays. When the bandwidth occupied by the transmitted signal is larger than the inverse of the delays, the receiver is able to resolve individual signals. However, when this is not the case, the received signals overlap causing Inter-Symbol Interference (ISI). The two effects that give rise to these delays are *time dispersion* and *frequency selective fading* [21]. If the bandwidth of the transmitted signal is B_x and is increased until distortion becomes unacceptable, the point at which this occurs is termed the coherence bandwidth of the channel B_c, which is defined as the inverse of the delay spread, i.e. it is the frequency separation at which two components undergo independent attenuations [21]. Thus it can be stated that if the transmitted signal bandwidth, B_x is far greater than the coherence bandwidth of the channel B_c, then it is subject to frequency selective fading.

The other distortion that is experienced by a signal transmitted in a multipath environment is the doppler-frequency spread. If the transmitter-receiver pair are in motion with respect to each other the signal suffers a doppler shift. The same results, as in an HF environment, if the object off which the signal reflects is in motion. When doppler shift occurs simultaneously with multipath propagation, the result is frequency dispersion and time selective fading. Frequency dispersion is the increase in bandwidth required by the transmitted signal.

The term doppler spread is given to the parameter representing the largest frequency shift of various paths in a multipath system. If the period of a transmitted signal is relatively long, there is no time selectivity. As the period becomes smaller and hence by definition the spectrum occupied larger, the echoes caused by the doppler effect begin to overlap causing distortion as was the case for ISI. The channel is then said to be time selective. The time at which the signal's period, T_x, becomes significant is called the coherence time T_c.

A multipath channel can thus be characterised by the previously mentioned parameters. This is shown diagrammatically in Figure 4.6 [21]

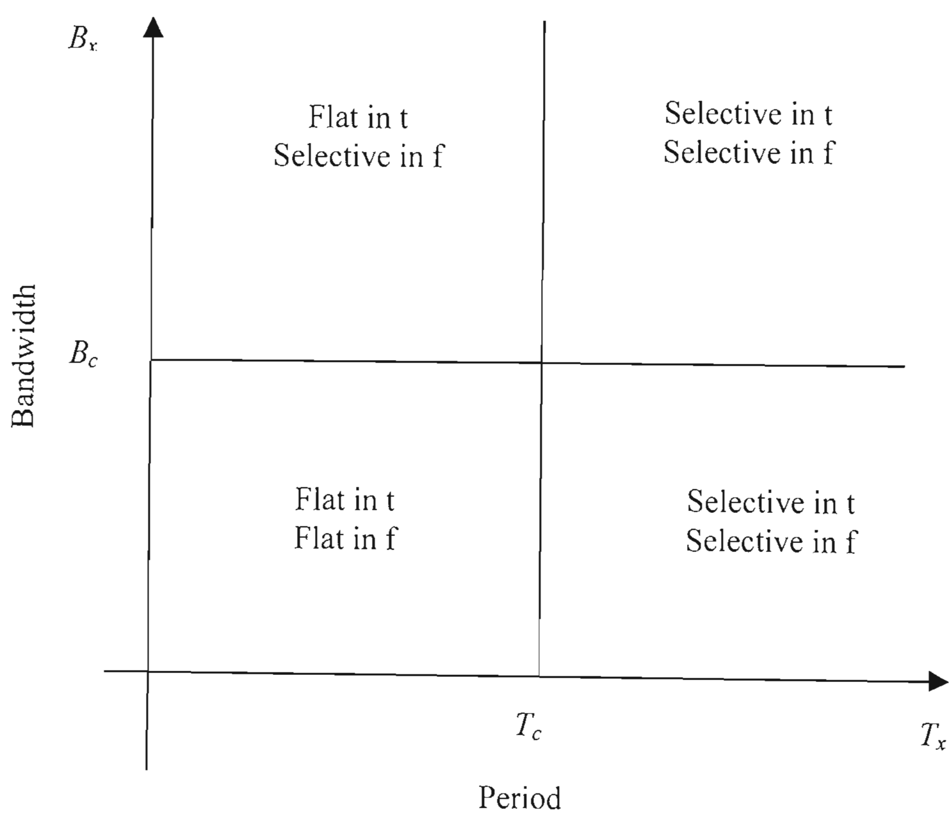


Figure 4.6 Fading Channel Types

5. Simulation of the CDMA-HF Radio System

As discussed previously, part of the research objectives were to construct, implement and simulate a channel model in the HF range. The main purpose of this was

- For comparative analysis of the various error control techniques.
- Providing insight into effects of processing gain in an HF environment.
- Provide starting points for hardware design.
- To determine if a spread spectrum signal can provide protection in a high multipath environment.

The simulation environment used was Matlab and Simulink. The easy manipulation of blocks in the Simulink environment and the computational dexterity of Matlab allowed the rapid prototyping of channel models and error correction schemes. Simulink is a time flow simulation package [8] as opposed to a data flow package. This means that each block in a simulation advances with each simulation step. In time flow simulators, each block is simulated after the end of the previous block. The advantage of using the time flow method is that one can quickly see the effect of a change in a parameter, in real time on performance variables (e.g. BER) without having to wait until the entire simulation is ended.

5.1 Low Pass Equivalent Modelling

The bandpass signal representation used in the models and subsequent simulations are their complex bandpass representations or low pass equivalent. This is a practical way of simulating signals without incurring the massive simulation time penalty of using a signal modulated onto a high frequency carrier.

In a baseband system for instance, the simulated system has to obey the Nyquist sampling theorem such that,

$$T_d > \frac{1}{f_c} > 2T_s$$

where T_d, is the duration of the bit, f_c the carrier frequency and T_s is the sampling time.

With f_c, in the MHz range it becomes obvious that simulating at passband becomes impractical, due to the extremely long simulation times. Using, the complex baseband representation this requirement is reduced to a more practical

$$T_d > 2T_s$$

The general form [9] of a bandpass system can be given as

$$x(t) = A(t)\cos[2\pi f_c t + \phi(t)] \qquad (5.00)$$

where $A(t)$, f_c and $\phi(t)$ are the amplitude carrier frequency and phase of the signal respectively. Through trigonometric expansion, this can be expanded to

$$x(t) = u_I(t)\cos 2\pi f_c t - u_Q(t)\sin 2\pi f_c t \qquad (5.01)$$

where

$$u_I(t) = A(t)\cos\phi(t) \quad \text{and} \quad u_Q(t) = A(t)\sin\phi(t)$$

describes the complex envelopes of the carrier frequency components. Using complex notation, this can be written as

$$x(t) = \Re\{x_+(t)\} \qquad (5.02)$$

where $x_+(t)$ is the pre-envelope of $x(t)$. This can be represented as a phasor, giving the bandpass signal as the image along the real axis

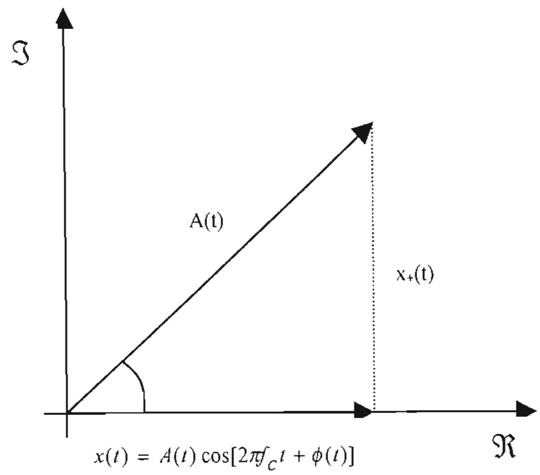


Figure 5.00 Complex Envelope Representation

Thus x_+ can now be written as

$$x_+(t) = u(t)\exp(j2\pi f_c t) \tag{5.03}$$

where

$$u(t) = u_I(t) + ju_Q(t)$$

$u(t)$ is known as the complex envelope or complex low pass equivalent of $x(t)$. The pre-envelope $x_+(t)$ can be recovered by rotating $u(t)$ with an angular velocity of f_c. Thus it can be said that $x(t)$ is completely described by $u(t)$ and f_c. It is this complex representation that is used in the following channel models and simulations.

5.2 Channel Categories

To help describe the different parts of the model, the following catagories of channels are used [9]

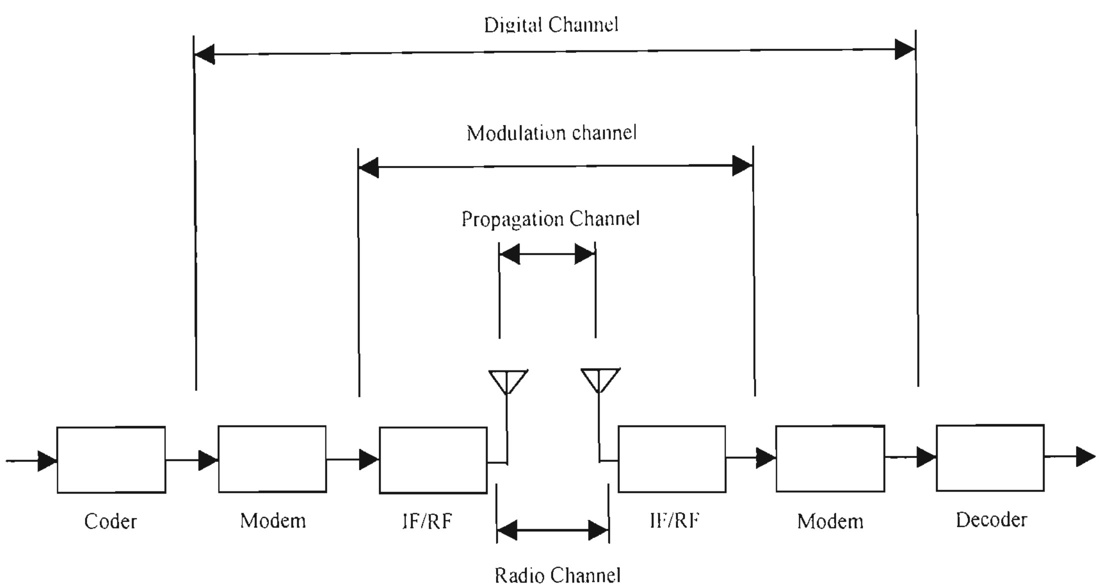


Figure 5.01 Channel Types

The propagation channel describes the physical medium through which transmission of the signal occurs. In the current application, it describes the ionosphere and the effect it has on propagation. The ionosphere is not linear due mainly the ionisation of plasma and cannot be considered time invariant due to the fact that the layers are always moving and that multipath occurs.

The radio channel constitutes both the transmitting and receiving antennas and the propagation channel. This channel is more fully described in the hardware section of the dissertation. The modulation channel describes the other parts of the hardware such as the amplifiers, mixers and input/output stages and includes the radio channel. This channel is also more fully described in the hardware section.

The digital channel describes all the blocks connecting the unmodulated signal from the transmitter to the receiver. It is this channel that is modelled in the following simulations. Various factors of signal distortion are taken into account, which will now be described.

5.3 HF Channel Model

There are three main disturbances that need to be included when designing a channel model of the HF band. These are

- Distortions caused by the ionosphere
- Interference caused by other users
- Thermal noise and other disturbances

The following section will describe the systematic design of a HF channel model that was used in the simulations. The choice of modulation is BPSK.

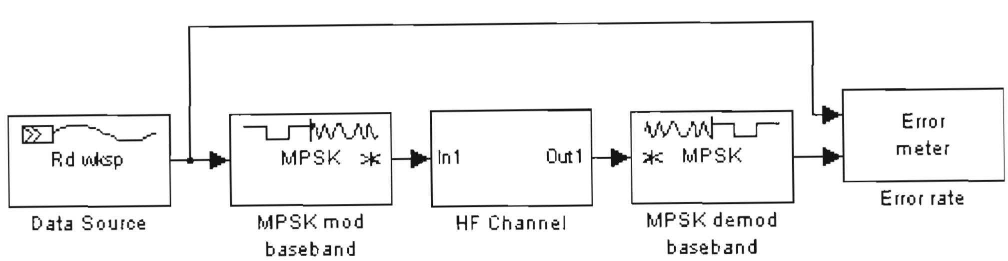


Figure 5.02 Unspread, Uncoded System

Shown in Figure 5.02 is the main block diagram of the uncoded, unspread system. The information source is randomly generated data. This data is then MPSK modulated, giving a complex envelope output. The modulated data is then "transmitted" through a model of the HF channel, demodulated by the MPSK complex envelope demodulator and fed into an error meter. The error meter measures the BER and it is this performance variable that is used when making comparisons of different parameters. This basic system is used as the benchmark for comparing the different spread spectrum schemes and error correction methods.

As can be seen it is just a basic communications system with no error detection or spreading performed. The internal composition of each block will now be discussed.

Data Source:

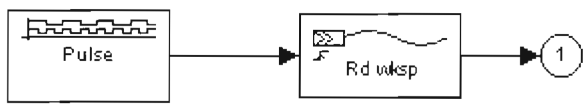


Figure 5.03 Data Source

The data source consists of a pulse generator connected to a triggered input port from a workspace block. The parameters that can be set in the data source block are:

Parameter	Setting
Variable	Randint(1000,1,2)
Data output sample time	1
Cyclic read	1
Initial output	0

Table 5.0 Data Source Block Parameters

The *randint* function together with the parameters *(1000,1,2)* outputs a random integer 1 column wide, 1000 bits long, in binary format.

The *Data output sample time* controls the output bitrate (BR_{output}) i.e.

$$BR_{output} = \frac{1}{T_{Data\_out}} \qquad (5.04)$$

The *Cyclic read* parameter forces the block to output the same vector continuously, i.e. when it gets to the end, it starts an output cycle from the beginning again.

Finally the, *initial output* parameter controls what the first bit will be at the start of a simulation.

MPSK mod baseband:

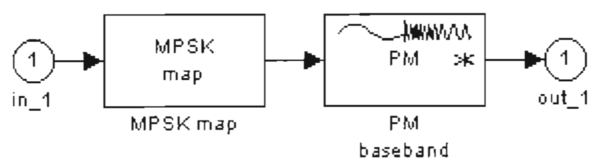


Figure 5.04 MPSK mod baseband

Digital signal mapping codes an input digital signal into a signal ready for modulation [8]. *Digital signal demapping* decodes a demodulated signal back into a digital signal. A digital modulation block can be divided into two major parts: digital-to-analogue mapping and analogue modulation. A digital demodulation block can also be divided into two major parts: analogue demodulation and analogue-to-digital demapping. The process is shown more clearly in Figure 5.05.

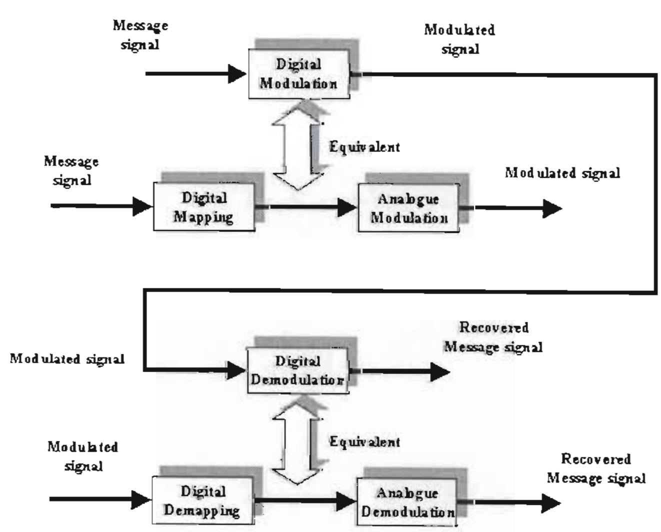


Figure 5.05 MPSK Modulation and Demodulation

MPSK map:

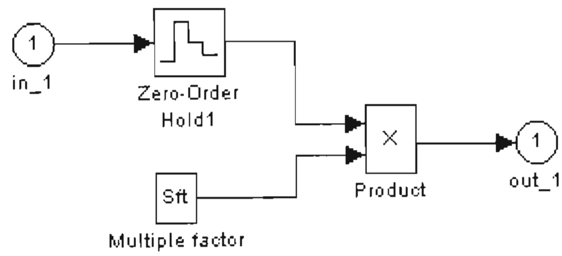


Figure 5.06 MPSK map

The MPSK map block converts an integer signal (binary in this case), into a Phase Shift Keyed signal.

PM baseband:

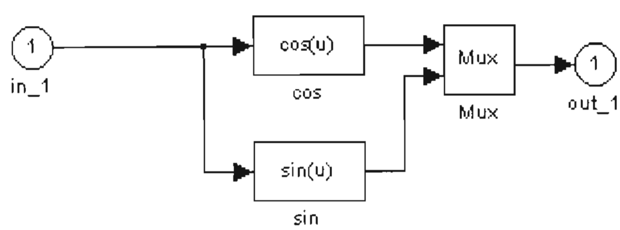


Figure 5.07 PM baseband

The PM baseband block shown in Figure 5.07, outputs the complex envelope of a phase modulated signal.

MPSK correlation Demodulation:

The MPSK Correlation Demodulation block calculates the correlation between an input signal and a vector of carrier frequency sinusoidal signals, all of which have the same frequency but vary in phase [8]. The block implementation is given in Figure 5.08.

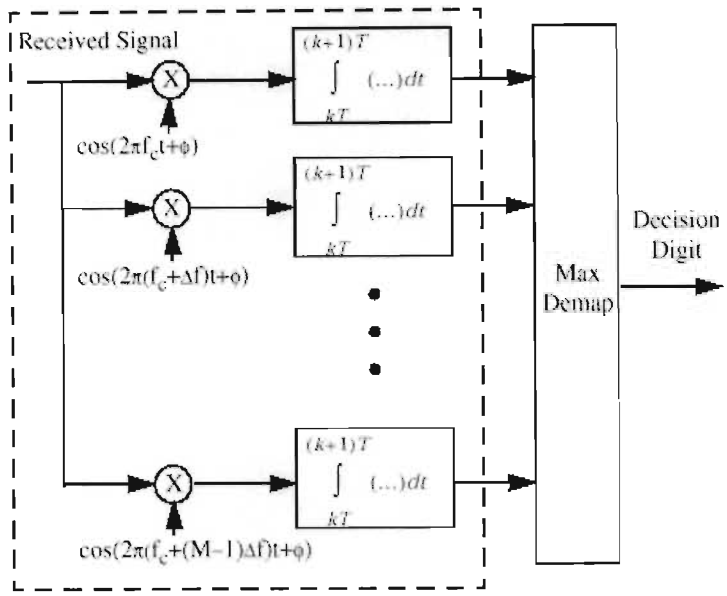


Figure 5.08 MPSK Correlation Demodulation

The block implementation of the MPSK Correlation Demodulation module is shown in Figure 5.09. For completeness, the internal Matlab implementations are shown in figures 5.09a, 5.09b and 509c. The first diagram shows the complete system and the following two diagrams show the sub-blocks that make up the complete demodulator.

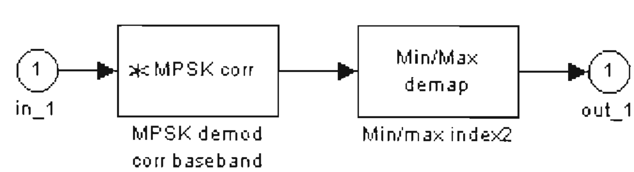


Figure 5.09a Demodulator Sub-blocks

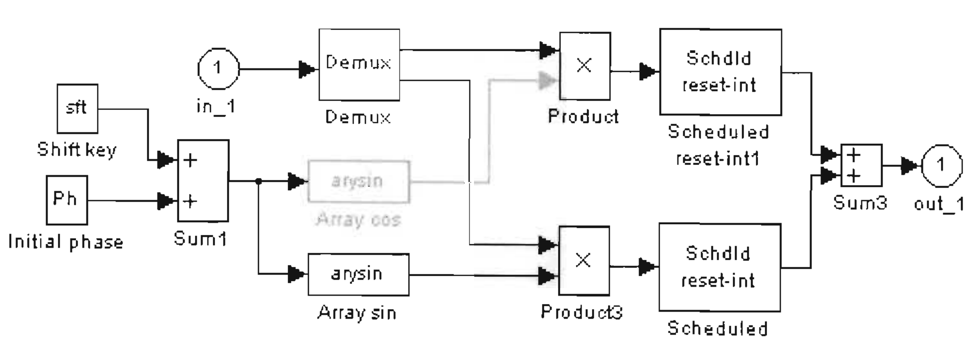


Figure 5.09b Demodulator Sub-blocks

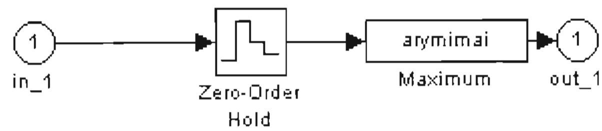


Figure 5.09c Demodulator Sub-blocks

The parameters that can set in the MPSK modulation block are given in Table 5.1 with the simulated values.

Parameter	Setting
M-ary number	2
Symbol interval and offset	1
Initial phase	0

Table 5.1 MPSK modulator Parameters

The parameters that can be set in the MPSK demodulation block are given in Table 5.2, also with the simulated values used.

Parameter	Setting
M-ary number	2
Symbol interval and offset	1
Initial phase	0
Sample time	0.01

Table 5.2 MPSK Demodulator Parameters

HF channel:

The actual HF channel model that was designed for the simulation will now be discussed. The complete block diagram of the channel model, is given in Figure 5.10. This system is contained in the block labelled *HF channel*, in Figure 5.02.

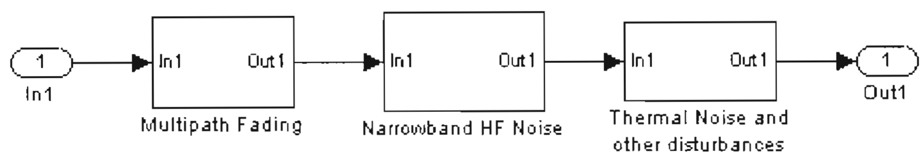


Figure 5.10 HF Channel model

The three main sub-systems that make up the model are the *Multipath Fading* block, the *Narrowband HF Noise* block and the *Thermal Noise and other disturbances* block. The internal design of these sub systems will now be discussed.

HF channel sub-blocks:

Multipath fading

The multipath fading subsystem is modelled as a Rayleigh faded channel with varying phase and fading envelope to simulate the effects of multipath distortion. The fading channel in the baseband signal transmission has the following form where $u(t)$ is the transferring signal; $y(t)$ is the output of the channel; $\alpha(t)$ is the fading envelope and $\phi(t)$ is the phase shift in radians:

$$Y(t) = u(t)\alpha(t)exp(j\phi(t)) \qquad (5.05)$$

Shown in Figure 5.11 gives the block diagram representation.

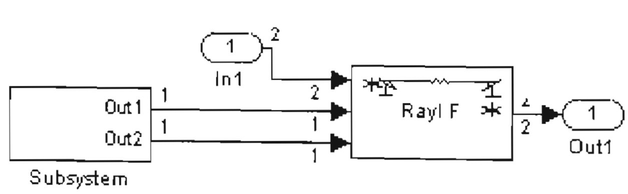


Figure 5.11 Multipath fading

The topmost input is the incoming complex enveloped MPSK signal. The other two inputs are Rayleigh noise generators that control the amount of phase distortion and the amount of fading applied to the envelope. The Rayleigh noise generator block generates signals that are Rayleigh distributed. The PDF of the Rayleigh noise is given as:

$$f(x) = \frac{x}{\sigma^2} \exp\left(-\frac{x^2}{2\sigma^2} \right) \qquad (5.06)$$

where σ^2 is the common variance and x, the Rayleigh random variable is given as:

$$x = \sqrt{y_1^2 + y_2^2}$$

where y_1^2 and y_2^2 are two-dimensional, zero mean Gaussian variables [8]. The Matlab implementation is shown in Figure 5.12.

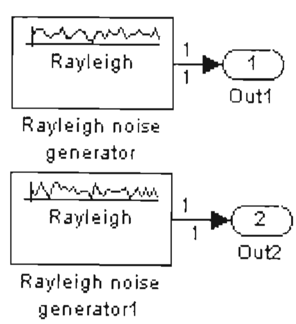


Figure 5.12 Noise Generators

Further, these noise generators have been implemented by Matlab [8] using the method shown in Figure 5.13

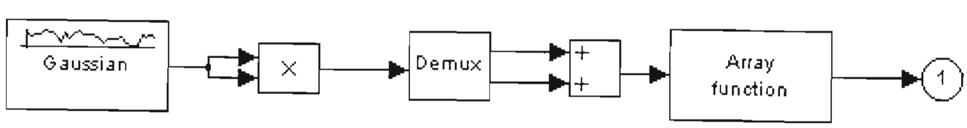


Figure 5.13 Noise Generator Implementation

Narrowband HF noise

The next block in the channel model is the modelling of narrowband HF noise such as other narrowband users.

This is implemented using the equations discussed in section 4.3. The noise is added after being generated in a separate sub block as is shown in Figure 5.14

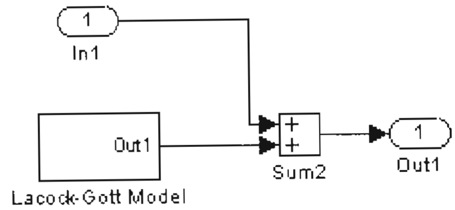


Figure 5.14 Laycock-Gott Model Implementation

The laycock-Gott model sub block implementation is shown in Figure 5.14.

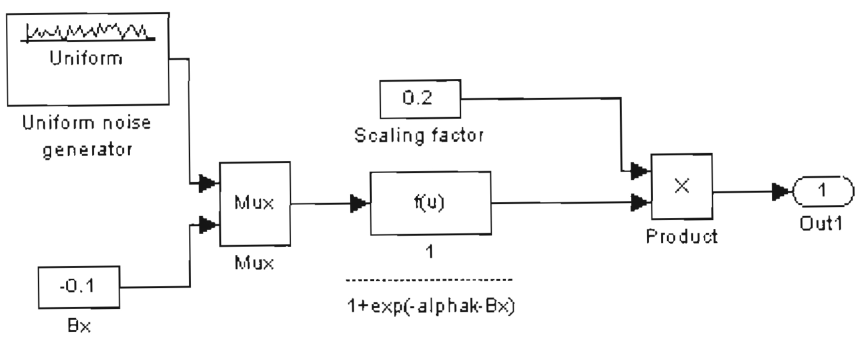


Figure 5.14 Laycock-Gott model sub-blocks

Thermal noise and other disturbances

Finally, the third sub system in the channel model takes into account thermal noise, which is modelled as AWGN, whose mean can be specified, and any other disturbances that is simply modelled as a constant. This can be set to any arbitrary level to simulate the random effects of a disturbance. The complete sub system is shown in Figure 5.15.

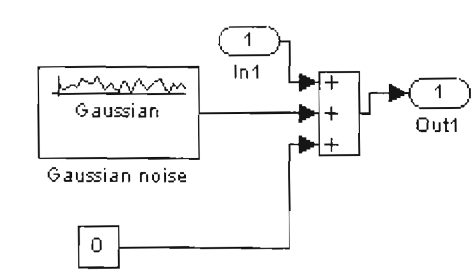


Figure 5.15 Other Noise Disturbances Implementation

5.4 Uncoded, Unspread System

The initial simulation runs were done using no spreading and no coding. This was done using trial and error to obtain values for the many different parameters in the model, until a BER was achieved which was comparable to that found in the literature. To observe the effects of the channel in the time domain, a random bit sequence was simulated. Figure 5.16 gives the output of the MPSK modulator where *M=2*. The phase transitions can clearly be seen as each bit is sent.

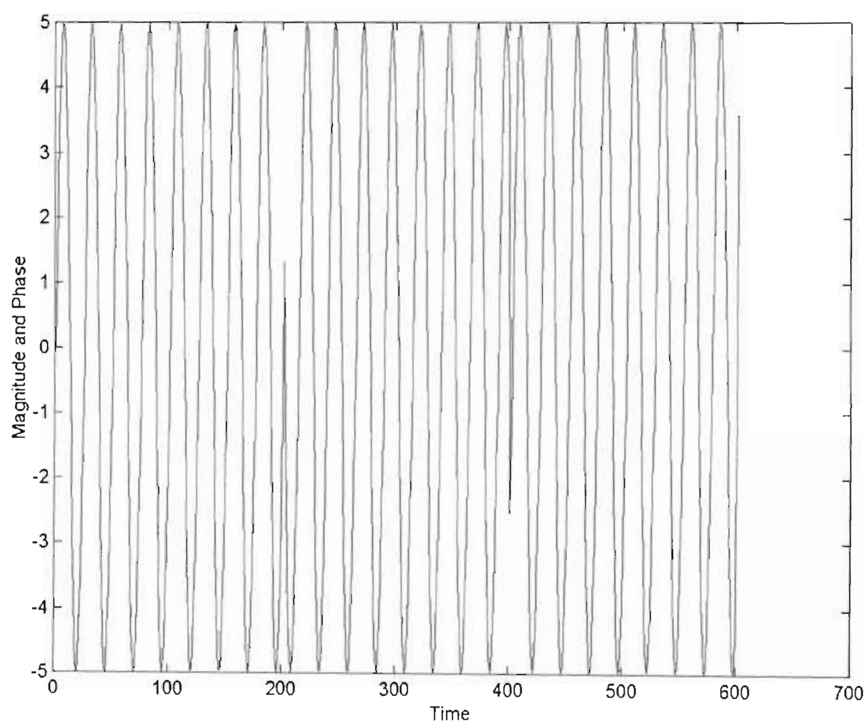


Figure 5.16 MPSK Modulation

The effects of the channel are displayed in Figure 5.17 for the same bit sequence. The multipath distortion induced fading can also be clearly observed.

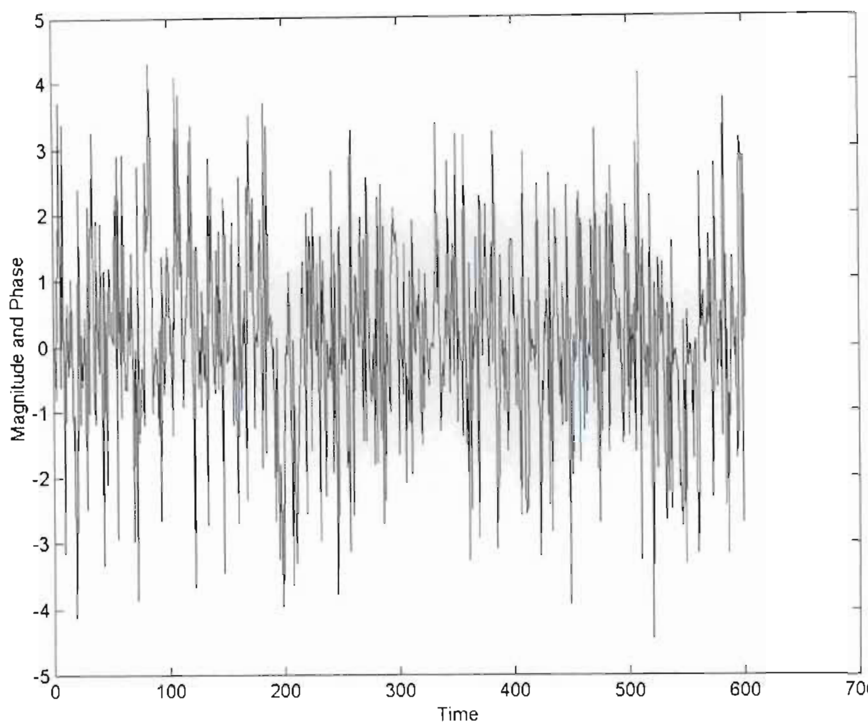


Figure 5.17 Channel Distortion

The parameter names, symbols and values used in the various sub-blocks in the system are tabulated in Table 5.3, as well as the performance of the channel.

Functional block	Parameter	Symbol	Value	BER
Multipath fading block	Mean	σ	0.95	0.452E-3
Narrowband HF interference	Kth ITU bias, HF spectrum constant	α_K , β_x	0.1 , [-16,9]	
Thermal noise and other disturbances	Other Disturbances, [Mean, Variance]	OD , [δ, δ^2]	0.1 , [-16,9]	

Table 5.3 Unspread, Uncoded Results

5.5 Spread System

The block diagram of the spread system is shown in Figure 5.18. For clarity those blocks already discussed have been incorporated into there own sub-systems. For consistency purposes, the same model parameters, obtained in section 5.4, were used in these simulations to aid comparisons between the spread and non-spread systems. The PN sequence used was the actual code used in hardware, discussed in chapter 6

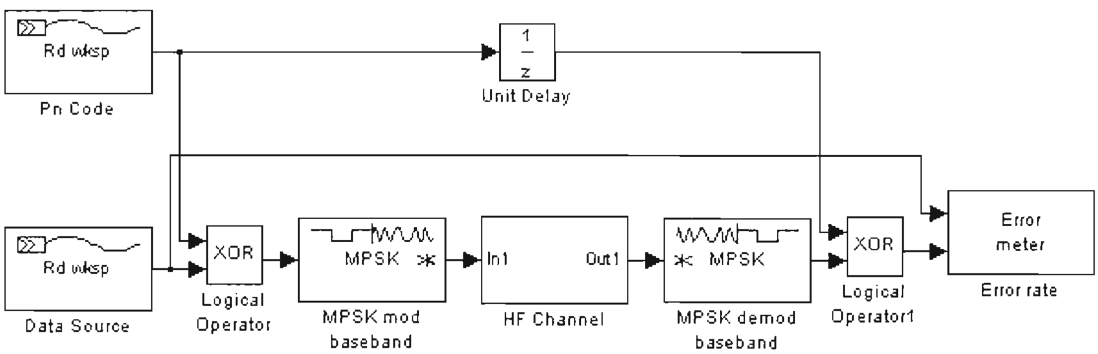


Figure 5.18 Spread, uncoded system

Spectrum:

Figure 5.19, and 5.20 shows the spectrum of the unspread system and that of the spread system.

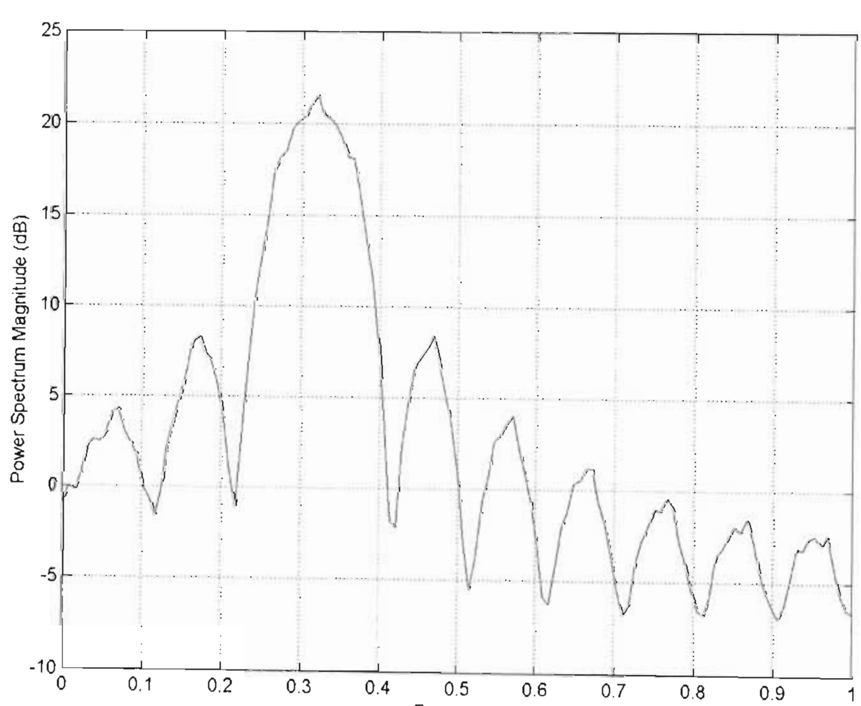


Figure 5.19 Unspread Spectrum

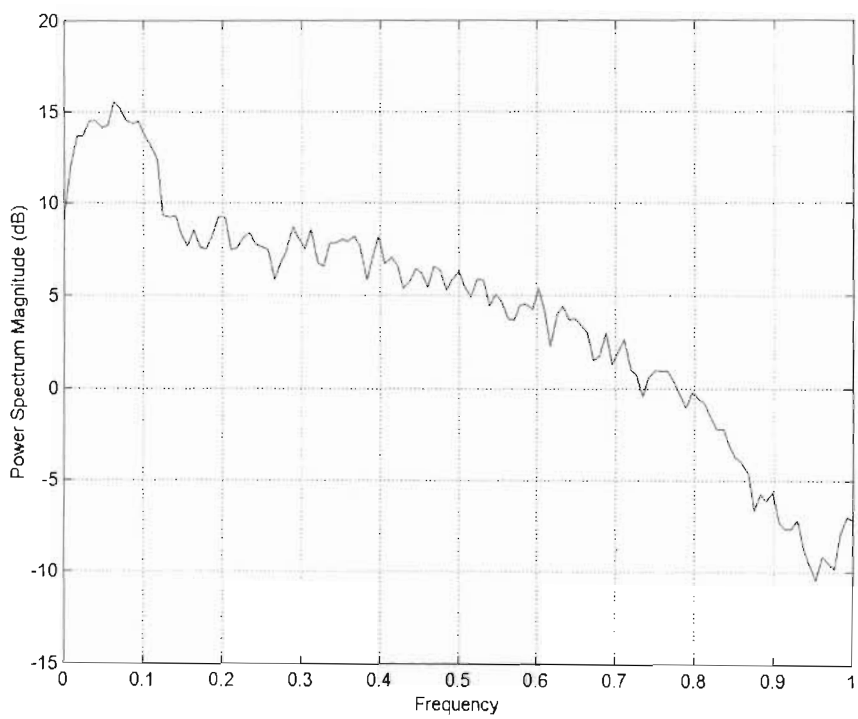


Figure 5.20 Spread Spectrum

Results:

Table 5.4 gives the results for the spread system. The parameter that was varied this time was the PN length.

PN length	BER
15	6.53E-4
255	6.48E-4

Table 5.4 Spread, Uncoded results

5.6 Spread and Coded System

Finally a system that is both spread and coded is now simulated. The two main reasons for coding are to combat the averaging effects of Direct Sequence as highlighted by Anderson et al. [20], and to combat the effects of narrowband noise.

The code chosen was a Reed Solomon FEC code as it provides good protection against burst errors as caused by drop out fades and burst noise. The block diagram is shown below.

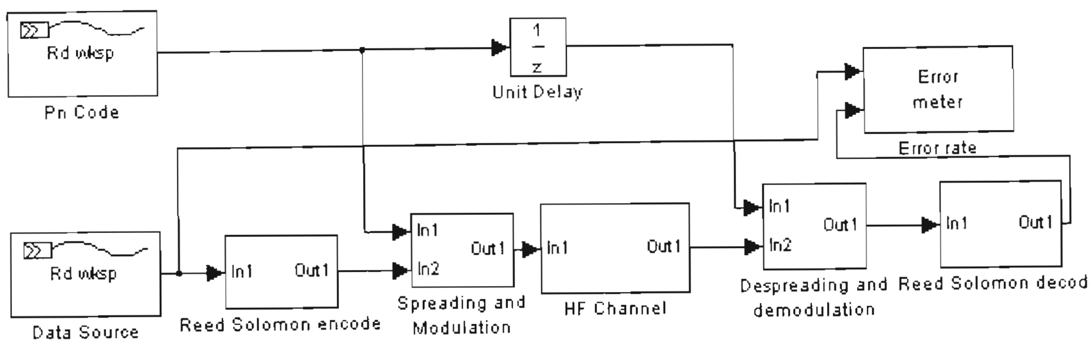


Figure 5.21 Spread Coded System

Results:

The results for the coded spread system are now presented below

PN length	BER
15	0.194E-4
255	0.028E-4

Table 5.5 Spread Coded Results

5.7 Discussion

The uncoded, unspread system was again simulated until the bit error rate achieved was comparable with those obtained in the literature. This was accomplished by varying the parameters given in table 5.3, and running the simulations until the desired results where obtained. The error rate achieved was the target while the parameters of the model were the variables. This was then used as benchmark in evaluating the effectiveness of the spreading and the error correction scheme.

For the spreading system, the two codes that were chosen were the actual 15 bit and 255 bit codes which were available on the hardware development boards. The results show that although the BER is not substantially reduced, there is a slight reduction. The failure of the spreading to reduce the BER performance to a level of several orders of magnitude can be ascribed to the fact that spreading provides no protection against two of the noise sources in the model. It only provides protection against the burst noise of the narrowband users, controlled in the model by the Laycock-Gott equation. The model is thus limited in the sense that although it provides BER values which agree with those published in the literature, the weighting and values of the error parameters in the noise models, is still not fully described. Indeed these values will be dependent on location of the link [20]. To fully describe a link with the current model, measurement of factors such as local spectral occupancy by narrowband users would have to be undertaken.

The final stage of analysis involved using a Reed-Solomon Forward Error correction scheme. Again the effectiveness of this form of diversity proved to be small. This is ascribed to the fact that with the emphasis of the noise components being on thermal and narrowband users, the scheme used and suggested in the literature [16] of Reed-Solomon encoding, provides more protection against the burst errors cased by natural phenomena such as lightening, plasma irregularities and solar storms. Further work should focus on hybrid schemes incorporating the forms of FEC described in [17]. These will however, need to be adapted to a wideband environment, as they have been optimised for narrowband use.

6. Implementation of the CDMA-HF Radio System

6.1 Introduction to Hardware

Part of the research also involved investigating the various practical aspects of the system that included the design and implementation using readily available technologies. This chapter presents the results and performance of the equipment used in implementing a practical CDMA HF link as shown in diagram 4.1. The main modules of the system were the user terminal, CDMA transceiver, HF band power amplifier and antenna systems.

6.2 Hardware Description

User terminal

The user terminal or Man Machine Interface (MMI) used was a Pentium class PC. For field measurements, a notebook computer was used to facilitate a more portable structure. The operating system on both machines was Windows 95^{TM} running *Spherecomm,* an application developed as part of the project. The choice to use standard PC's and not propriety hardware was made to show the viability of using standard modules to build relatively inexpensive communications systems for a rural environment.

ASTRA (Advanced Spread-spectrum Transceiver ASIC) *CDMA Development board*

For the CDMA transceiver, a development board from *Sirius* communications was utilised. The board performed all of the functions of spreading\despreading, pre-modulation and IF frequency control. The actual development board is shown in Figure 6.00.

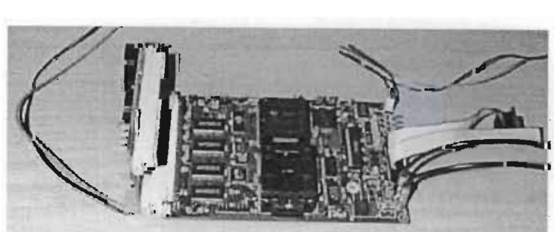


Figure 6.00 ASTRA Development Board

A block diagram of the ASTRA is shown in Figure 6.01.

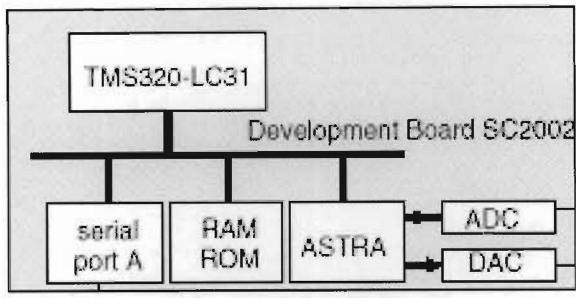


Figure 6.01 ASTRA Block Diagram

Basic operation of the board is as follows [22]:

The ASTRA is a highly programmable, integrated baseband chip for building direct sequence spread spectrum modems. The board also contains a DSP that can be used to configure the modem parameters in the ASTRA, execute synchronisation algorithms and monitor the modem behavior. The board also features a serial communication channel, a DAC and an ADC. Real time tests can be performed for a wide range of modem specifications. The supporting development software includes low-level ASTRA access functions, access functions for the serial I/O channel and an example of an acquisition and synchronisation algorithm. Once a modem design is initialised, all settings, software and test messages can be stored in a 4 Mbit flash EPROM on the board, without the need for any external programming device.

Of the various functions that the ASTRA board is able to control, the most important are PN length, symbol rate, IF frequency and modulation type. The PN lengths that are supported are a 15 and a 255 bit PN code. Other codes are possible but the firmware requires re-flashing. In the current application only the standard 15 and 255 bit codes were tested. With the length of the PN code and symbol rate controlling the process gain, this quantity is given as

$$G_p = 10\log[PNcodelength] \qquad (6.00)$$

Figure 6.02 shows an autocorrelation plot of the PN sequence on the ASTRA board showing early, medium and late gate synchronisation.

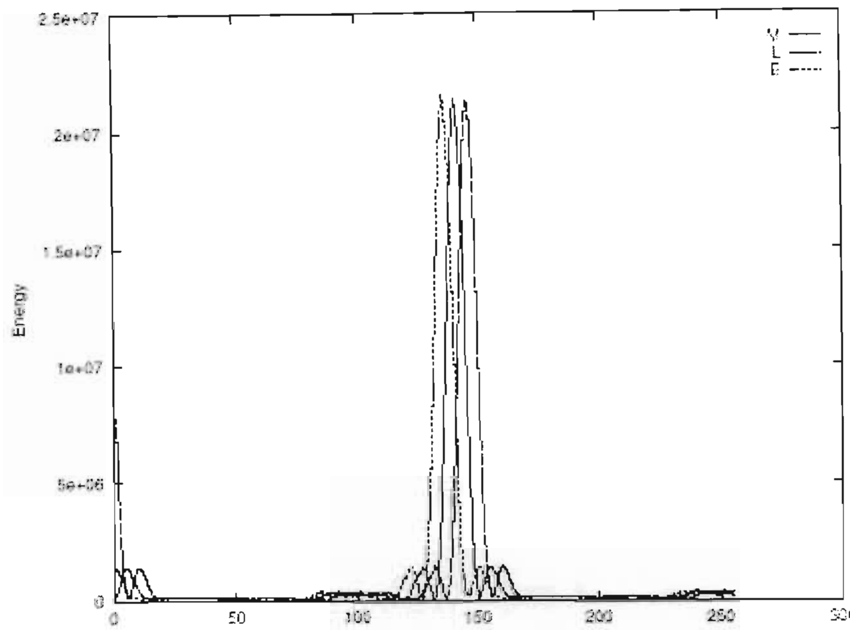


Figure 6.02 PN Code Autocorrelation

Various phase modulation schemes are possible with the board, such as PSK, QPSK etc. However, for the research project only PSK modulation was used. Figure 6.03 shows an oscilloscope screen shot of the output of the ASTRA taken from the DAC, clearly showing PSK modulation.

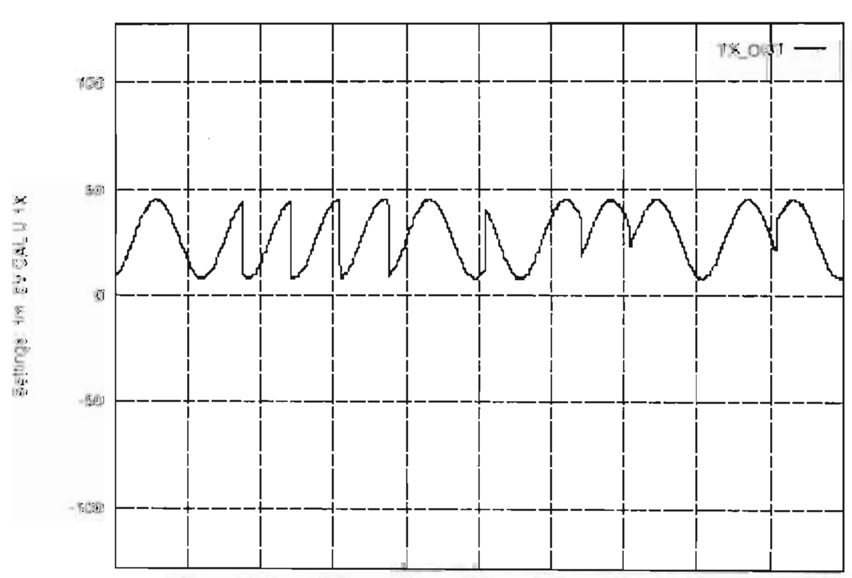
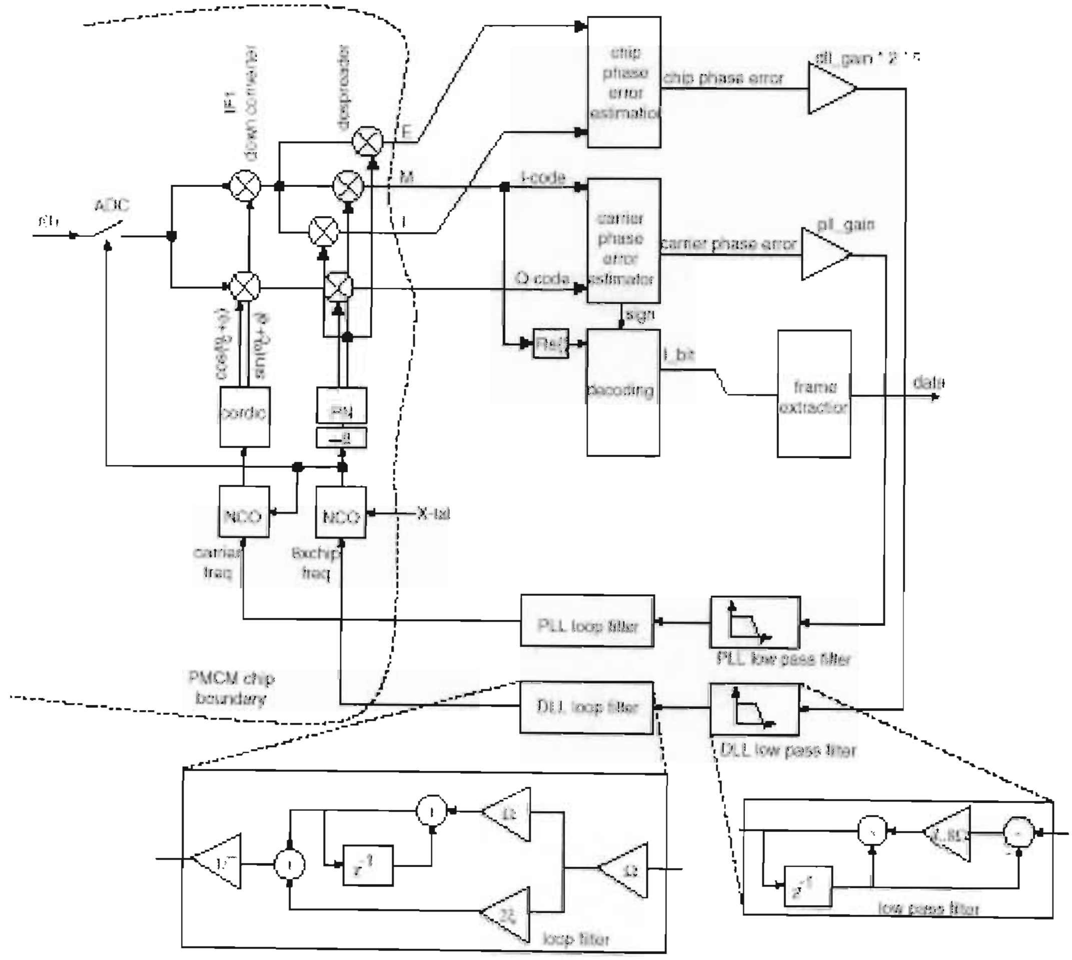


Figure 6.03 ASTRA PSK Output

It is beyond the scope of this text to fully describe the spreading and de-spreading functions of the ASTRA. A full explanation is given in reference [22]. Suffice to say that coarse acquisition and fine-tuning are achieved using a Delay Locked Loop (DLL) and Phase Locked Loop (PLL). Both the DLL and PLL are implemented digitally on the DSP that resides on the development board. This process is shown in Figure 6.04.



Mulier Kristoffel, 'Introduction to the ASTRA DS-SS board'; www.sirius.com

Figure 6.04 ASTRA DLL and PLL Implementation

HF Band Power Amplifier

As with all communications systems, a final stage of power amplification is necessary for the transmission of a radio signal. The end stage of the ASTRA contains a DAC that converts the digital signal of the board to an analogue signal, ready to drive a power amplifier.

The amplifier used was a 25W module that operated in the HF frequency range. Initially it was found that overheating caused non-linear performance. A suitable heat sink with forced air cooling was used to solve this problem. Another problem encountered was the very high current rating required for the power supply. For proper operation the module required 5A. A solution was achieved using a switched mode supply operating at 24V and having a current rating of 7A. Although this performed adequately in a test bench setup, problems may be encountered in a rural environment. A solution would be to use sealed gel type rechargeable cells that have rating of 65 A\hr. This would give approximately 12 hours mean use between charges. Figure 6.05 shows the HF amplifier module used.

Figure 6.05 HF Power Amplifier

Antenna system

The antenna system was a compromise between size and spectral capability. Initially two inverted-V antennas were used for both transmission and reception. These were wide-band antennas and could operate over the entire HF spectrum. One was mounted on top of the roof of the department's building and the other was assembled at the various sites at which measurements were taken. The latter setup proved to be too cumbersome however as, at approximately 30m in length, the 'mobile' anntenna was difficult to assemble and reassemble at each location.

The solution used was a mobile antenna mounted on a tripod system. This made for far easier mobility as the entire structure was collapsible and was only 2m high. The penalty was that for the increased flexibility, the mobile setup was 'narrowband'. This was partially overcome by using a locally developed product called a multiwhip antenna.

The multiwhip allowed up to six frequencies to be used on one antenna. To change to a different operating frequency, the user simply plugs a banana plug into the appropriate tapped frequency slot.

This arrangement proved successful in the various field measurements that were made. Figures 6.06 and 6.07 show the two antenna systems. Unfortunately the V cannot be seen as it is a thin cable.

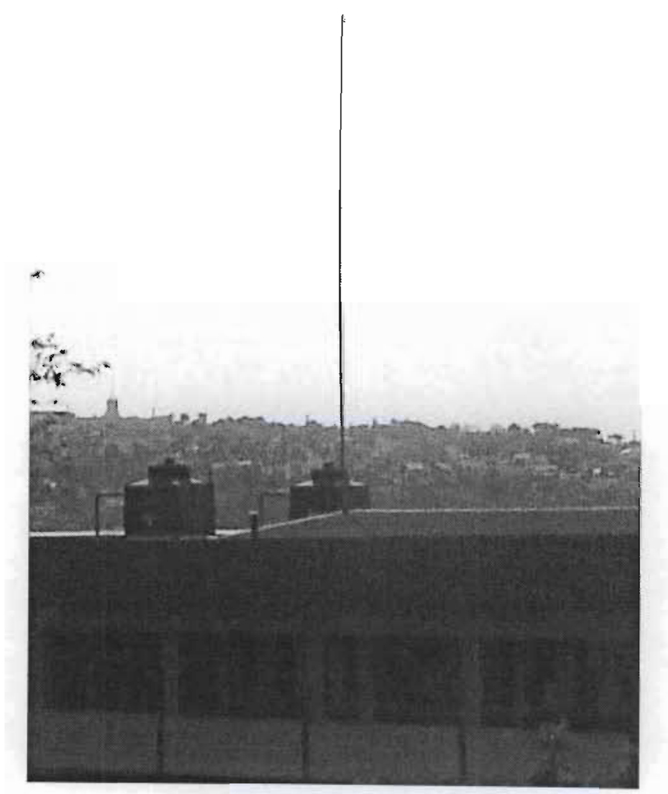


Figure 6.06 Wideband HF V Antenna

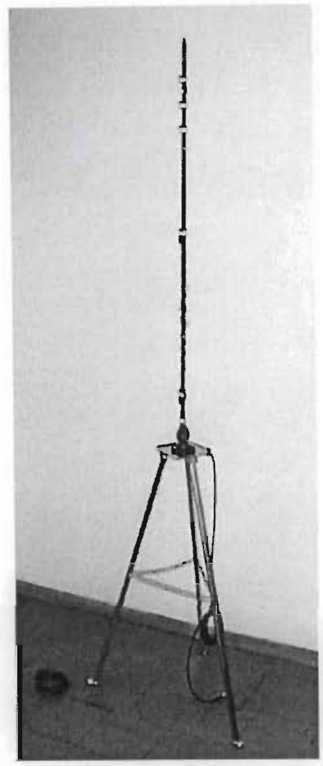


Figure 6.07 HF Mobile Antenna

6.3 Field Measurements

Various field measurements were taken over several weeks. The procedure that was followed was that a known bit pattern was transmitted from a distant location and the received signal was observed and measured on the laboratory's spectrum analyser. Initially it was hoped to perform real-time bit error measurements but this was not possible due to the inability of the development board's loops to track the received signal. This was traced to timing issues within the board, that were not user changeable. The boards were not designed to operate at such high IF frequencies and a better solution would have been to use some form of heterodyning. Thus received signal strengths were used to infer bit error probabilities. This had the drawback of introducing a margin of error into the results as an empirical equation cannot model all the physical phenomena associated with an RF communications link.

To calculate the relevant operational frequencies, a shareware program called Miniprop was used. This allowed the use of transmitter and receiver locations as well as sunspot information. The output gave the E-layer cut-off frequencies as well as the Maximum Usable Frequency.

These were then used to choose the correct tap on the mobile antenna and the correct IF setting on the ASTRA board. A typical output of the program is given in Figure 6.0.

```
MINIPROP (TM) SHORT-PATH PREDICTIONS        10-10-1998       Path Length :    206 km
Sunspot Number :   62.0    Flux : 112.3    F Hops :   1     Radiation Angle : 71 deg
TERMINAL A : 30.72 S     29.80 E     Farm                   Bearing to B : 337.8 deg
TERMINAL B : 29.00 S     29.00 E     Varsity                Bearing to A : 158.2 deg
Terminal A Sunrise/Set : 0332/1604 UTC   Terminal B Sunrise/Set : 0336/1606 UTC
```

			----------- SIGNAL LEVELS ABOVE 0.5 uV -----------				
UTC	FMUF	ECOF	3.0 MHZ	4.0 MHZ	5.3 MHZ	6.0 MHZ	7.0 MHZ
0000	4.0	0.5	71.0 A	68.5 A	66.1	65.0	
0200	3.9	0.5	71.0 A	68.5 B	66.1		
0400	5.1	2.1	66.5 A	65.2 A	63.8 B	63.1	62.1
0600	6.6	3.3	59.6 a	60.1 A	60.2 A	60.0 A	59.7 B
0800	7.8	3.8	54.9 a	56.6 A	57.7 A	58.0 A	58.1 A
1000	8.4	3.9	53.4 a	55.5 A	56.9 A	57.3 A	57.6 A
1200	8.6	3.7	55.5 a	57.0 A	58.0 A	58.3 A	58.3 A
1400	8.2	3.2	60.7 a	60.9 A	60.7 A	60.5 A	60.1 A
1600	7.5	1.9	67.8 A	66.2 A	64.5 A	63.6 A	62.6 A
1800	6.4	0.9	71.0 A	68.5 A	66.1 A	65.0 A	63.7 B
2000	5.5	0.5	71.0 A	68.5 A	66.1 A	65.0 B	63.7
2200	4.3	0.5	71.0 A	68.5 A	66.1	65.0	

Table 6.0 Miniprop Output

The first column shows the time for which the various predictions are made. The second and third column gives the F-layer Maximum Usable Frequency and E-layer Cutoff in MHz, respectively. Finally the columns with frequency headings give the predicted received signal level into a 50Ω load. Various locations were used for field measurements. Shown in Figure 6.08 is a contour map detailing two locations (terminal A) which were used, with the university (terminal B) as the peering point.

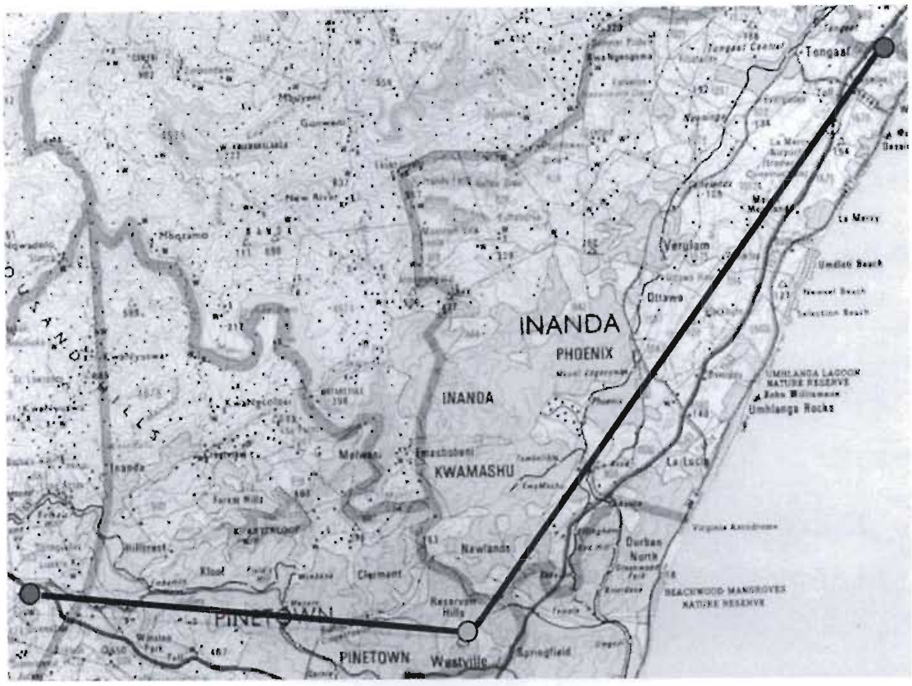


Figure 6.08 Contour Map of Test Area

Typical values of received signal strength which were read from the spectrum analyser, are shown in Table 6.1

Time	Frequency [MHz]	Signal strength [dBm]	Noise level [dBm]
8:21	4.3	-73.8	-90
11:13	5.3	-82.1	-95
12:54	7.6	-71.8	-90

Table 6.1 Field Measurements

6.4 Software Considerations

As discussed in the introduction of the chapter, the terminals need to run software, which will allow the various forms of communications to occur. An application called *Spherecomm* was developed to demonstrate the form this software could take and to provide a useful interface for the ASTRA board described in this section.

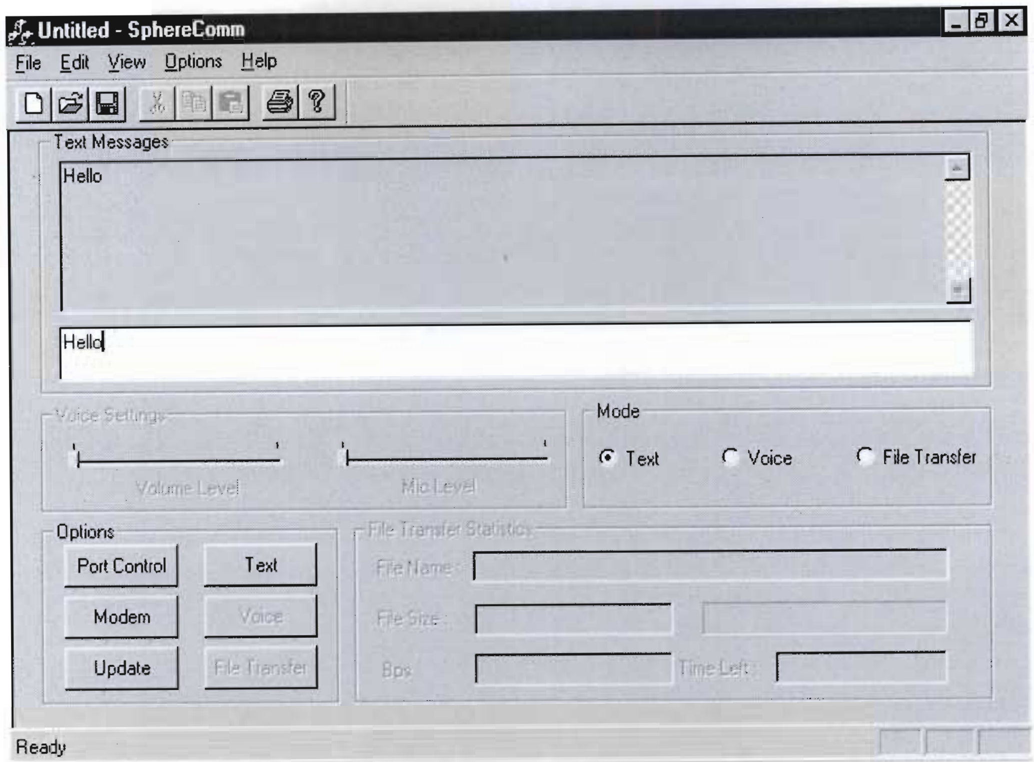


Figure 6.09 Spherecomm Main Screen

The start up screen of Spherecomm is shown in Figure 6.09. It is a single document interface (SDI) application written in Visual C++™ and Visual Basic™ where the document classes are the parameters for the CDMA modem. The title bar shows that there is an untitled document that means that either the default modem parameters are loaded or new parameters have been entered but have not yet been saved.

If this is the case then, like most Windows applications, a dialog box will ask the user whether he wants to save them and give them a file name. Then, in the future instead of re-entering all the parameters, of which there are 35, the user can just load the saved set.

As can be seen, the screen is divided into a number of frames. The *Options* frame gives focus and functionality to the three frames supporting the three different forms of communication, text, data or speech. When the *Text* radio button is selected (as shown), the *Text* frame gains focus and becomes operational. Communicating via text is very similar to communicating using an Internet Relay Chat (IRC) client. The users outgoing text is shown in the bottom white text box.

When the Enter key is pressed, the text message is transmitted. Incoming text is displayed in the upper, greyed text box.

The next option that can be selected in the *Mode* frame is the voice option. Although not implemented it could allow users to communicate using either real-time or non real-time voice. The third option allows the user to transfer files such as text documents or binary files such as images.

The *Options* frame has six buttons that allow the user to set various parameters for the modes described above. The first one is the *Port Control* button. Although the communications port is opened automatically when the application is run, this button is provided for manual control in the event of trouble shooting. The modem control button allows the user to alter the various parameters provided by the ASTRA and brings up the dialog shown in Figure 6.10. The *Update* button uploads the new parameters to the ASTRA.

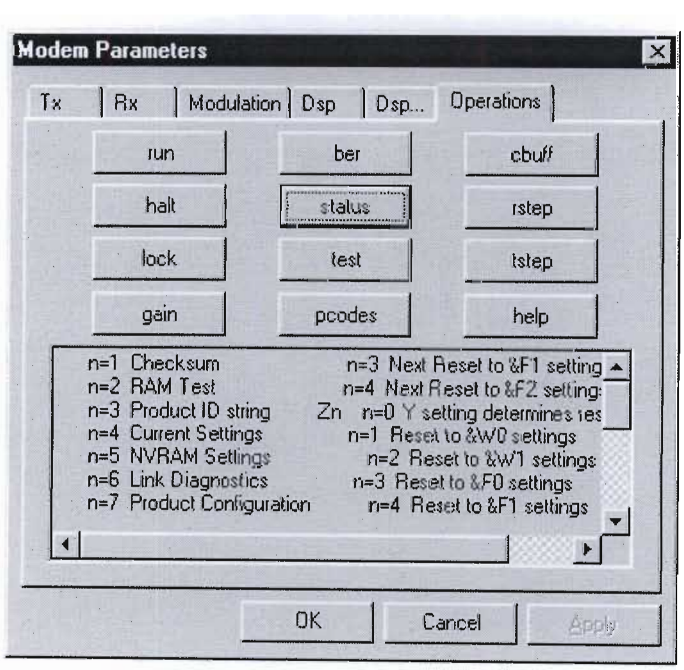


Figure 6.10 Modem Parameter Dialog Box

The *Text*, *Voice* and *File Transfer* buttons bring up the following dialogs, allowing the user to fine tune parameters that are applicable to the mode of communication.

Clicking the *Voice* button brings up the dialog of Figure 6.11, which allows the user to select the voice codec and the FEC error control strategy. With changing channel conditions this could be dynamically assigned.

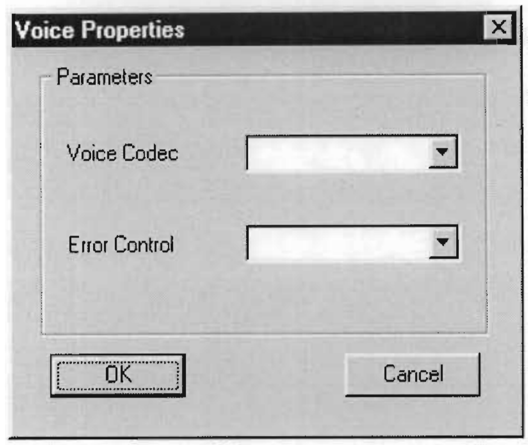


Figure 6.11 Voice Properties Dialog Box

The codec selection drop down menu allows the user to select a range of low bit rate codecs. As explained previously, these are hooked into the Audio Compression Manager that is part of the Windows™ operating system. An example of this choice is shown in Figure 6.12.

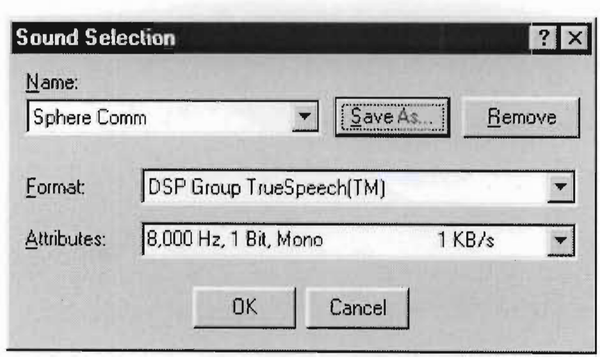


Figure 6.12 Voice Codec Dialog Box

Finally the *File Transfer* button allows the user to choose the local receive and send directories and file names. It also allows the choice of various ARQ algorithms that could also be dynamically assigned.

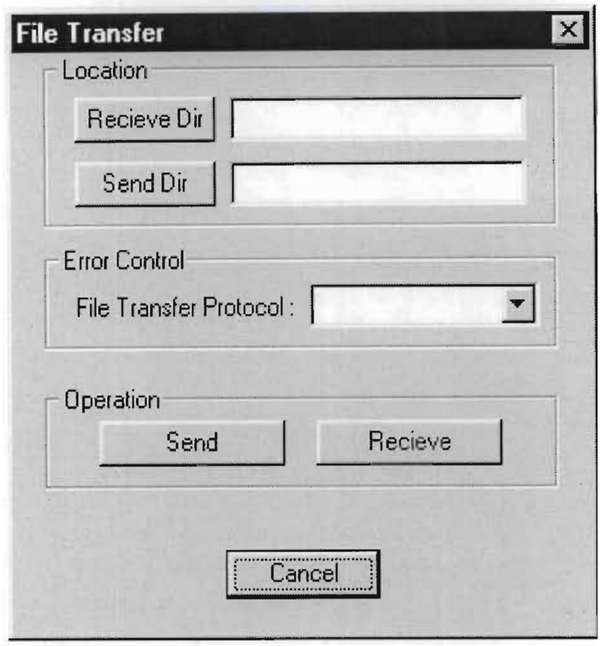


Figure 6.13 Data Transfer Dialog Box

6.5 Analysis

From Table 6.1, approximate values for the SNR can be calculated, given the corresponding signal and noise values at the measured frequency. A function on the spectrum analyser was used to measure these SNR's directly. Thus, bit error probabilities can be estimated from the discussions contained in sections 3.3.2 and 4.3. Table 6.2 shows the BEP obtained from these average SNR's. The frequency values used were limited by those that where available on the taps of the Multiwhip receive antenna. Figure 6.14 shows these values graphically on the P_e Vs SNR graph.

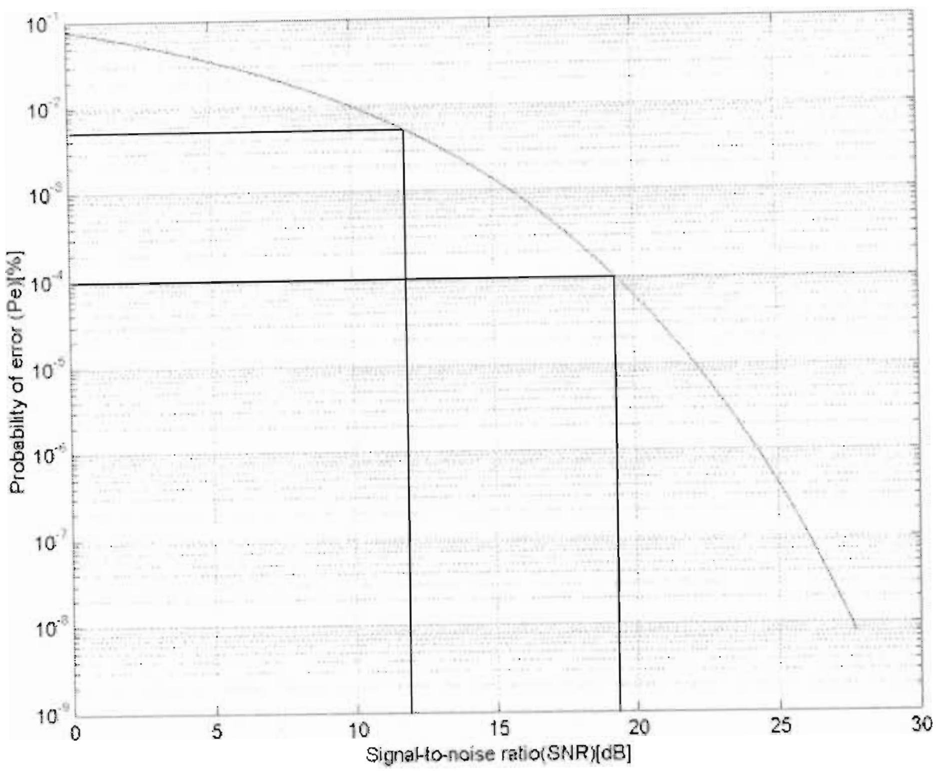


Figure 6.14 Measured SNR values and corresponding BEP

Frequency [MHz]	SNR [dB]	≈BEP
20	19	1.0×10^{-4}
13.85	19	1.0×10^{-4}
5.3	12	4.8×10^{-2}

Table 6.2 Estimated Bit Error Probabilities

The figures in table 6.2 need to considered with caution. They provide an approximate upper limit on what is achievable, as the effects of fading, burst noise and narrowband interference cannot be taken into account by simply using empirical formulas. They do however, provide useful insight into the practical ballpark figures which are obtained in a practical system. An approximate worst case can be obtained by applying the measured SNRs to the Laycock-Gott model and the results of Gunnar Anderson, discussed in section 4.3. Equation 4.08 and Figure 4.5 indicate BEP values in the range of 10^{-1}. This is of course, for an uncoded, "straightforward" DS system. Practical implementations could be expected to perform within these two boundaries.

Conclusion

This work has presented the design of an HF band direct sequence spread spectrum link for a rural environment. An introduction to the problem of providing communications infrastructure in a rural environment was given. It was shown that the two main problems associated with this were the very long distances associated with providing this kind of infrastructure and also that the small population densities associated with rural communities, made it difficult to make a business case. The advantages and disadvantages of various flexible radio channels were given. It was concluded that the attributes of a HF band radio make it a suitable candidate for rural access due to its ability to provide a long distance, free-to-air interface. The various aspects of ionospheric propagation were treated, showing how poor a channel the ionosphere is for the transmission of electromagnetic waves. This lead to the conclusion that some kind of diversity is needed in order to compensate for the poor channel operating conditions. The concept of a spread spectrum signal was developed, showing the reason for the spreading and the advantages gained by the spreading process. The various forms of multiple access were introduced and it was shown that spread spectrum can serve not only to protect the signal from interference caused by factors such as multipath distortion, but can also function as a multiple access medium.

The concept of a CDMA-HF radio link was investigated. A novel model of the HF band as a communications channel was developed and simulated, giving a performance similar to that achieved in the literature. This channel was then simulated again, using some of the diversity techniques described earlier, i.e. spreading and error control.

It was found that although the spreading provided some protection against corruption from the severe channel operating conditions, this was not enough for acceptable performance. This is partially due to the averaging affects associated with a direct sequence signal. The error correction technique that was used also provided a slight increase in performance, but again needing improvement. It can be concluded that a more thorough measurement of the local channel conditions, such as that undertaken

by Anderson et al., is needed to provide acceptable performance as well as more robust, adaptive error control methods.
The simulated results confirmed that more efficient coding techniques are needed to make the DS system more effective in a HF environment. This is due to the averaging nature of a DS spread spectrum system. However, the simulated results show that meaningful data can still be passed through the channel given the process gains empirically calculated from the field measurements. Assuming the band plan given in Figure 3.6, kilobit data rates could theoretically be supported, thus supporting voice using a low bitrate codec.

Finally the implementation of a practical CDMA-HF radio system was discussed. Various aspects of the actual physical link were dealt with such as selection of the correct power amplifier, antenna's and the ASTRA board. Various field measurements were made showing that the PLL and the DLL of the ASTRA need to be developed further in order to track the signal for demodulation. This is seen as an area of further research by the author as current commercial transceivers are not designed for the kind of signal distortions incurred in the HF channel. Software was developed to control both the CDMA transceiver and act as a man-machine-interface . Various forms of communications were illustrated, showing the kind of applications that could be supported by the proposed system.

In conclusion it can be said that although not all problems are solved and not all questions are answered, the work shows the way forward for the implementation of a CDMA based HF band communications system. It is envisioned that this kind system could play a key role in supplying communications infrastructure to under-served, rural populations in South Africa, where low population densities and distance are critical factors.

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