TRANSMISSION OF DATA UNDER MULTIPATH TRANSMISSION CONDITIONS

THESIS

BY

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SUMMARY

The explosive growth in microprocessor technology and the increasing use of computers to store information has increased the demand for data communication channels. Because of this, data communication to mobile vehicles is increasing rapidly. In addition, data communication is seen as a method of relieving the current congestion of mobile radio telephone bands in the U.K. Highly reliable data communication over mobile radio channels is particularly difficult to achieve, primarily due to fading caused by multipath interference. In this thesis a data communication system is described for use over radio channels impaired by multipath interference.

The thesis first describes radio communication in general, and multipath interference in particular. The practical aspects of fading channels are stressed because of their importance in the development of the system. The current U.K. land mobile radio scene is then reviewed, with particular emphasis on the use of existing mobile radio equipment for data communication purposes.

The development of the data communication system is then described. This system is microprocessor based and uses an advanced form of automatic request repeat (ARQ) operation. It can be configured to use either existing radio-telephone equipment, totally new equipment specifically designed for data communication, or any combination of the two. Due to its adaptability, the system can automatically optimise itself for use over any channel, even if the channel parameters are changing rapidly.

Results obtained from a particular implementation of the system, which is described in full, are presented. These show how the operation of the system has to change to accomodate changes in the channel. Comparisons are made between the practical results and the theoretical limits of the system.

Keywords

MULTIPATH INTERFERENCE, DIGITAL COMMUNICATIONS,
LAND MOBILE RADIO, FREQUENCY SHIFT KEYING (F.S.K.),
MICROPROCESSOR SYSTEMS

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AFC	automatic frequency control
AM	amplitude modulation
ARQ	automatic request repeat
ASCII	American standard code for information interchange
ASK	amplitude shift keying
BER	bit error rate
CRC	cyclic redundancy check
DSBSC	double side-band suppressed carrier
FEC	forward error correction
FFSK	fast frequency shift keying
FSK	freuency shift keying
FM	frequency modulation
HF	high frequency
IF	intermediate frequency
LF	low frequency
MF	medium frequency
PLL	phase locked loop
PM	phase modulation
PMR	private mobile radio
PSK	phase shift keying
RF	radio frequency
RX	receiver
SAW	stop and wait (ARQ)
SNR	signal-to-noise ratio
SRT	selective re-transmission (ARQ)
SSB	single side-band
TX	transmitter

TXR	transceiver
UHF	ultra-high frequency
VFO	variable frequency oscillator
VHF	very-high frequency
VSB	vestigial side-band
VXO	variable crystal oscillator
WGN	white Gaussian noise

CHAPTER 1

INTRODUCTION

1.1 BACKGROUND

Over the last five years one electronic component has emerged which stands to make as much impact in the lives of everyone as did the transistor. However, the rate at which this component is developing is so much faster than that of probably any other electrical device ever made, that its impact will not only be felt sooner but have more direct consequences. This component is, of course, the microprocessor. Its development and that of its support circuits (memories, I/O controllers, and the like) has been so rapid that equipment designed using 'state of the art' components is likely to be out of date by the time it reaches the consumer market.

Fortunately, from the many microprocessors that have been commercially available a few have emerged as industry standards. Although technically superseded they will still be produced, perhaps not quite in their original form, for some years to come. Microprocessor manufacturers have also been, in some cases, wise enough to produce improved processors which are upwards compatible with their own, and sometimes with other manufacturers', earlier products. Variations on a particular processor have also been produced. These usually incorporate some memory and support functions, and allow designers to use their existing software and specialised hardware whilst reducing total component counts.

It is these microprocessors with on-chip clock, memory, and input/output ports, that are being used in domestic equipment. Professional equipment is also benefiting from these devices. Fully synthesized, programmable communications receivers and transmitters are now common, and at a price which would not have been possible without the microprocessor. This is obviously leading to a time when all complex pieces of equipment will incorporate one or more microprocessors. This will have advantages, for even though the complexity of equipment may increase, the operator will no longer have to be fully trained in its operation to use it correctly. The equipment itself can prompt the operator and check for invalid instructions. Microprocessors can give equipment limited 'intelligence', which can be extended if the processor in one piece of equipment is allowed to communicate with either a mini-computer or with other processors associated with it. It is up to researchers and designers to exploit this growing trend and to look at their products not as isolated pieces of equipment but as terminals in a processing network. Unfortunately this trend will be slowed down somewhat by the lack of suitable data communication channels where the processors are more than a kilometre or so apart, or where one or more is mobile. Whilst such links are being planned and integrated into the existing communications network there will be a transitional period where data and the more presently common, voice communication will have to share channels and links. There will also be times when it will not be practical, economically, to provide a dedicated data link between locations, and data will always have to share with voice communication. It is in everyones interest that this sharing of resources does not cause undue inconvenience to any users. A suitable data communication system needs to be created to provide this service. It should be compatible with the limitations imposed by voice-only channels, and it should, preferably, be useable over any type of channel.

This thesis is concerned with data communication over fading channels. These differ from the more common 'static' channels in that the signal received from the originating station is variable in strength, and frequently falls below the level that gives acceptable copy at the receiving station. 'Static' channels in this respect are those including the public telephone network, all types of wired links, microwave and satellite links. All of these channels suffer from various types of noise and interference and may also distort the data signal. Their properties, however, are readily predictable in all normal circumstances and are stable over both long and short periods of time. Fading channels encompass HF, VHF, and UHF radio links when the communicating stations are not in direct line-of-sight. On the high frequency bands the stations are usually fixed, but several hundred or even thousands of kilometres apart, so communication depends entirely on the ionosphere, whose short-term properties are anything but constant. On the VHF and UHF bands at least one of the communicating stations may well be mobile. The mobile channel is the worst of all the fading channels. This is because the rate of fading is usually very rapid compared to that experienced on the HF bands, and the presence of other vehicles close to the mobile makes ignition interference a great problem. This is in addition to the other impairments common to all communication channels.

It is because the VHF/UHF mobile channel is the worst case that it was decided to concentrate on this channel. Any system that works acceptably well over this channel should perform equally as well over other types of fading channel and would, quite clearly, give good performance when used over a 'static' channel. Another reason for working on the mobile channel is that it is very likely, in the near future, that data transmission to mobile vehicles will become commonplace, [32]. There are two major reasons for this.

The first is that while the microprocessor industry has been growing rapidly, usually in the public eye, a similar but less apparent growth has been taking place in the number of users of mobile radio-telephone systems. These systems, which at one time were considered a luxury, are now almost a necessity to people who operate more than a few vehicles. The increasing cost of labour, petrol and diesel fuel, and the vehicles themselves, has forced employers to maximise the efficiency of their employees who are 'on the road'. One easy, and largely successful, way of achieving this is to install a radio-telephone system, [35]. The larger the number of vehicles involved the more attractive this becomes. The resulting increase in the number of users has caused the near-saturation of some existing mobile radio bands in industrial areas.

Most of the messages generated by users are short, and involve the exchange of small amounts of information. Voice communication is relatively inefficient at this, particularly when the time required to initially set up the call is taken into account. If the operator is not present at the time of calling then at best the information is delayed, at worst it may be lost completely. The message may also be incorrectly received by the operator, this fact not being noticed till time and money have been lost. Data communication is ideal for these messages. Messages can be passed automatically once they have been entered, and any messages received while the operator was not present would not be lost or delayed. Suitable encoding of the data would ensure that there were no unacceptable errors in the received message, the coding being chosen to suit the type of message to be transmitted. For example, numeric data would require an error rate many times lower than that acceptable for plain text. Techniques like time-sharing of channels and packet switching can be implemented to further increase efficiency. The transmission of graphics and facsimile also becomes possible. Base stations can operate in full duplex mode, an option which is not usually taken up with voice communication. Using such techniques would ease the congestion of the mobile radio bands and would make optimum use of the new (1981) mobile radio allocation around 900 MHz.

The second reason why data communication to mobiles will increase is that large amounts of data are being held in data bases organised by mainframe computers, [40]. This is particularly the case with public services; both the Police and Fire Service make extensive use of computers at their headquarters, [60]. Hospitals and doctors' surgeries are following suit with patient records. To be able to access this information directly from a mobile terminal is of major importance to the continued well-being of these services. The facsimile facility would, no doubt, interest the Police Force whereas the graphics would certainly be of interest to the Fire Service. With the recent advances and the widespread use of computer graphics, the plans of all large buildings and installations could now easily be held in a data base, ready for immediate transmission to the Fire Service in an emergency. This information can be as detailed as required, can be updated easily, and can be available quickly, all of which are most desirable qualities in information storage and retrieval systems. Other advantages, such as selective calling and data security against third parties, are also to be gained if they are required by the user.

However, this can only happen on a wide scale if low enough error-rates can be achieved over mobile channels. Methods which use forward error-correction would be far too inefficient over mobile channels affected by the fading problem, and no amount of redundancy in the message will cope with a mobile which has accidentally parked in an RF null. To be reasonably sure that the mobile has received the message correctly the mobile must send confirmation of receipt back to the base station, and vice-versa when the mobile originates the message. This protocol is termed Automatic Request Repeat, or ARQ.

Normal ARQ systems involve the transmission of a small block of data and then a delay whilst the recipient sends back an acknowledgement. If no positive reply is forthcoming then the block is sent again, and so on until the message is complete. This is impractical in the mobile case due to the time required for the mobile transceiver to switch from receive to transmit, and then back to receive. Full duplex operation is not practical from a mobile vehicle for reasons which will be explained in a later chapter. Thus normal ARQ would give an unacceptably low overall data throughput. In an attempt to improve matters the block size could be increased, but then block rejections become high due to errors caused by fading.

The system which will be described in this thesis attempts to overcome the problems of conventional ARQ systems by sending many data blocks together. Each data block is independant of any other data block but is labled so that its position in the message is clear. The receiver may receive some blocks and not others. However, only repeats of the missing or incorrect blocks would be required. To allow, primarily, for different fading rates the size of the blocks would be variable. The optimum block size would be chosen by the system by using data obtained from previous communications. As impulsive interference can be a problem with large block sizes, optional forward error-correction by various techniques can be used. Again the system itself would make the decision on whether to use them or not during any particular transmission. When used correctly, forward error-correction can provide limited broadcasting facilities, something which is not normally possible with ARQ systems.

Despite all the advantages that data communication has over voice in normal circumstances, there will always be situations when voice communication is preferable. Any system which did not allow for voice at these times would not gain universal acceptance. Neither would a system which required all new equipment to be fitted to installations, or required new channel allocations to be drawn up. No matter how technically desirable a thing may be, it will not be implemented if its cost is prohibitive. Any major reshaping of land mobile radio would involve an investment totally unacceptable to most users. These observations were of prime importance during this research and so the system which will be described allows voice communication at any time, and requires very little modification to existing voice equipment for it to operate effectively.

It was not the intention to produce a system optimised to a particular channel or modulation scheme. The aim was to provide a reliable data link whatever the conditions. The system would attempt, as far as it was able, to optimise itself to the current characteristics of the channel. The effectiveness of this approach depends on the control the system has over its own operating parameters.

1.2 OUTLINE OF WORK

This chapter has described the reasons why this work was done. The constraints on the final system as to its integration into the existing voice communications network have also been mentioned.

In Chapter 2 the fading radio channel is examined, in particular the VHF/UHF mobile channel. All the pertinant factors concerning the design of a system are mentioned with any particular problem areas highlighted.

Chapter 3 deals with the current state of land mobile radio in the U.K. The types of users and the systems they operate are reviewed, as is current commercial equipment.

The development of the system is explained in chapter 4. There follows, in chapter 5, a detailed description of the particular implementation that was built and tested. The results obtained from this system are given in chapter 6.

Finally, the conclusions are to be found in chapter 7, together with suggestions for further work.

A bibliography of relevant publications follows the references. Throughout the text, references are denoted by the number of the reference enclosed in square brackets.

CHAPTER 2

FADING CHANNELS

2.1 INTRODUCTION

In this chapter the fading radio channel is examined. Section 2.2 mentions, briefly, propagation at HF, VHF and UHF. Section 2.3 deals with polarization of radio waves and some common VHF and UHF aerial types. In section 2.4 the problem of fading is discussed followed by noise and interference in section 2.5. Lastly, in section 2.6, modulation and bandwidth limitations are briefly mentioned.

2.2 PROPAGATION

It is convenient to talk about the propagation of radio signals in terms of waves. The radio wave is a special combination of an electric field and a perpendicular magnetic field, both fields being perpendicular to the direction of propagation of the wave. Radio waves bear some resemblance to light waves; they are both electromagnetic radiation, consequently some terms which are used to describe the properties of light are also used to describe the propagation of radio waves.

Radio waves may be propagated in many ways. The predominant mode of propagation, that is the mode over which communication normally takes place, varies with the frequency in use and the distance between the transmitter and the receiver. The various modes may be broadly classed into three Groups, Ground waves, Tropospheric waves, and Sky waves.

The Ground Wave.

This is a collective term for waves which travel close to the surface of the earth, and so are substantially affected by the terrain between the transmitter and the receiver. The two components of the ground wave are the space wave and the surface wave. The surface wave travels in contact with the ground and must be essentially vertically polarized to be utilised to advantage. The attenuation of this wave is rather high and increases rapidly with frequency, making it useful only below 3MHz. It is this wave which is utilised in domestic broadcasting on the medium and long wave-bands.

The space wave, shown in figure 2.1, is of special interest due its dominating influence in ground wave propagation at VHF and UHF, [9,21]. It is the resultant of two waves, the direct wave and the ground reflected wave. It should be pointed out that neither of these waves actually travel in straight lines, but are refracted downwards slightly by the atmosphere. The waves then appear to partially follow the curvature of the earth, allowing a longer radio line-of-sight path than an optical line-of-sight path, [1]. The amount of refraction varies with local atmospheric conditions, but on average this effect can be allowed for by assuming straight-line propagation and increasing the Earth's radius by one-third when calculating the radio horizon. Refering to figure 2.2(b), the maximum line-of-sight distance is D1+D2 and depends on the respective heights of the transmitting and receiving aerials, h1, and h2.

$$D1 = 4124 \times (h1)^{0.5}$$

$$D2 = 4124 \times (h2)^{0.5}$$

Where D1,D2,h1,h2 are in metres.

This assumes an unobstructed path and the modified Earth radius. Figure 2.2(a) shows a graph of distance to radio horizon, D, as a function of aerial height, h.

Tropospheric Waves

Weather conditions in the atmosphere at heights ranging from less than one kilometre to 3 km are, at times, responsible for refracting radio waves downwards. This makes communication possible over far greater distances than can be covered with the ground wave alone. The effect increases with frequency, becoming significant above 30MHz. This propagation is regarded as abnormal; hence it is not usually a prime consideration when a system is designed. Consequently, interference from transmitters on the same frequency separated by what would normally be regarded as a safe distance, can be a problem when this mode is active. There are other, more exotic, modes of troposperic proagation also giving rise to long distance communication at VHF and UHF. However, they are not normally used for reliable communication.

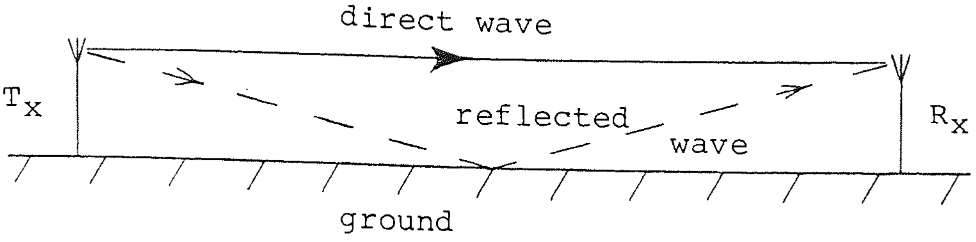


Figure 2.1 The Space Wave

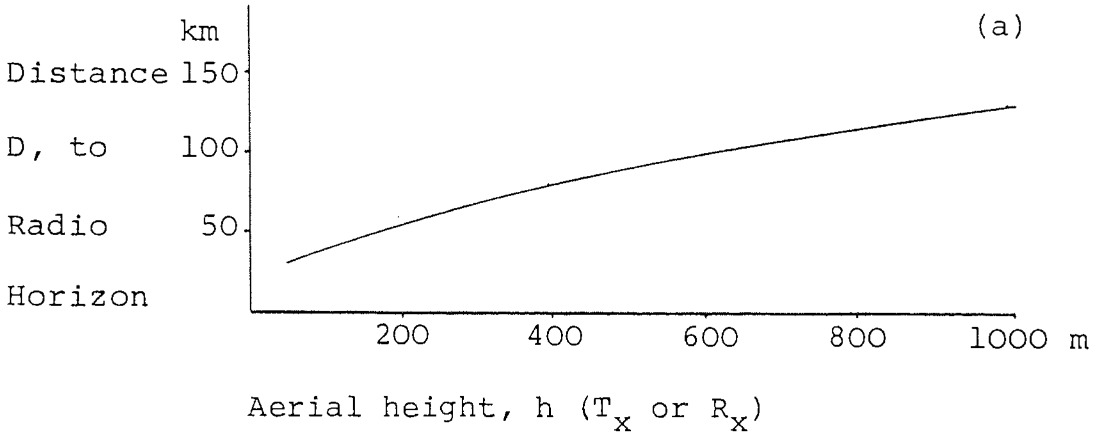
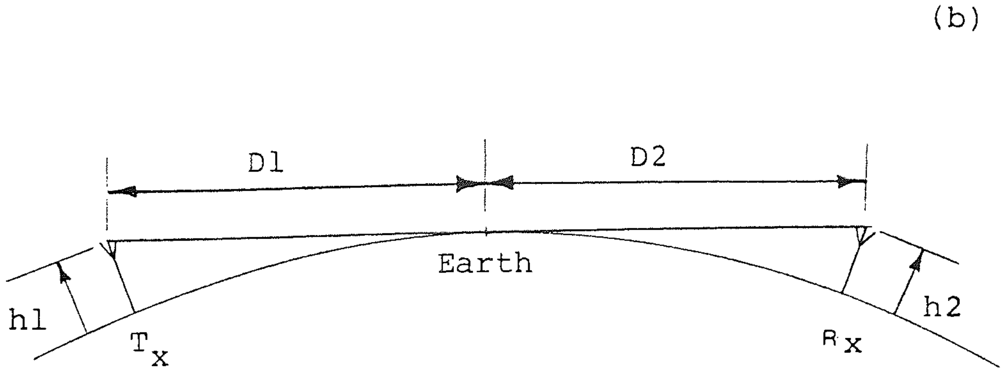


Figure 2.2 Line of sight propagation



Sky Waves

At frequencies below 30MHz almost all long-distance communication is by means of the sky wave, figure 2.3. This is the wave that would normally travel out into space if it were not for the presence of ionized layers in the upper atmosphere which reflect and refract radio waves. There are several ionized layers at heights ranging from 60 km to 420 km. Their individual heights and intensities at any particular location above the Earth vary in daily, monthly and yearly cycles, and over longer periods depending on conditions prevailing on the Sun. Sudden disturbances on the Sun's surface can produce equally sudden changes in propagation by affecting these ionized layers. During such disturbances sky wave propagation may be enhanced or degraded. The layers are unstable over short periods giving rise to fading.

The HF bands are not used by land mobile systems very much. They are used, however, to carry large amounts of data in the form of radio-teleprinter signals over both local and long-distance, particularly Maritime, links.

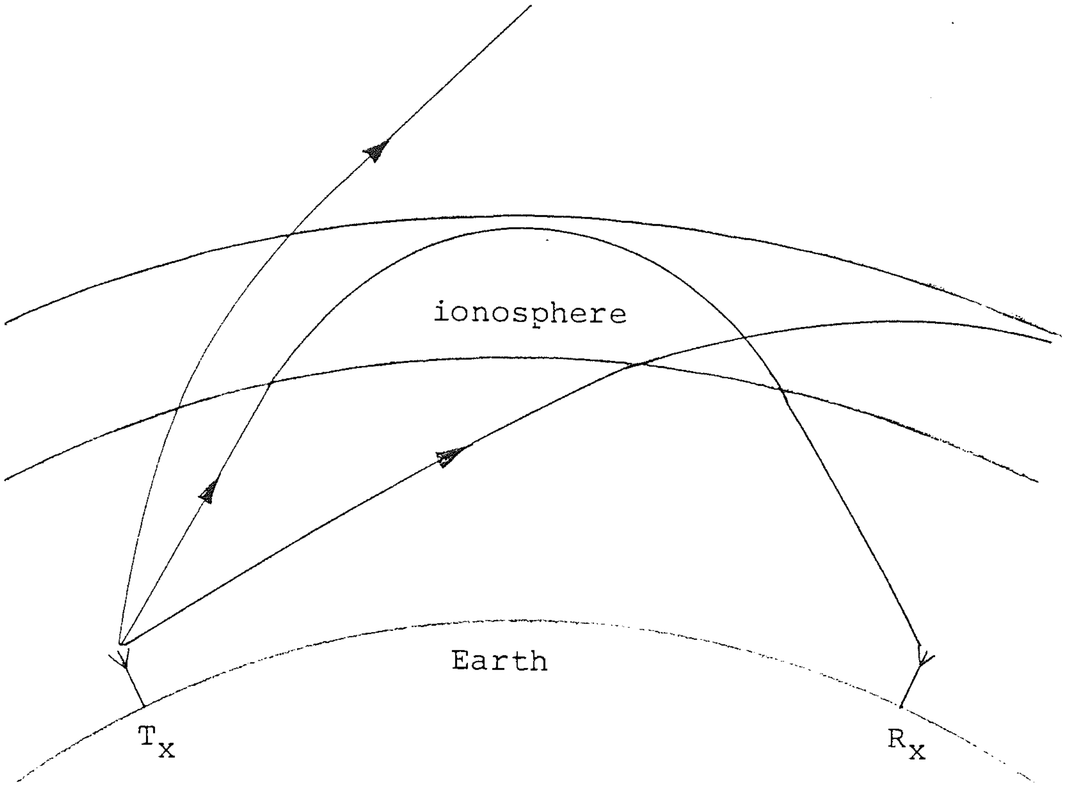


Figure 2.3 The Sky Wave

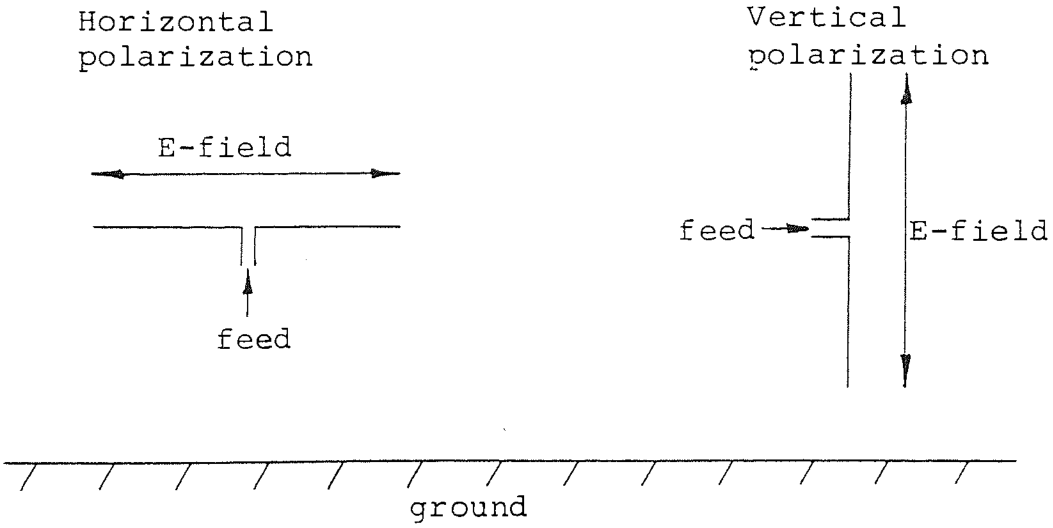


Figure 2.4 Polarization

2.3 POLARIZATION and AERIALS

This topic cannot be covered in any depth in a short section such as this, therefore only that which is relevant to this thesis will be mentioned.

Polarization

This is the orientation of the electric, or E-field, of a radio wave referenced, usually, with respect to the ground. A wave whose electric field maintains the same orientation over an entire radio-frequency cycle is said to be plane, or linearly, polarized. Such a wave may be horizontally or vertically polarized, as in figure 2.4, or slant polarized if it is neither parallel nor perpendicular to the ground. Single radiating elements will produce plane polarization, as will arrays of simple elements fed in-phase. If individual elements are fed out of phase then the E-field vector will appear to rotate, and possibly change in magnitude, over each radio-frequency cycle. The wave is then said to be eliptically polarized. The special case of eliptical polarization, when the magnitude of the E-field vector remains constant throughout each rotation, is known as circular polarization. Although mathematically a special case, it is by far the most common form of eliptical polarization.

Although the initial polarization of a radio wave is determined by the transmitting aerial, it will be changed, to a greater or lesser degree, by the medium through which it passes and by any refraction or reflection it may undergo. This is particularly true of sky waves which may return to earth with a substantially random polarization.

The choice of which polarization is better for a particular path usually attracts differing views, but the choice is often dictated by circumstances. Vertical polarization is used in virtually all mobile installations because of the ease of obtaining omnidirectional coverage with simple and robust aerials. Eliptical polarization is seldom used at HF, VHF and UHF, its main use is at microwave frequencies, especially on satellite links.

Aerials

The choice of aerials for use at high frequencies is wide, but at VHF and UHF the half-wave dipole and its extensions, the yagi, and the collinear array dominate. HF aerials may be simple dipoles or large multi-element arrays, the choice depends on the service area required and the area of land available for the aerial array. Aerials which radiate most of their input power at small angles to the ground are necessary for reliable long-distance communication.

Due to the essentially random polarization of sky-waves returning to earth, similar aerial polarizations at both transmitter and receiver are not essential on medium and long-haul HF links. Indeed there are occasions when it is advantageous to transmit, say, vertically polarized waves and to receive on a horizontally polarized aerial. The installation of orthogonally polarized receiving aerials may help when the signal fades severely. The fading statistics for two orthogonal polarizations are, in the main, uncorrelated. This should provide a combined signal substantially free from severe fading.

The ground wave at VHF and UHF retains its original polarization for the most part, particularly over open country. In built-up areas there may be substantial reflection and refraction of the transmitted wave but in general the polarization remains similar. Therefore it is essential for the receiving aerial to be of the same polarization as the transmitting aerial. Where the signals travel over a line-of-sight path, for example between two base stations, either vertical or horizontal polarization may be used with equal results. Where the path is obstructed horizontal polarization gives marginally better performance. If coverage has to be omnidirectional, as in the mobile case, the situation is entirely different. To achieve true omnidirectional coverage with horizontal polarization is difficult compared to vertical polarization. Such an omnidirectional aerial on a mobile would appear strange, possibly attracting unwelcome attention. Omnidirectional, vertical polarization is readily acheived by the use of half-wave dipoles or collinear arrays at the base station, and whip aerials on the mobile. Such aerials are easy to fit and maintain, and due to their similarity to normal broadcast aerials pass relatively un-noticed when mounted on a vehicle.

2.4 FADING

The primary characteristic of radio channels which makes their use difficult for the transmission of data is their susceptibility to fading. That is, the envelope of the radio signal present at the receiver is subject to large scale amplitude variations. These variations may be up to 40dB or more on mobile radio channels, even within the normal service area of the transmitter. Fading has two primary causes, these are shadowing and multipath interference.

Shadowing

Here, some large body, with respect to the carrier wavelength, is interposed between the transmitter and the receiver, causing a reduction in the signal amplitude at the receiver. This is seldom a problem with ground wave propagation between fixed sites if the body in question is also fixed, ie: hills, trees, and large man-made structures. This results in a permanent decrease in signal amplitude and consequently is not regarded as fading. However, if the body is moving relative to the transmitter or the receiver then fading will occur. At VHF and above the obstruction may be a moving vehicle, whereas at HF irregularities in the ionosphere will be of sufficient magnitude to produce variations in the amplitude of the received signal. The severity and duration of the fade will depend on the nature of the obstruction.

Assuming a transmitter and/or receiver in a moving vehicle then all obstructions previously considered fixed will appear to be moving relative to the transmitter/receiver. The possibility of fading is now greatly increased. The severity of the fade will still depend on the obstruction but the duration will now be related to the speed of the vehicle. Mobile vehicles are in a very unfortunate position in this respect as their aerial(s) are normally only a few feet above ground level. Every obstruction around the vehicle, from other vehicles to nearby hills, will have some effect on the received signal. Hence the requirement for the base station aerial to be mounted as high as possible above the surrounding terrain in order to minimise the effect of these obstructions. Shadowing is not responsible for the rapid fading that occurs over VHF and UHF mobile channels, the fades that it does produce are relativly slow and occur only once for each obstruction, ie: passing a large building or travelling under a bridge.

There is another cause of fading which is related to both shadowing and to multipath interference. However, the nature of the fade it produces is similar to shadowing. This occurs when a radio wave which has undergone reflection and/or refraction arrives at the receiving station with a different polarization to that of the receiving aerial. If the polarization differs by an angle approaching 90 degrees then a severe fade may be experienced, [37]. The severity is greater if there are no significant multipath components of the transmitted wave arriving at the receiver. This type of fade can occur at any frequency from HF on upwards.

Multipath interference

Many of the obstructions just mentioned do not simply attenuate radio signals, they also reflect them. In particular, man-made structures contain large amounts of steel and other metals in their construction and prove to be quite efficient reflectors in this respect. Sky wave propagation at HF can produce multiple reflections due to irregularities in the ionosphere, more than one reflected wave may then arrive at the receiving aerial, [4]. Consequently the signal that is presented to the receiver is a summation of many copies of the original transmitted signal. Each copy will have a different amplitude and phase depending on the distance it has travelled and any extra attenuation it has experienced. The resultant signal produced by the summation will vary with time in both amplitude and phase. The situation is further complicated if either the transmitter, the receiver, or the reflecting body is moving. The individual components will aquire a Doppler shift proportional to the relative motion of the receiver, transmitter and the reflector, if one is present.

Multipath interference at HF is normally attributable to multipath propagation of the sky wave via the ionosphere. However, in extreme cases both the ground wave and the sky wave can be present at the receiver with similar amplitudes simultaneously. Variations in the ionosphere take place fairly slowly, typically fades can last in the order of a few seconds to many times this duration. Fades caused by multipath interference are normally accompanied by severe distortion of the demodulated signal when the radio signal is reduced in amplitude. The amount of distortion increases as the fade gets deeper. Over HF data communication links the spreading in time of the transmitted signal can be just as damaging as fading. Sky wave path lengths can differ by many hundreds of kilometres. A single transmitted pulse will be received as many individual pulses slightly displaced in time, each with differing amplitudes. The nature of this problem is similar to pulse dispersion in other types of channel. Added to this the HF spectrum is not being used for new data links, particularly in the land mobile service, so this problem need not be expanded on.

Multipath interference at VHF and UHF.

Analysis of recordings taken in the VHF and UHF land mobile radio bands using a moving vehicle (receiver) have revealed that the envelope of the radio signal is Rayleigh distributed, [42,43,62]. This is a most important observation as the fades, which appear at first sight to be random, can thus be analysed in an attempt to provide a mathematical model of the channel. Such a model can then be used to test the efficiency of various techniques for providing reliable communication. It should be emphasised that a model is only a tool, any system or technique developed using a model must be given field tests to be sure that its performance is as expected, [14,23,38].

Simple model

This model relates to a moving receiver using vertical polarization, the transmitter and all surrounding objects are assumed to fixed. The book 'Microwave Mobile Communications', edited by Jakes, contains a rigorous treatment of what is to follow, only the pertinant results are presented here. Refering to figure 2.5, this shows one of n incoming waves which are present simultaneously at the receiving aerial. The Doppler shift, ω, in the wave is given by

$$\omega = (2\pi/\lambda).V.\cos\alpha_n \quad (\text{for } V<<c)$$

The maximum Doppler shift in Hz, Fm, is given by

$$Fm = V/\lambda \text{ Hz} \quad (\text{for } V<<c)$$

where V is the vehicles velocity, and λ is the wavelength of the carrier wave. An unmodulated carrier will therefore have an effective bandwidth of twice the maximum Doppler shift. The maximum Doppler shift increases with increasing frequency and increasing speed of the vehicle. The signal appearing at the receiver will be the sum of all the individual signals impinging on the receiving aerial.

The field components are:

$$Ez = E_o \sum_{n=1}^{N} C_n \cos(\omega_c t + \vartheta_n)$$

$$Hx = -\frac{E_o}{\eta} \sum_{n=1}^{N} C_n \sin \alpha_n \cos(\omega_c t + \vartheta_n)$$

$$Hy = \frac{E_o}{\eta} \sum_{n=1}^{N} C_n \cos \alpha_n \cos(\omega_c t + \vartheta_n)$$

Where $\vartheta n = \omega_n t + \phi n$, ωn is the Doppler shift on the nth incident wave, ϕn is a random phase angle uniformly distributed from 0 to 2π, ωc is the carrier frequency of the transmitted signal, η is the free space wave impedance, and EoCn is the real amplitude of the nth wave in the Ez field. The Cn are normalised so that

$$\left\langle \sum_{n=1}^{N} C_n^2 \right\rangle = 1.$$

The three field components may be described as narrow-band, random processes. If the number of individual components is large then the processes approximate to Gaussian random processes. Using the existing literature devoted to Gaussian processes the following results can be obtained for the probability density functions, (pdf).

$$P(x) = \frac{1}{\sqrt{2\pi B_o}} e^{-\frac{x^2}{2B_o}} \qquad \text{Gaussian}$$

$$P(r) = \frac{r}{B_o} e^{-\frac{r^2}{2B_o}} \qquad \text{Rayleigh}$$

For r >= 0 else P(r) = 0

where r is the amplitude value of the fading envelope, and Bo is the mean power ($Bo = (Eo)^2/2$).

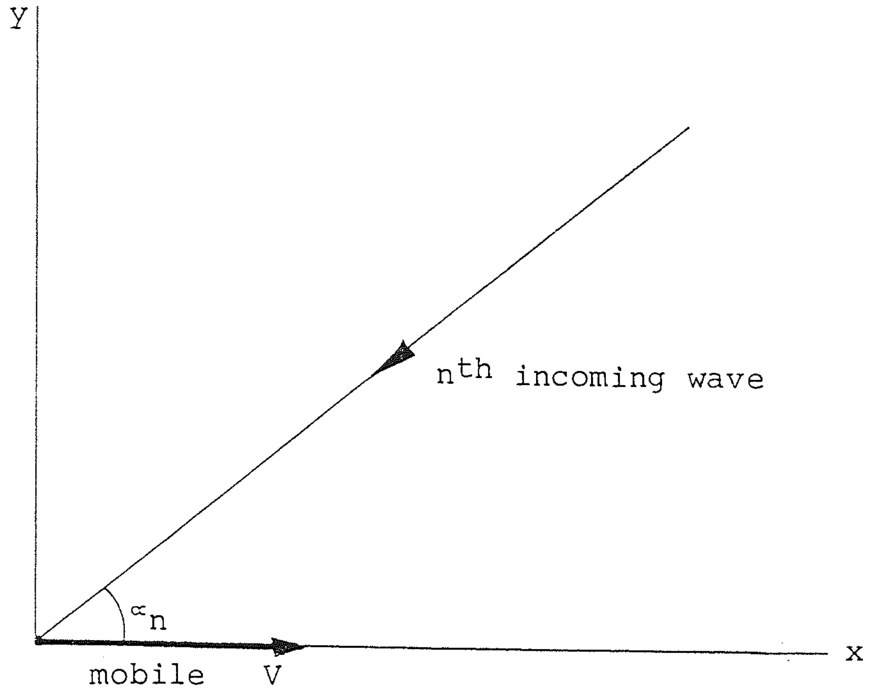


Figure 2.5 Multipath interference

Simple Model

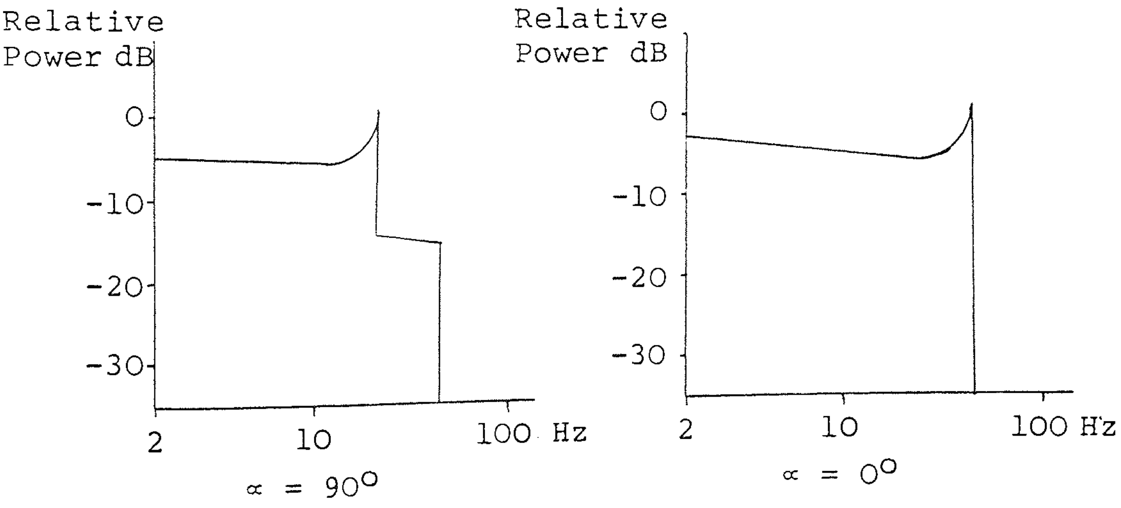


Figure 2.6 Baseband frequency spectrum

Having found the pdf of the amplitude of the signal at the receiver the baseband frequency spectrum can be found. The graphs in figure 2.6 show the theoretical baseband spectrum when a direct wave from the transmitter, (arriving at 0° and 90° to the direction of travel of the vehicle), is added to the input signal. In both the graphs the carrier frequency is 400 MHz and the vehicle's velocity is 30 mph. The maximum Doppler shift is, therefore, 20 Hz. In both cases the spectrum cuts off at twice this figure. The signal envelope experiences shallow fades much more frequently than deep fades. Performing the necessary steps to discover the rate at which the field components cross a particular amplitude level, the maximum level crossing rate is found to be close to 2Fm. Again this is an important result which agrees closely with actual measurements. Knowing the maximum rate at which fades can occur the minimum time that elapses between two consecutive fades can be found. If a block code is to be used to transmit data then the probability that a block will be received without a serious fade should, naturally, be high. Knowing the duration between fades the maximum block size that is practicable at a particular frequency, and at a particular vehicle speed, can be estimated. The usefulness of this result is not limited to block codes, other coding methods require any time-variant properties of the channel to be known for them to acheive maximum efficiency.

Subjective appreciation of fading

Although a mathematical appreciation of fading is most important it is also very useful to actually observe the effects of both shadowing and multipath interference in a real situation. The audible effects of fading differ greatly depending on the cause of the fade.

At high frequencies the fades appear as gradual changes in signal strength with durations which can range from one second to a minute or more. Severe distortion may accompany the fade if multipath interference is the cause. This distortion frequently causes loss of intelligibility even though the signal strength would be perfectly adequate in the absence of the fade. Suppressed carrier and angle modulation is subjectively more tolerant in this respect than amplitude modulation.

At VHF and UHF it is only under exceptional circumstances when various long-distance modes of propagation are active do these frequencies suffer the kind of fading common at HF. Fading is generally found only on mobile channels when vehicle motion is involved. However, it is by no means uncommon for aircraft to cause fading on fixed channels, and any station located near to an airfield could suffer to a noticeable extent. Fading due to multipath interference is characterised by its rapidity. At 100 MHz with a vehicle speed of 30 mph fades can occur at up to 10 per second. At 1000 MHz and 70 mph the maximum rate increases to 230 per second. This rate is now within the audio bandwidth of the receiver. Subjectively this can impair voice communication more than the mathematics might suggest. Also, data communication can become very difficult with less than 5 mS between fades. Fortunately the probability that a fade will be experienced is inversely proportional to its severity, this means that deep fades occur infrequently. Normally the fade will not be severe enough for the signal to lose intelligibility with voice communication, while data communication will still be impeded to some degree. The severe distortion found at HF is much less apparent at VHF and UHF. This is partly due to the widespread use of FM over AM, and the rate at which the signal amplitude changes is too fast for an operator to notice, to a large extent, any distortion present.

2.5 NOISE AND INTERFERENCE

Noise

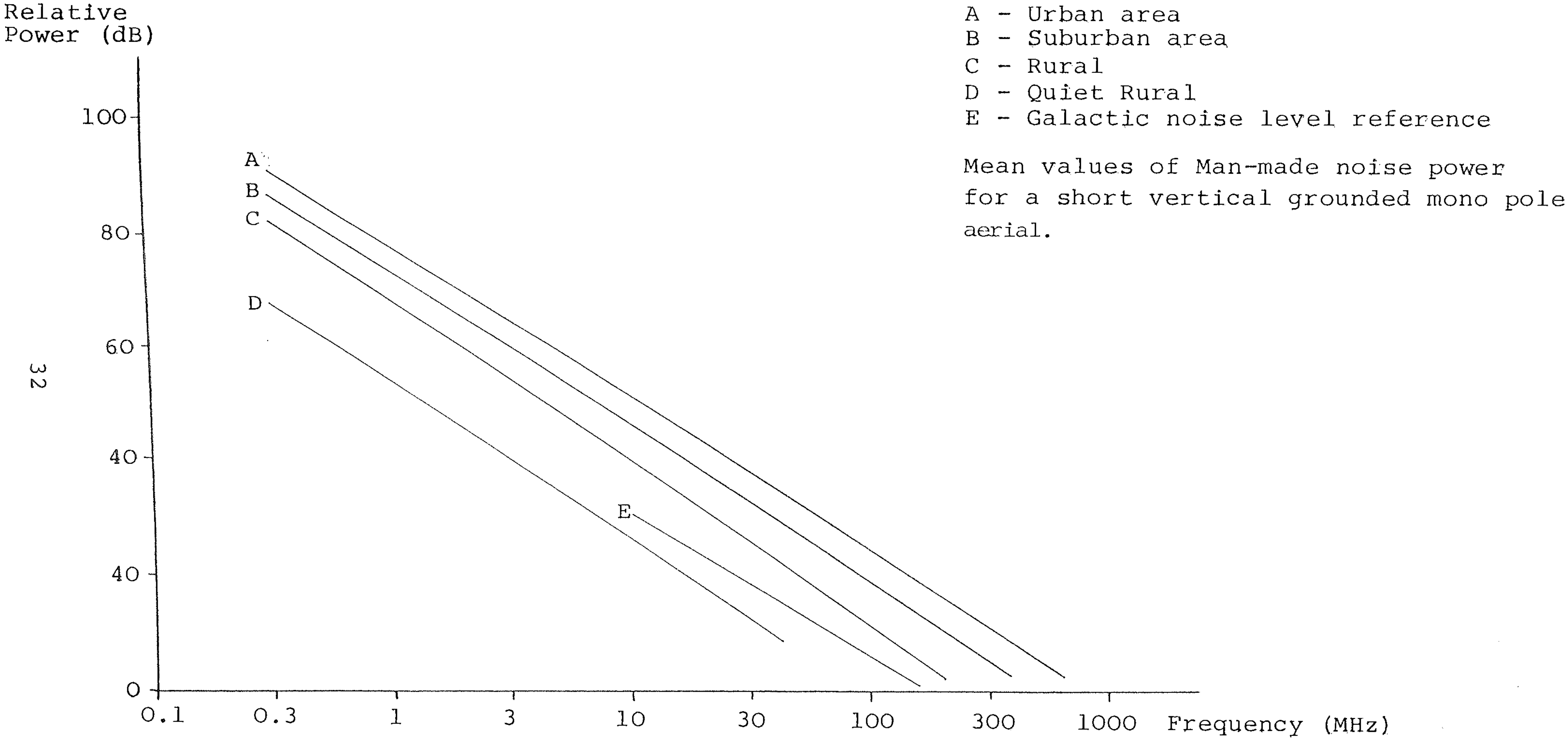
Noise is the one common factor in all communication channels. It enters the channel from external sources and is created by the individual components within a communication system. Usually all unwanted energy which falls within the bandwidth of a system is regarded as noise. Noise in the context of this chapter is only that energy produced by sources other than equipment designed for radio communication purposes. Hence frequency and phase modulation effects and fading are not regarded as noise, neither is interference caused by other users of the radio spectrum.

In radio systems noise originates from three primary sources

(a) Natural external noise
(b) Man-made noise
(c) Noise generated within the communication equipment

Of these, (c) is only a problem from mid VHF upwards, as below these frequencies (a) and (b) predominate. It is caused by thermal and other noise generated by individual components in the signal path. Below a limit this noise can only be reduced by cooling the receiving equipment, fortunately this is not necessary as the continued development of components has resulted in equipment which, for practical purposes, approaches this limit. This noise is essentially Gaussian in character if taken over the limited bandwidths found over communications channels.

Natural noise, (a), is principally due to atmospheric noise over the lower frequencies of interest. It is usually regarded as that produced by lightning discharges, the noise level being a function of the radiated frequency and the proximity of the discharge. This type of noise decreases rapidly with frequencies above 50 MHz. The other principal component of natural noise, cosmic noise, has a somewhat greater bandwidth and is produced by extra-terrestrial bodies. It is greatest in the range 10 to 300 MHz and is usually the limiting factor in the mid VHF region in the absence of man-made noise. As with internally generated noise it is essentially Gaussian in character over limited bandwidths.



Mean values of Man-made noise power for a short vertical grounded mono pole aerial.

Figure 2.7 Man-made Noise

The two sources of noise already mentioned are significant only if the receiver is located away from centres of population. If the location is in, or near, a town or city then, (b), incidental man-made noise will be the limiting factor in the VHF region and will still be significant up to 1 GHz. See figure 2.7. Man-made noise can be produced by many electrical devices, some of which use radio frequencies in their operation. In general however, the devices which cause the most trouble are not those normally associated with radio emissions. Common devices are electric motors, mechanical themostats, gas and oil ignition equipment and vehicle ignition systems. Television receivers and overhead power distribution cables also radiate in the HF region. Above 30 MHz the major problems come from ignition systems and noisy thermostats,etc. These sources of noise have arcs present which, with their associated wiring, form rather inefficient spark transmitters. Thermostats and similar devices can usually be traced and corrected should they cause trouble to fixed receivers. Vehicle ignition noise, particulaly from older vehicles and motor cycles, will continue to cause many problems till laws governing spurious radiation are enforced. Man-made noise is quite different from Gaussian noise as it is usually impulsive in character, [24]. Mathematical modelling of this noise is very difficult and requires actual measurements to be made if useful results are to be expected. The operation of mobile equipment using AM is seriously degraded by this type of noise, hence the widespread use of FM which can offer a good deal of immunity to impulsive noise. Data transmission will be adversely affected whatever the modulation scheme employed, as such it is up to the system to prevent, as far as it is able, the passing of uncorrected errors, [55,56].

Interference

This topic is frequently associated with noise but differs in as much as interference is caused by other users of the radio spectrum. Unlike noise problems, interference can be minimised by correct design, construction and maintenance of the transmitter, receiver and aerial system. Where interference still remains despite precautions then un-authorised use of the spectrum by various people is the most likely cause.

All transmitters radiate unwanted signals, usually these are harmonics of the carrier frequency and, where frequency multipliers are used, sub-harmonics. Transmitters employing a single quartz crystal for frequency control, ie: no mixing, produce predominantly these unwanted signals which can be reduced to an acceptable level by simple filtering. Transmitters also produce noise sidebands around the carrier. These depend to a large extent on the method of frequency generation, the quartz crystal being one of the best in this respect. Modern transmitters employing frequency synthesis may generate unwanted signals which are not harmonically related to the carrier, and, due to the mixing processes involved, some of these signals may be quite close to the carrier. In addition the noise sidebands may be stronger and have a greater bandwidth. With transmitters incorporating linear power amplification there is intermodulation to consider. When two RF signals of slightly different frequency, f1 and f2, are amplified in a 'linear' amplifier then due to inherent non-linearities intermodulation will occur. Unwanted outputs of the form 2f1, 2f2, f1±f2, 2f1±f2, f1±2f2, 2f1±2f2, 3f1, 3f2, etc, will be produced. High order products will be attenuated by tuned circuits but products of the form 2f1-f2, f1-2f2, etc, are very close to the wanted frequencies, and it is not possible to filter them out. A mal-adjusted amplifier can easily produce signals of this sort of sufficient magnitude to cause interference on adjacent channels. As only SSB (and DSBSC) operation requires linear power amplification this problem is confined mainly to the HF bands were SSB is used extensively, but it could become significant at VHF and UHF if SSB is adopted in the future.

Problems associated with transmitters seem trifling when compared to possible problems associated with receivers. Receivers employ mixing techniques having at least one local oscillator/mixer combination, usually two in the VHF and UHF bands, and possibly three or more for specialist applications. Fortunately, the very low powers present in receivers does alleviate the matter somewhat. Problems peculiar to receivers include second channel, or image, interference and overload.

Overloading and blocking may occur if the receiver is operated very close to a radiating transmitter even if the frequency of the transmitter is well removed from the receivers' own frequency. The front end of the receiver may be driven into the non-linear portion of its characteristic thus accentuating other problems and causing a marked loss of sensitivity. In extreme cases the loss of sensitivity can be so great as to render the receiver unuseable. Closely related to this is when two strong signals are present at the receiver input. These may mix in an early stage producing a product within the passband of the receiver or around its intermediate frequency, (IF). Also a strong signal on an adjacent channel may impress its own modulation onto the required signal. A signal of sufficient strength on the second channel, or image frequency, of the receiver will also cause interference. This occurs if the unwanted signal is twice the first intermediate frequency away from the wanted frequency, ie:

IF = F<wanted> - F<osc> also IF = F<osc> - F<unwanted>

This can be minimised by choosing a high first IF and by providing adequate front end selectivity. Receivers in mobile service have to cope with both the weak signals found at the edges of the transmitter's service area and very strong signals such as can be experienced if another mobile is transmitting close by.

Problems are not limited to transmitters and receivers, communal aerial systems can suffer from the 'rusty bolt' effect. The contact of two dis-similar metals, particularly if corrosion has started, can produce a partial diode which can absorb power from several transmitters and radiate a mixing product. This is made worse in damp weather. Due to spectrum re-use there is a limited distance between transmitters on the same frequency, and so there may be places where a mobile may receive a signal from a distant transmitter which is strong enough to cause interference. This can happen extensively if any long-distance propagation mode is active.

2.6 MODULATION and BANDWIDTH LIMITATIONS

In order to transmit information the carrier wave produced by the transmitter must be modulated in some way. Although modulation methods may differ in detail between data and voice communication they are all related to one of three main types of modulation. These are amplitude modulation, (AM), frequency modulation, (FM), and phase modulation, (PM). The following comments apply when using voice and sub-carrier data transmission techniques.

Amplitude modulation

Here the amplitude of the carrier wave varies in sympathy with the instantaneous voltage of the modulating waveform. This is essentially a multiplicative process which produces a carrier and two sidebands. The information, which is carried in the two sidebands, can be recovered by either envelope or product detection. AM is relatively inefficient in mobile use and is degraded by amplitude variations in the received signal such as those caused by fading and noise. Variations of simple AM include double sideband suppressed carrier, DSBSC, and single sideband suppressed carrier, SSBSC, which is usually shortend to just SSB. Both involve the suppression of the carrier wave at the transmitter, which has to be re-inserted at the receiver for the information to be recovered. This complicates both the transmitter and the receiver and is still suseptable to amplitude variations of the signal.

Frequency and Phase modulation

With frequency modulation the instantaneous frequency of the carrier is proportional to the modulating voltage, whereas with phase modulation it is the instantaneous phase which varies. Recovery of the information is fairly straightforward and both have considerable resilience to amplitude variations of the signal, the penalty being the greater bandwidth occupied by the signal.

Bandwidth limitations

With the vast number of users, world-wide, of the HF spectrum, bandwidth is of paramount importance. The additional complications of SSB are readily accepted in return for the saving in bandwidth this mode achieves, though AM is still used in the broadcast segments and in some of the Maritime allocations. Conversely, land mobile radio at VHF and UHF has not, until recently, had such pressure from a large number of users. The requirements of ease of use, easy maintenance, and reduction of interference has favoured FM and PM. In the last few years bandwidth has become an important consideration, the use of DSBSC, SSB, and even AM, is now being discussed for use in new allocations and new systems.

CHAPTER 3

LAND MOBILE RADIO

3.1 INTRODUCTION

Land mobile radio has been in existence for almost as long as radio itself. The earliest form of mobile radio operation dates back to 1901 when radio equipment was fitted into a steam bus. An account of this can be found in Wedlake's 'SOS' The Story of Radio communication. Since such times mobile radio equipment has, fortunately, changed a great deal, yet communication to mobiles is still almost entirely by voice. There are a few exceptions to this, the most common is the use of simple tone-encoders which convert simple, frequently used, messages into a sequence of audio tones which are transmitted using standard equipment, [47]. This situation is about to change.

In recent years there has been a vast growth in the use of computers to store large amounts of information. A mobile operator who wishes access to this information must do so via another operator at the place where a suitable computer terminal is situated. The information appearing on the terminal is passed to the mobile using voice. This is clearly inefficient in terms of both man-power and spectrum utilisation, and is also prone to errors. In addition, some computer information services are being designed to have direct access to the mobile, [5,15,50]. The ability for a mobile to access this data directly will soon be a necessity for the efficient operation of many large-scale enterprises, for example, the Police Force and the Fire Service. It will also be of great use to many other users needing access to such data. However, a total conversion to data communication, with the inevitable loss of voice, would be totally unacceptable to those users and would similarly be rejected by most other users. A total conversion to data communication, although in theory quite possible, is in practice not so. This puts constraints on any new data transmission system hoping to make efficient use of mobile radio resources.

In this chapter the current VHF/UHF mobile radio situation will be reviewed, with particular emphasis on any possible problem areas that may occur when dealing with data transmission. The important question of system hardware, both current and future, will also be dealt with. The book 'Frequency Engineering in Mobile Radio Bands' by Pannell, contains much useful information on the practical aspects of mobile radio in the U.K., as does reference [17]. They should be consulted if a fuller explanation of some of the topics covered in this chapter is required.

3.2 MOBILE RADIO SERVICES

In the context of this chapter and chapter 4, mobile radio services should be taken to include any land radio service (other than broadcasting) in which one or both ends of the communication circuit can move whilst communication is taking place.

Mobile radio services can be classified in many ways; as far as overall system design is concerned there are two major classifications. The first is by geographical coverage, the second, by the number of users.

Classification by geographical area

(a) Radiating cable systems
(b) Local systems
(c) Large area systems
(d) Paging systems

Classification by number of users

(e) Private single user
(f) Private multi-user
(g) Public services
(h) Radio links

(a) Radiating cable systems

These give very limited coverage over a small, well defined area. This type of system is usually found in industrial complexes, tunnels, and large buildings. The frequency used is in the UHF spectrum. Due to the limited range of the system the frequency can be re-used over short distances. The hardware involved in a radiating cable system is normally the same as that of a UHF mobile system, with the exception of the base station aerial which is replaced by a 'leaky' feeder. Systems for special applications, including data transmission, can easily be implemented as compatibility with other systems is not required.

(b) Local systems

These give radio coverage over a geographically small area, roughly a large factory site to part of a city. Operation is at UHF with a power level and aerial system chosen to provide only the coverage that is required. Typical examples of these systems are university campus security, and inner-city radio coverage for police on foot.

(c) Large area systems

Until recently, these have been the most common type of system. Coverage may range from a large town, or city, to a whole county. A well sited VHF transmitter is necessary to provide this size of service area. However, with increasing pressure on the VHF allocations large areas may have to be covered by several UHF transmitters, each one serving part of the new service area required. This is naturally more expensive than with a single VHF transmitter, but it may be the only way new users can achieve the coverage they require without interference to and from other users, [61]. This approach does have some advantages. If all the UHF transmitters are co-channel and their individual service areas overlap to some degree, then the possibility of a mobile encountering a 'dead' spot is very much reduced. The deliberate breaking up of an existing VHF service area in this manner is now done on occasions for this and related reasons, [31,45].

(d) Paging systems

These can range from small local systems within large buildings to national schemes. Two-way communication is not usually required, the public telephone service forming part of the system when communication back to the control station is necessary. Signalling is done by transmitting tones on a common transmit frequency, each user having a receiver that responds only to a particular sequence of these tones, [36].

(e) Private single user

In this situation there is a single user, such as a small company or shop, with a single base station and a few mobiles or portables. The coverage of the system will usually be local. The low overall utilisation of the channel inevitably means that it will have to be shared with other similar users. This can lead to problems if one or more users begin to use the channel excessively to the detriment of the other users. In addition, tone signalling has to be used to ensure that only the required mobile responds to a call. Security on such systems is not particularly high.

(f) Private multi-user

Users such as Television rental companies, taxi and security services often comprise this category. The coverage they require varies, but the larger users often have several base stations controlling many mobiles and portables. These users will be allocated several exclusive VHF and UHF channels.

(g) Public services

Into this category fall the Police, Fire and Ambulance services, County councils, Water authorities and the like. Coverage and allocations will be similar to (f), plus the need for some services to be integrated into larger overall schemes.

(h) Radio links

This category encompasses systems comprising a single voice channel to multiplexed systems carrying several thousand channels. They are not limited to voice, many radio links carry telemetry between remote sites and central control stations. Their common factor is that only point-to-point communication is involved. As such, problems of fading and interference seldom occur.

3.3 CURRENT SYSTEMS

In this section it is intended to give a brief description of the various mobile radio voice systems currently in use in the U.K. Due to the nature of this thesis only the more common types can be included.

Irrespective of its classification in the previous section, systems can immediately be divided into two quite different groups. Those which use a single frequency for both transmission and reception, and those which use two different frequencies. The system's protocol is very much determined by whether it uses one or two frequencies. The system hardware also has to be chosen and installed to suit.

Single-frequency working

These use the same frequency for both transmission and reception of messages. The great advantage of this type of working, known as single frequency simplex, is that any mobile unit can be heard by any other unit that happens to be in range. This can be of great assistance in sparsely populated areas and in moving convoys. Messages can also be relayed to a mobile in a poor location relative to the base station by a more favorably situated unit. The main disadvantage of this working is that a receiver in a mobile situated close to another transmitting mobile can be adversely affected by such a strong signal as explained in section 2.5. In addition, this working precludes simultaneous transmission and reception by the base station. As will become apparent, this requirement is most important in the data transmission system to be described. To be fair to the manufacturers of mobile radio equipment, the susceptibility of modern receivers to overload is much lower than with older, especially solid state, equipment. However, this only makes the second disadvantage more apparent since full duplex, ie: simultaneous transmission and reception, is now quite practicable with the all-round improvements in radio equipment.

Two-frequency working

This is usually known as split frequency, half duplex, working. The base station transmits on one frequency, to which all the mobile receivers are tuned, and receives on the mobiles' common transmit frequency. The separation between the two frequencies is usually around 4 to 5MHz, although closer spacing can be achieved with cavity filtering. Problems of receiver overload and blocking are very much reduced with this type of working, and full duplex working of the base station is easily achieved. Full duplex operation by the mobile is very seldom an advantage with voice, and the increase in equipment cost and maintenance it entails makes this impractical at the moment. Full duplex operation of the mobile is a big advantage with data transmission, but the decision as to whether the cost is justified must be left up to the user.

Full duplex working allows the base station to be used in repeater mode. That is, the audio from the receiver is fed into the transmitter allowing mobile-to-mobile communication. It is frequently the case when two mobiles wish to communicate that they cannot contact each other direct but are well within the service area of the base station. In normal operation it is usual for the base station to transmit an 'engaged' signal when it is receiving a message. This is to inform other mobiles that their transmit frequency is in use and so should not call in.

Hardware requirements

The choice between single and two-frequency working affects the hardware requirements at the base station, particularly when the base station site is shared with other users. The noise sidebands from a transmitter can interfere with receivers on the same site, even if they are tuned several megahertz from the transmitted carrier. The transmitter power, the physical separation of the aerials and the frequency separation are just some of the parameters that have to be examined. The complex inter-relationships between systems on communal sites are now well known for voice systems, [16], it would be wise therefore, not to make any major changes in the hardware of existing systems when considering data transmission.

Channel occupancy

Another factor to be considered is channel occupancy. With single frequency working the channel is fully occupied while a message is being passed, ignoring the short time taken for the transceivers to change from receive to transmit, and back. The limit of channel occupancy is thus close to 100% if messages are being passed continuously. In the case of split frequency, half duplex working, two frequencies are in use when a message is being passed, but only one frequency carries information at any one time. The maximum channel occupancy, taken over both the channels, cannot exceed 50%. This represents a considerable loss of channel utilisation in return for the benefits of split frequency working. Ideally the base station should work in full-duplex mode, transmitting and receiving independant messages simultaneously. The occupancy limit can now be higher than in the single frequency case as no time is required for the base equipment to change from receive to transmit. With a single operator at the control point such high channel occupancy is out of the question. Two or more operators could be employed but this would, no doubt, lead to difficulties.

A suitable data transmission system would have the capability to support full-duplex operation, each station being controlled by a mini-computer or several microprocessors. The entire system must be planned with this in mind from the outset if the two-channel occupancy is to aproach its limit.

Radio links

The service area of a radio system depends on, amongst other things, the height of the aerial at the base station. Where only limited coverage is required then the aerial height necessary can often be achieved by a mast actually on the premises of the user. If this is not possible, or if the coverage required necessitates the positioning of the aerial on a local high spot, some means of connecting the base station to the point of control is needed. This is usually done by a land-line or radio link. Land-lines to the site may already be available or can easily be installed, in which case there is no problem. Some sites cannot be reached satisfactorily by land-lines for geographic or financial reasons, so a radio link has to be used.

A radio link is a simple two-way radio communication system operating simplex, half, or full duplex. This can be done directly if a line-of-sight path exists, otherwise repeater stations will be required. The link has to carry the information transmitted and received by the base station and, possibly, some control signals as well. If data transmission is envisaged for an existing system the characteristics of any land-lines or radio links will have to be taken into consideration if it is intended that they should be retained.

From this it is clear that should even a minor change in the specification of a large, existing, system be deemed necessary to accomodate data communication, the cost of modifications to the equipment may be much higher than the user is prepared to accept. The use of existing equipment for data communication without modification may not be technically desirable, but it is certainly financially advantageous.

Spectrum re-use

A frequency used by one station may also be used by a different station, providing that it is some minimum distance away from the first. This distance, called the re-use distance, is a function of the stations' aerial height and gain, transmitter power, and the intervening terrain. Figure 3.1 shows the situation when two stations use omnidirectional aerials over substantially flat terrain. The coverage range encloses the normal service area of the transmitter; inside this area the field strength of the transmitted signal is sufficient to produce an acceptable signal-to-noise ratio at the output of a receiver. At a certain distance the field strength will have dropped to a low enough level that it will not, with normal propagation, cause undue interference to other users on the same frequency. This distance is termed the re-use range. Note that the larger coverage range of A2 over A1 results in a larger re-use range. The coverage range of A1 could be increased slightly without affecting the re-use distance, D. Thus the re-use distance is the greater of:

coverage range of A1 + re-use range of A2

coverage range of A2 + re-use range of A1

Excessive coverage areas can cause the re-use distance to be unnecessarily large, resulting in under-utilisation of resources.

Figure 3.2 shows how the intervening terrain can affect the re-use distance to the benefit of users, allowing a closer spacing of transmitters. However, obstacles are not always situated between service areas. Indeed, they are usually found well within a required service area. If coverage was required on both sides of the obstacle in figure 3.2, then there would be two possible alternatives.

(1) Use one VHF transmitter on top of the obstruction.

(2) Use two UHF transmitters, one either side.

Although (1) would be cheaper to install and maintain than (2), the re-use range of the transmitter would be very large. Consequently, (2) may be the only solution when channel re-use is considered, as is becoming more the case.

The coverage and re-use ranges of a transmitter can be greatly modified by the use of directional aerials. The polar diagram of the aerial now has to be incorporated into the estimations of these ranges. Unfortunately, the supporting structure on which the aerial is mounted will affect its polar diagram to some degree. This can be allowed for but on-site measurements are necessary to confirm any estimations. The polar diagram of a high-gain aerial may change with time because of corrosion and changes in adjacent aerial positions. This also has to be allowed for.

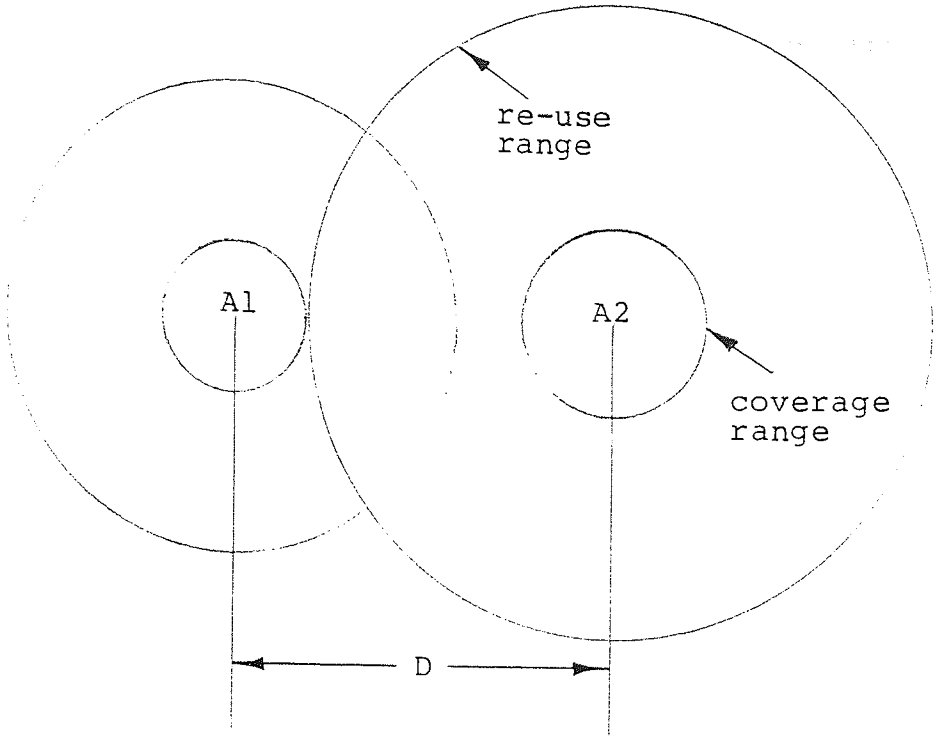


Figure 3.1 Channel re-use distance

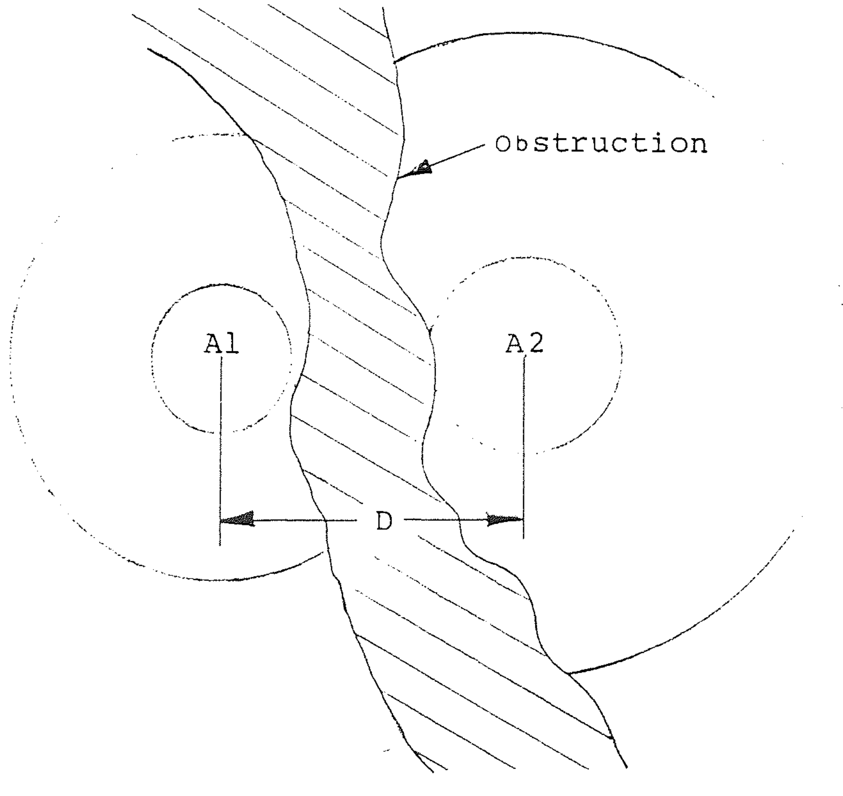


Figure 3.2 Effect of terrain on re-use distance

3.4 MOBILE RADIO EQUIPMENT

Two major requirements of mobile radio equipment are that it should be reliable and simple to operate. The latter requirement is straightforward enough but the former is often difficult to achieve. This is because most mobile radio sets have to operate under very adverse conditions. When mounted in a vehicle they are subjected to severe mechanical vibrations and shocks, and to extremes of temperature in both the long and short term. Added to this their supply voltage can vary from less than 10V, to over 14V on a typical 12-volt electrical system.

Nearly all mobile equipment now available is the result of many years experience by the manufacturer in producing this type of radio equipment. Modern technology may increase the reliability of this equipment still further if it is applied correctly. Drastic changes in the design of mobile radio equipment are not to be recommended for fear of producing a set which works fine in the laboratory, and yet often breaks down in the field. An on-going research and development programme is required, with plenty of feed-back from both customers and field engineers, if a high level of reliability is to be maintained.

This is one more important point to be borne in mind when thinking of data communication systems. They too must retain, and preferably exceed, this high level of reliabilty. The more new designs that have to be introduced then the greater the cost to both manufacturer and consumer, and the greater the possibility of a serious design fault being revealed. It is advisable, therefore, to retain as much existing equipment as possible when transmitting data, at least initially. 'State-of-the-art' designs more suited to efficient data communication can be introduced when they have proved to be sufficiently reliable in the field.

Manufacturers have been criticised in the past over the long delay before new technology appears in their products. The reasons are primarily due to those mentioned above. Hopefully, with LSI techniques proving reliable in their own right, and modular construction speeding up on-site fault diagnosis and repair, this delay will be considerably shortend in the future.

In the remainder of this section the major items of equipment used in mobile radio communications will be discussed. Items of special interest in relation to data transmission will be so mentioned.

Aerial systems

The introduction of data communication will not affect the hardware of aerial systems at all. In general, the problems encountered in this respect are the same for both voice and data. Changes may have to made to some aspects of aerial planning, but the exact nature of these changes will not be known till some trial systems have been running for some time.

Aerials currently in use on vehicles are various whip aerials, quarter wavelength, 5/8 wave and 7/8 wave, etc. At the base station, ground planes, half-wave dipoles, yagis, and collinear arrays are all used extensively. The choice depends primarily on the service area required, the polarization is usually vertical.

Summarising aerials:

Mobile

Quarter-wave whip: Simplest aerial type for a given frequency, easiest to construct and mount, good omnidirectional characteristics when suitably mounted, used as 0dB reference for other whip-type aerials.

5/8 wave whip: Larger than quarter-wave and requires loading coil, similar characteristics, has around 3dB of gain relative to quarter-wave.

7/8 wave: As 5/8 wave, larger size limits it to higher frequencies, gain can be around 4.5dB relative to quarter-wave.

Base station

Ground plane: As quarter-wave whip with artificial ground plane.

Half-wave dipole: Very common aerial, easy to construct and mount, omnidirectional characteristics if mounted clear of metalwork, predictable modifications to polar diagram if mounted close to supporting structure, used as 0dB reference for other aerials.

Yagi: Complex design and construction, can be highly directive and so have high gain, polar diagram is not modified to the same extent as the dipole when mounted on supporting structure.

Collinear arrays: Multiple driven elements so can be difficult to feed and mount, offers gain and more control over H-plane polar diagram than the yagi.

Mobile transceivers

Until the development of suitable high-frequency transistors all mobile transceivers used valves. The necessary HT supplies were generated by an invertor incorporating a mechanical vibrator. Later, the vibrator was replaced by high power germanium transistors. Valve equipment was large, heavy, and very inefficient in terms of power consumption. They were however, reasonably reliable and capable of withstanding considerable electrical mis-use. The receivers were insensitive by today's standards, but suffered little from problems of blocking and overload. Selectivity was not a strong point, but this was more attributable to the absence of crystal filters than to the use of valves.

The next generation of sets had solid state receivers and modulators. This reduced the receive-only power consumption by a factor of between 10 and 50, and reduced the size of the set by almost half. These early transistorised receivers suffered from blocking and overload problems; a mobile transmitting high power could destroy the front-end devices in a receiver parked close by. The transmit sections still used valves, semiconductor devices of sufficient reliability and robustness were not yet available.

Today all mobile transceivers are fully solid state, and perform significantly better than the earlier sets. One component of these sets, the quartz crystal, has remained unchanged throughout, and is only now giving way to new techniques. Both single and multi-channel sets use one transmit crystal and one receive crystal for each channel. Each crystal has to be made to give the required frequency, so to change channel one or both crystals have to be changed. Coming onto the mobile radio scene now is the frequency synthesizer. Although this too uses a crystal for ultimate frequency control any standard channel frequency can be obtained from one reference frequency. A simple adjustment is now all that is necessary to change the frequencies in use. The digital techniques employed in these synthesizers are ideal for interfacing to other logic and to microprocessor systems.

Base station transceivers

The base station transceivers in use are very similar in electrical design to those used in mobiles, perhaps even using the same circuit modules. They usually operate in a better environment than the mobile, but they are often in use 24 hours a day. On remote shared sites there may be several base stations in a small room with little ventilation. In the summer months operating temperatures can be very high, putting considerable stress on the power components in these sets.

The equipment used for radio links is somewhat similar to that used at UHF base stations. Transmitter powers are usually lower and aerials are high gain types. Large users may have microwave links, but these are outside the scope of this thesis.

Future trends

Together with the change from valves to transistors, there has been a change from AM to FM and PM. Increasing pressure to reduce channel spacing from 25 kHz to 12.5 kHz at UHF and from 12.5 kHz to 6.25 kHz at VHF is now threatening the continued supremacy of FM/PM. It is widely believed that the resulting reduction in signal bandwidth would obviate some of the advantages of these modes. AM is once again being considered, but it is much more likely the reduced and suppressed carrier modes will take over in the future.

Data transmission, which usually uses subcarrier FSK techniques, is likely to use subcarrier PSK or direct FSK at the carrier frequency in the future. SSB is not an ideal mode for data transmission; the local carrier signal necessary for the coherent demodulation of SSB is likely to be difficult to generate in the mobile situation due to fading and Doppler effects. One mode which may become popular for both voice and data communication is VSB. Its use overcomes some of the problems of using SSB with only a small increase in signal bandwidth. If VSB is found to be acceptable for data communications then it will be an obvious choice for serious consideration in future systems.

Frequency synthesizers now make it easy to change channel quickly, [59]. Transceivers having a choice of several hundred frequencies within a band 1 MHz to 5 MHz wide, depending on their centre frequency, are now readily available. This capability should not be ignored in the design of future systems. One possibility, which may be incorporated in the system to be described, is to let many independant users share several channels. Unlike the present systems where each user can only operate on one channel, the users' equipment would search for an unused channel when communication was required, in a similar way to telephone systems. The complex operation of such a system could easily be handled by a minicomputer at the control station and by microprocessors in the mobiles.

The incorporation of a microprocessor in a set allows it to be considerably more complex than a standard set, yet be as simple to operate. As vehicle electrical systems also become increasingly complex and begin to incorporate microprocessors, there is no reason why all the microprocessors in a vehicle could not communicate with each other. The processor in a mobile transceiver could learn of the status of the mobile automatically, allowing it to adapt faster to the situation.

CHAPTER 4

SYSTEM DEVELOPMENT

4.1 INTRODUCTION

In this chapter the information presented in the previous two chapters is used as a foundation for the development of a data transmission system. The system is primarily intended for use over fading radio channels, in particular the mobile channel. It can also be used over any fading or non-fading channel where highly reliable data communication is required. The system adapts, as far as it is able, to the conditions on the channel. Data throughput will fall should the channel deteriorate, but data integrity remains at the level set by the coding employed, largely irrespective of channel variations.

As has been stated the worst fading channel is the mobile radio channel and it is for this channel that the system has been specifically designed. The two prime considerations during the system's development were that it should be compatible with current and future mobile radio hardware and channel bandwidths, and it should allow both voice and data communication with the same equipment. The system itself is transparent to the binary data presented to it, but it does assume that the data will be in 8-bit bytes. This one assumption was felt to be quite justifiable as most microprocessors operate with 8 or 16 bit data busses. In addition, there is increasing use of the ASCII code for the transmission and storage of textual material. In this code each character is represented by seven data bits plus a parity bit.

The system is microprocessor based with many of its operating parameters under software control. In the following sections the development of the system will be described, leading to the single-channel, single-user system that has been built. A proposed multi-channel, multi-user system will be described later.

4.2 ERROR CONTROL TECHNIQUES

When a communications channel introduces errors in the data passing over it there are two basic strategies available to the system designer to reduce these errors to an acceptable level. Although both methods introduce redundancy into the data before transmission, in other respects they are quite different and are not usually interchangeable.

The two strategies are:

(1) Forward error correction (FEC)
(2) Automatic request repeat (ARQ)

Forward error correction

In this strategy redundancy is introduced into the data by the addition of check bits. Simple FEC coding can be achieved by an extension of the familiar parity bit that is added to a block of data. Whilst detecting whether there is a single bit in error (more exactly it will detect any odd number of errors), it cannot give any indication as to where in the data block the error may be. However, the addition of more parity bits, each bit checking parity on selected data bits only, will give enough information to pinpoint the actual bit in error. An example of this type of coding, often given in textbooks, are Hamming codes. Codes can provide single or multiple error detection and correction depending on the amount of redundancy that is added to the data block. The larger the data block then the more efficient these codes become, but as the probability of multiple errors increases with block size there is ultimately a trade-off. It follows, therefore, that there will be a finite, optimum block size for a given channel. If the data is not in blocks a similar type of coding can still be used, a convolution code operating on continuous data is suitable for this situation, [29].

FEC codes have one major disadvantage, for them to work efficiently any errors introduced by the channel should either be few and randomly distributed or occur in bursts not exceeding a certain length, depending on the code. Information may well be lost or corrupted if this is not so. Should the channel fail partially or completely, then information will certainly be lost as the encoder has no way of knowing if the decoder is receiving an accepable signal. Over fading channels this happens to an appreciable degree, so FEC is not useable under these circumstances.

Automatic request repeat

This strategy is possible only when a two-way channel is available,and this is its main disadvantage. Instead of assuming that there will be errors, as in the case of FEC, ARQ operation assumes that there is a good possibility of no errors. Thus the data block is only protected by a secure error-detection code. If the decoder finds an error in a received block the data is discarded and a request is made, via the return channel, for a repetion of the same block. This continues till the block has been received with no errors. Because of this two-way, fail-safe, communication between the encoder and decoder this technique is ideal for use over fading channels. It was this technique that was chosen as the basis for the communication system.

ARQ operation will now be dealt with in more detail and a modified form of ARQ, termed SRT-ARQ, described.

4.3 SELECTIVE RE-TRANSMISSION

Conventional ARQ

Conventional or stop-and-wait (SAW) ARQ, [6,11], operation involves the transmission of a single block of data from A to B as in figure 4.1. 'A' then waits for a positive acknowledgement, ACK, from B indicating that the data was received with no detected errors, or a negative acknowledgement, NACK, indicating that errors were detected. If the acknowledgement is not received within a reasonable time, or it is corrupted, the same block is re-transmitted as a matter of course. Edwards, [18-20], has shown that where full duplex operation is possible then this approach has much to commend it. Should only a simplex or half-duplex channel be available the efficiency of this approach drops dramatically. The reason for this is a purely practical one. Viz, the efficiency of conventional ARQ increases as the block size is reduced, [13,41]. If the equipment in use has to change from transmit to receive and back for each acknowledgement, then for most of the time this is exactly what it will be doing and not transmitting the data. The time taken for a radio transceiver to change from transmit to receive, (or receive to transmit), can be anything between 100 mS and one second. This is at least sixty times the transmission time for one block of the size and bit rate used by Edwards, [19].

Selective Re-transmission ARQ

Various improvements have been suggested for SAW-ARQ, [3,49,53]. Further improvements have recently been proposed by the author, [7]. These have since been echoed by Turney, [58]. The improvements relate to the transmission of several data blocks successively, preceeded by a header. In this scheme the decoder treats each block individually until an acknowledgement transmission is to be made. Now, any re-transmission request is for only those blocks containing errors. This is called Selective Re-transmission (SRT) ARQ. Turney has already produced some results obtained from a simple implementation of this technique, [58]. This new approach improves data throughput over simplex and half-duplex channels quite considerably as the transceivers are required to change mode far less often.

Automatic selection of block length

For a constant un-detected bit error rate (BER) over the entire system there is an optimum data block length which achieves maximum data throughput. This is demonstrated in chapter 6 and reference [13]. If the channel's BER varies widely, as with the mobile channel, then the block length should also be varied in sympathy to maintain maximum throughput. Any ARQ system, be it SAW-ARQ or SRT-ARQ, that uses a fixed block length will not be performing optimally over a mobile channel for much of the time. A means of automatically changing the block length whilst communication is taking place is required when the channel BER varies.

In this system automatic selection of block length is acomplished by a microprocessor. The throughput and block error-rate of the system during each transmission are calculated and compared with the optimum values for the block length in use. If they are different to an appreciable degree the processor will adjust the block length. It is, in practice, necessary to take averages to avoid the system changing block lengths too often.

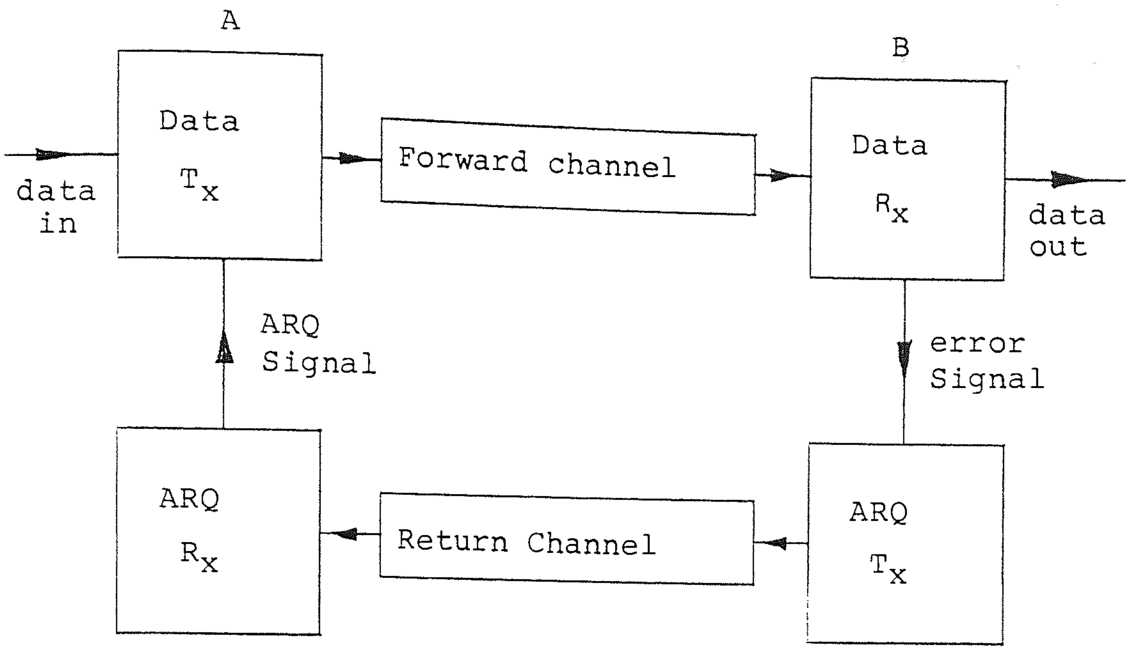


Figure 4.1 Conventional ARQ

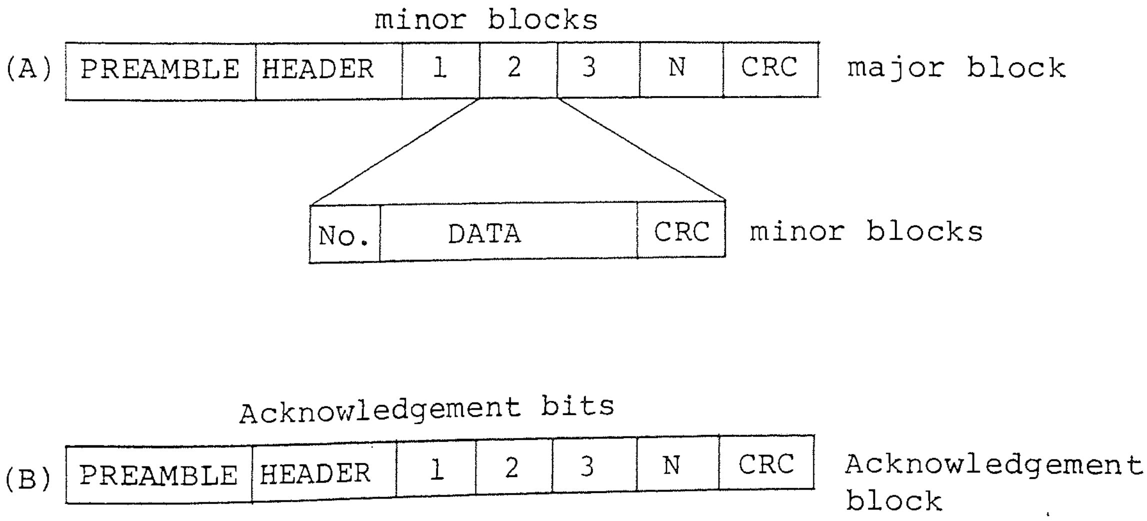


Figure 4.2 SRT-ARQ

Informing the processor directly of a low receiver signal-to-noise ratio can be beneficial, [19]. However, adequate design of the system software can give similar results and requires no modification of the equipment. More importantly, the current status of the mobile can be very useful in helping the processor choose the optimum block size of the initial transmission. For example, whether the moble is actually moving and, if so, at what speed, if the vehicle's engine is running and if the vehicle is occupied.

Block format

One of the many ways the data blocks can be formatted is shown in figure 4.2. The major block, A, consists firstly of a preamble. This is to signal the start of a major block. It is a pattern of bits that cannot legally occur anywhere else in the block. Next is a header which gives information concerning the minor block size, the number of minor blocks and their coding scheme. In multi-user systems a code indicating the recipient can be included here. The header information is protected by a cyclic redundancy check (CRC) to ensure the information is received correctly before it is acted upon. The minor blocks follow, each one numbered and having its own CRC. This check is always present even if additional coding is present within the minor blocks. Finally there is a CRC for the entire major block.

The acknowledgement, B, follows the major block format quite closely, but now the minor blocks are just single bits the value of which tells the encoder whether errors were detected. These bits can be set according to the minor block's CRC alone, or after further decoding of data contained in the minor blocks.

Additional coding

For most applications the checks within the blocks will be sufficient for an acceptable error rate when transmitting plain text and other information that contains natural redundancy. For more secure transmission, particularly over poor channels, [7], additional encoding can be incorporated into the minor blocks before transmission. In this system it is possible to have a different code for each minor block. The only restriction is that the decoder must be capable of handling all the codes in use. Should the decoder be unable to handle a particular code, then providing it 'knows' at least one secure code, it can be re-programmed by the encoder via the channel. The freedom of choice of additional coding schemes this system supports is a major departure from other mobile radio data communication systems. This flexibility is due to the decoding of the data blocks being be done by a microprocessor. Should the workload caused by this extra decoding become too great for the system's microprocessor, one or more extra processors can be employed for this, and other, specific tasks. A commercial implementation of this system would most likely make use of multi-processor techniques to simplify system design and maintenance, in addition to the reason just mentioned.

One advantage of this system which appears largely as a by-product is its capability for data scrambling to preserve security. Complex codes can be changed quickly enough to prevent third parties having time to analyse the code in use.

Broadcast capability

The only other major disadvantage with ARQ systems is their lack of a broadcasting capability. In order to send the same message to many mobiles each one has to send back a positive acknowledgement before the message can be transmitted to the next mobile. This can be very wasteful of time and equipment. In this system the use of FEC codes within short minor blocks and over the whole of the major block can give a high probability of all the mobiles receiving the message correctly first time. Those that miss parts can apply for a re-transmission in the usual way. In extreme situations the message can be repeated to all the mobiles.

4.4 MODULATION SCHEMES

As in most data communication systems this system transmits information via a carrier wave and not at baseband. Therefore, some form of modulation technique has to be employed. Bearing in mind that the carrier wave is in the VHF/UHF spectrum there are two possible methods.

(1) Subcarrier techniques

(2) Direct modulation of the RF carrier

Both can be successfully used in this system with some changes in the circuitry of the transceivers.

Subcarrier techniques

Here the data first modulates a subcarrier which in turn modulates the RF carrier. As the subcarrier can be within the audio spectrum this technique allows the transmission of data using voice equipment. The audio bandwidth of a typical mobile radio channel extends from 300 Hz to around 3000 Hz, although individual channels can vary quite significantly because of differences in equipment. The group delay characteristic of these channels, important for data communications, is fairly constant over this range.

Zegers and Dekker have done a comparison of subcarrier techniques suitable for transmitting data over voice equipment, [63]. They conclude that a second-order bipolar code gives the maximum bit rate within current channel bandwidths. However, a high SNR at the receiver is necessary to keep the channel BER down to an acceptable value. A first-order bipolar code, whilst halving the bit rate, gives improved performance at lower SNRs but is still worse than either fast FSK or VSB at the same bit rate. Fast FSK unfortunately occupies a large bandwidth and so may not be practicable at high bit rates. VSB offers a similar BER as fast FSK at the same bit rate and SNR, and has minimal bandwidth requirements. Problems may arise with synchronisation using VSB which could make it less useful than might be supposed.

For simplicity a form of FSK was chosen with which to test the system. The results that were obtained are shown in chapter 6. Binary '0' was represented by one complete cycle of a 1 kHz tone and binary '1' by one cycle of a 2 kHz tone. The data was synchronised with the positive-going zero crossings of these tones producing a phase continuous subcarrier. This was fed into a typical FM transmitter. At the receiver the FSK was detected by a digital demodulator requiring no adjustments and allowing full microprocessor control. As far as the system is concerned the choice of subcarrier modulation is not important due to its ability to adapt. The choice will, however, affect the overall data throughput, and so should be made only after comparing the possible performance of each technique over the channel in use.

One other point to consider is the modulation technique used on the RF carrier. If it is AM then fading will be much more of a problem than with FM or PM. SSB is not practicable because of the in-coherent demodulation used by voice equipment. For data transmission the re-inserted carrier must be exact in frequency and phase.

Direct modulation of the carrier

Direct modulation of the carrier wave can give superior performance over subcarrier techniques. The disadvantage is that any voice equipment intended to be used in the system must be modified, perhaps quite extensively. Also, several modulation techniques are not suited to direct modulation. Amplitude shift keying is not a good choice because of the rapid amplitude variations of the signal caused by fading. SSB, DSBSC and VSB are all likely to be similarly affected but to a somewhat lesser degree, [10]. More significantly they have carrier synchronisation problems under these conditions. The obvious choice is FSK with its relative immunity to amplitude variations.

Direct FSK

Most mobile radio transmitters generate their carrier by frequency multiplication of a signal from a quartz crystal oscillator. In FM and PM transmitters the signal direct from the oscillator is phase modulated by the audio before multiplication. Recently crystal oscillators have been developed to enable the frequency of the crystal to be changed slightly by a control voltage, [2]. These can produce direct FM with good linearity and excellent stability. The frequency synthesizers now appearing in mobile radio transceivers can also be used to produce direct FM, [52]. The ease with which direct FM can now be generated has prompted work into the characteristics of this mode, [57].

To produce a direct FSK signal the baseband data signal needs only to be shaped prior to being applied directly to the frequency control element of the oscillator. The shaping can even be done digitally by a microprocessor to allow the system even more adaptability, [8,54]. Voice modulation can also be applied in the same way as the data thereby simplifying the design of data/voice compatible transmitters.

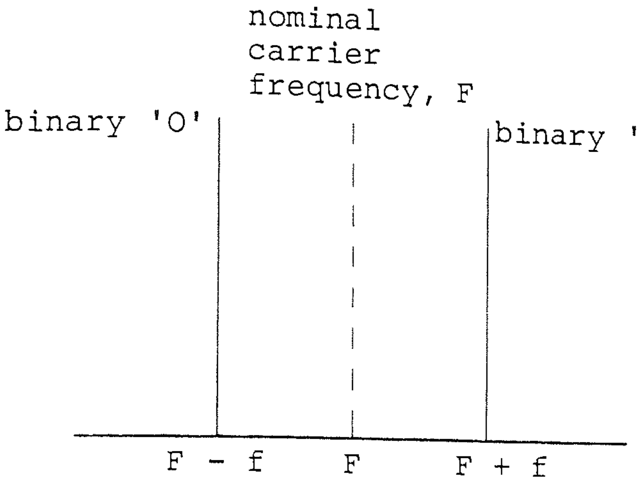
Demodulation

If subcarrier techniques are used with voice equipment the existing demodulator within the receiver will suffice for most applications. This is particularly true of FM equipment. Better demodulators could be fitted but the improvement in overall system performance is not likely to be worth the cost of modifications, unless they are done at the time of manufacture. Demodulation of a direct FSK carrier presents much more of a problem.

A direct FSK transmission within an existing mobile radio band will not be allowed to have a bandwidth greater than that of a voice transmission. This will limit the maximum bit rate and frequency shift of the carrier. Thus the frequency shift is likely to be only a few kHz. Refering to figure 4.3, the percentage change in carrier frequency between binary '1' and binary '0' is only a small fraction of 1%. For example, with a carrier frequency of 100 MHz and a frequency shift of ±5 kHz this corresponds to a shift of only ±0.005% or ±50 ppm, on the carrier. At 1000 MHz the same shift would be ±0.0005% or ±5 ppm. The maximum frequency error of a typical quartz crystal for Private mobile radio (PMR) use is around ±5 ppm over the temperature range -10°C to +55°C. This error is 10% of the shift at 100 MHz and equal to the shift at 1000 MHz. As this tolerance applies equally to the transmitter as the receiver, the possible offset between the two could be twice these figures.

Although greater shifts may be allowed in new allocations around 1000 MHz the examples demonstrate clearly the problem of frequency offsets with direct FSK. The tolerances that are acceptable for voice communcation are inadequate if full advantage is to be gained from direct FSK techniques. Some means has to be provided to keep the transmitter and the receiver co-channel.

When higher tolerances are required for narrow-band voice communications the usual method is to keep the quartz crystal at a constant temperature in a crystal oven. This method is not the most satisfactory in modern solid-state equipment due to the size and power requirements of the oven. In addition, it cannot overcome frequency offsets due to other factors. The problem is overcome in this system by allowing the transmit frequency to drift within normal limits and locking the receiver to it. This approach compensates for frequency offsets no matter where they originate.



shift = ±f Hz
for F = 100 MHz
and f = 5 kHz
then $\frac{f}{F} = 0.005\%$
or 50 ppm

Figure 4.3 Direct FSK

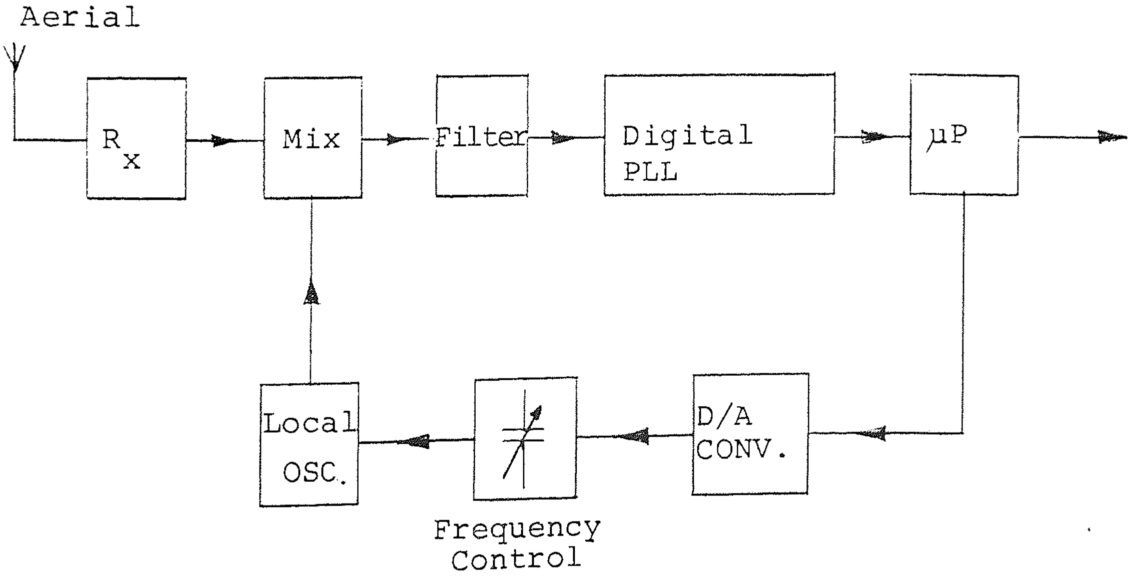


Figure 4.4 Frequency locking at the receiver

Receiver tracking

The way this is done is shown in figure 4.4. A standard PMR receiver is used up to the point immediately before the demodulator. At this point the IF signal is extracted and mixed with a signal from a voltage controlled local oscillator. The difference frequency is band-pass filtered and fed to a digital phase-locked-loop. The output of the PLL and its associated convertor is a binary number proportional to the frequency of the input signal. From this number the value of the binary data can be found. The numbers obtained from several consecutive bits are compared by the microprocessor with the numbers that would be obtained if the transmitter and receiver were exactly co-channel. Any frequency offset will show up as a unidirectional bias in the numbers, the magnitude and sign of which indicating how far and in what direction the receiver is off tune. Corrective action can then be taken by slightly adjusting the frequency of the local oscillator. This is basically a digital version of the automatic frequency control found in many FM receivers.

Digital PLL

One popular way to demodulate an FSK signal is to use a PLL, [25,33,34]. These work well but require analogue circuitry which requires initial setting up and does not easily lend itself to microprocessor control. A PLL has been described by Pasternack and Whalin which is totally digital in operation, [46]. Gupta also mentions this PLL in his review of current PLL types, [28]. A block diagram of a first order PLL of this type is shown in figure 4.5. The input signal is compared in the exclusive-or gate with the output of a binary counter. The counter can have either a high-frequency or low-frequency clock as its input depending on the output of the ex-or gate. This output is termed the gating function. It is a square-wave whose mark-to-space ratio is directly related to the input frequency and to the frequencies of the two clocks. Low-pass filtering of this gives the usual PLL analogue output. The lock and capture range of this PLL are the same and are solely dependant on the two clocks.

$$\text{lower lock/capture frequency} = \text{LF clock} / 2^n$$

$$\text{upper lock/capture frequency} = \text{HF clock} / 2^n$$

Where n is the order of the counter.

The dynamic performance of the PLL is determined by the n stage counter. The frequency of the clocks and the length of the counter can be under microprocessor control thereby allowing full control of the properties of the PLL by software. Reference [46] gives a complete description of this first-order PLL and of higher order PLLs of the same type. Just such a PLL in which n=10 was built. The high divide ratio of the counter, (1024), limited the highest lock frequency to 30 kHz. The high-frequency clock could not be taken above 30 MHz because of the TTL devices used. This is why the local oscillator and mixer was required in figure 4.4. The usual final IF of VHF/UHF PMR receivers is much too high for a PLL of this type. If a normal discriminator or PLL is used as the demodulator then the automatic frequency control can be incorporated around an existing local oscillator in the receiver.

Pulse to digital convertor

The gating function can be used to gate a binary counter as in figure 4.6. The output of the counter is made proportional to the mark duration of the gating function and is thus proportional to the input frequency of the PLL. This arrangement lends itself extremely well to soft-decision decoding of the data. Timing can be derived from the data either by the microprocessor or by conventional timing circuitry driven from the low-pass filtered gating function.

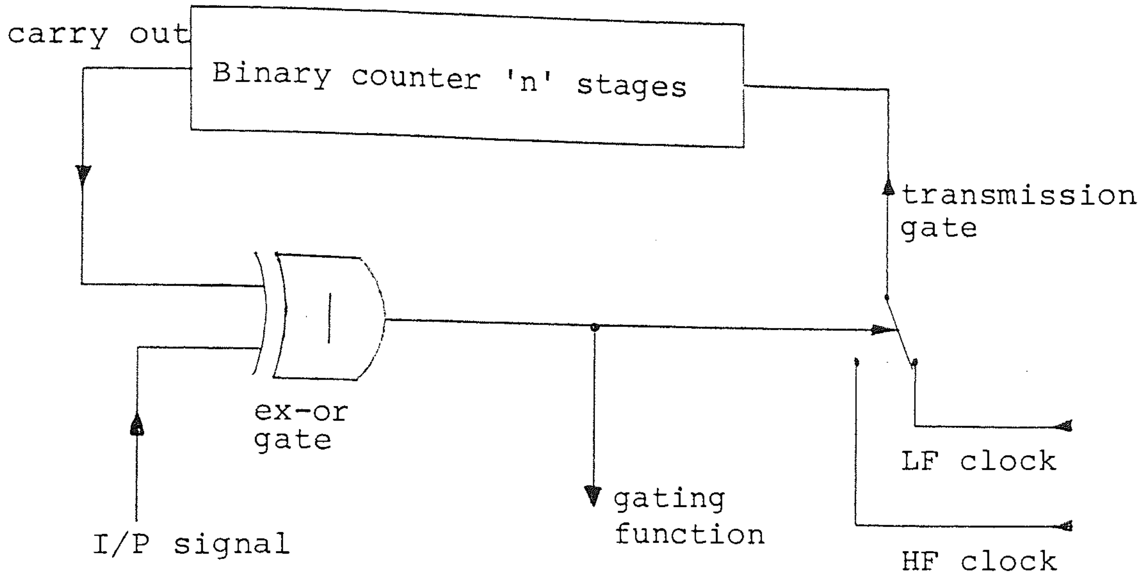


Figure 4.5 Pasternack & Whalin's 1st order PLL

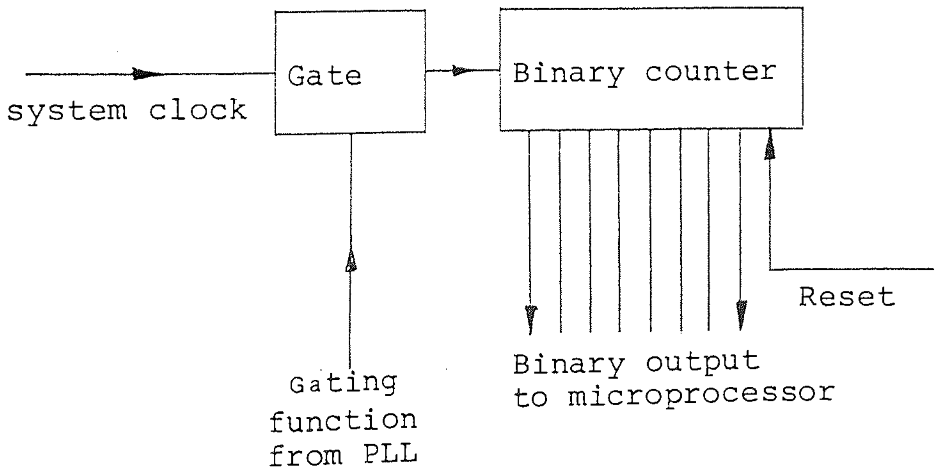


Figure 4.6 Period to digital convertor

4.5 CODING

In a system such as this coding is crucial to its performance. Whether it operates with FEC coding or error detection coding the overall system BER is, for all practical purposes, dependant on the chioce of code. As the system allows the use of many codes it is not intended to give any details of possible choices except those used in the trial system. Instead, a brief description of the coding requirements will be given for the two types of operation, FEC plus error detection and error detection alone.

Error detection codes

These codes only have to detect that an error has occured, consequently they add less redundancy to the data than an equivalent FEC code for a given system BER. For messages in plain text, and in similar circumstances where there is natural redundancy in the data, only a low level of protection is required. A CRC at the end of each minor data block is satisfactory for this purpose. Any particular part of the message that has very little or no inherent redundancy can simply be repeated within the message. For lower system BERs additional codes such as BCH codes can be employed to give whatever BER is required, (within limits). It is unlikely that a system BER better than 1 in 10^{10} will be required. In this system such a BER can be achieved with approximately 50% redundancy in the minor blocks, excluding the formatting and CRC. The system should support at least one code to give this BER at any time should re-programming of the system software be necessary.

Forward error correction codes

FEC with ARQ as a backup offers improved performance when the channel is not subject to severe fading or long bursts of interference, [39]. In addition to fading, the mobile channel is subject to impulsive interference, predominantly from the vehicle's own and nearby vehicles' ignition systems. These noise spikes can cause errors in a fade-free block. Such blocks would be rejected under normal ARQ operation but with simple FEC coding they may well be acceptable. Whether the use of FEC is more efficient in terms of data throughput than SRT-ARQ alone depends on the channel at the time the transmission is made and the code employed. For normal circumstances Hamming type codes operating on the minor blocks would suffice. For protection against short bursts of interference and fast, deep fading a convolution code, which spreads the checkbits further in time, operating on the major block would offer the highest probability of a recoverable block. It must be borne in mind however, that the additional redundancy required by these codes may, in the limit, make them less efficent than straightforward SRT-ARQ.

Soft decision decoding

The binary number passed to the microprocessor from the PLL can either be digitally 'sliced' to give a hard binary '0' or '1' result, or it may be treated as a analogue quantity that can be processed to give a maximum-likelihood type of decoding. This is soft decision decoding, the preceeding and following values from the PLL are taken into accout before a hard decision is made on the current value. This technique gives improved decoding when used either before the normal system decoding takes place or in conjunction with the coding within the data block, [12,26,27].

4.6 MICROPROCESSOR INTERFACING

The operation of the entire system centres on the microprocessor in the mobiles and the microprocessors or mini-computer at the base station. In order to function efficiently the information required by the microprocessor should be available at all times. The easiest way to achieve this with most microprocessors is to memory-map all the inputs (and outputs) onto the data bus. The information can then be accessed by any of the addressing modes that are available to the processor. Information that has a high priority can be used to trigger an interrupt, the processor can then respond to it with the minimum of delay.

The operator should have the option of over-riding the processor should the system fail. The transceiver should then behave as though it were a single-channel voice only set with limited manual control. Communication can then be maintained to some degree till the transceiver is repaired. The hardware of the system should also fail-safe should a fault occur when the operator is not present.

There are so many variables in a mobile system that the system cannot hope to operate at maximum efficiency all of the time. In an effort to make the system as adaptable as possible the software should be allowed to modify itself to a limited degree. The performance of the system can then be monitored to ascertain whether such a change was benificial or not. Eventually enough information should be gathered in this way to enable predictions to be made as to the outcome of changes in both hardware and software.

4.7 FURTHER DEVELOPMENT

Several ideas presented in this chapter have not been tried other than in the laboratory. These ideas will need practical tests to ascertain their real effectiveness and to arrive at optimum designs. The design of new commercial mobile radio equipment should be biased towards microprocessor control of its functions, with full use being made of the LSI circuits now available. However, extreme care will be needed to achieve the levels of reliability a system such as this requires.

The real further development lies not in the hardware but in the system software. There has been little, if any, software previously written specifically for this type of data transmission. Much more work in this area needs to be done before algorithms can be produced which are known to give near optimum system performance at all times over mobile radio channels. The more information that is accessible to the mobile transceiver the more efficient it can be expected to become. This information is not limited to that concerning the channel, it could include details about the state of the vehicle in which it is mounted.

With microprocessors appearing in vehicles' sub-systems it should not be difficult for a 'status-line' to be provided from the vehicle's electronics for use by any installed equipment. This could be extended into a packet switched 'ring' carrying digital information around the vehicle. Two-way communication between the vehicle's electronics and the mobile transceiver would be possible with this arrangement.

CHAPTER 5

SYSTEM DESCRIPTION

5.1 INTRODUCTION

In this chapter the single-channel, single-user system that was built is described. Figure 5.1 shows a simplified block diagram. The system hardware will be described in section 5.2 and the software in section 5.3. The operation of the system is covered in section 5.4.

The transceiver used as the base station is an all-valve Pye, 25 kHz channel spacing, FM, base unit operating in the 2 metre amateur radio band. This is representative of the older equipment still in use. The mobile transceiver is a current solid-state amateur FM unit. Its specification is similar to that of current commercial mobile radio equipment. In addition, it uses direct frequency modulation of a crystal oscillator and has a 40 channel synthesizer which can be programmed by external logic. No internal modifications to the transceivers were necessary for the subcarrier modulation technique employed. The transmit FSK signal from the hardware is fed to the microphone input socket, the output from the extension speaker socket is all that is needed to drive the FSK receive circuitry. The receive squelch line was initially bought out separately but it was found there was little advantage in doing this. The internal squelch circuitry of the transceiver was found to be quite adequate for subcarrier modulation. Switching between voice and data transmission was done manually, in a commercial system this would be done by the microprocessor.

5.2 HARDWARE

The system hardware was designed to produce and detect an audio FSK signal. The signal consists of two frequencies, one complete cycle of a 1 kHz tone representing binary '0' and one complete cycle of a 2 kHz tone representing binary '1'. Operation can be over a simplex, half-duplex or full-duplex channel.

All the microprocessor support circuits are either NMOS or schottky TTL devices. The purpose-built interface circuitry for both the transmitter and receiver sections use CMOS devices throughout. As a result, the power consumed by the system (around 7 watts, 80% of this at 5 volts) is almost entirely dissipated in the microprocessor and its support circuits. If CMOS devices were used wherever possible the power required could easily be reduced to less than one watt at 10 volts, including the receiver front-end.

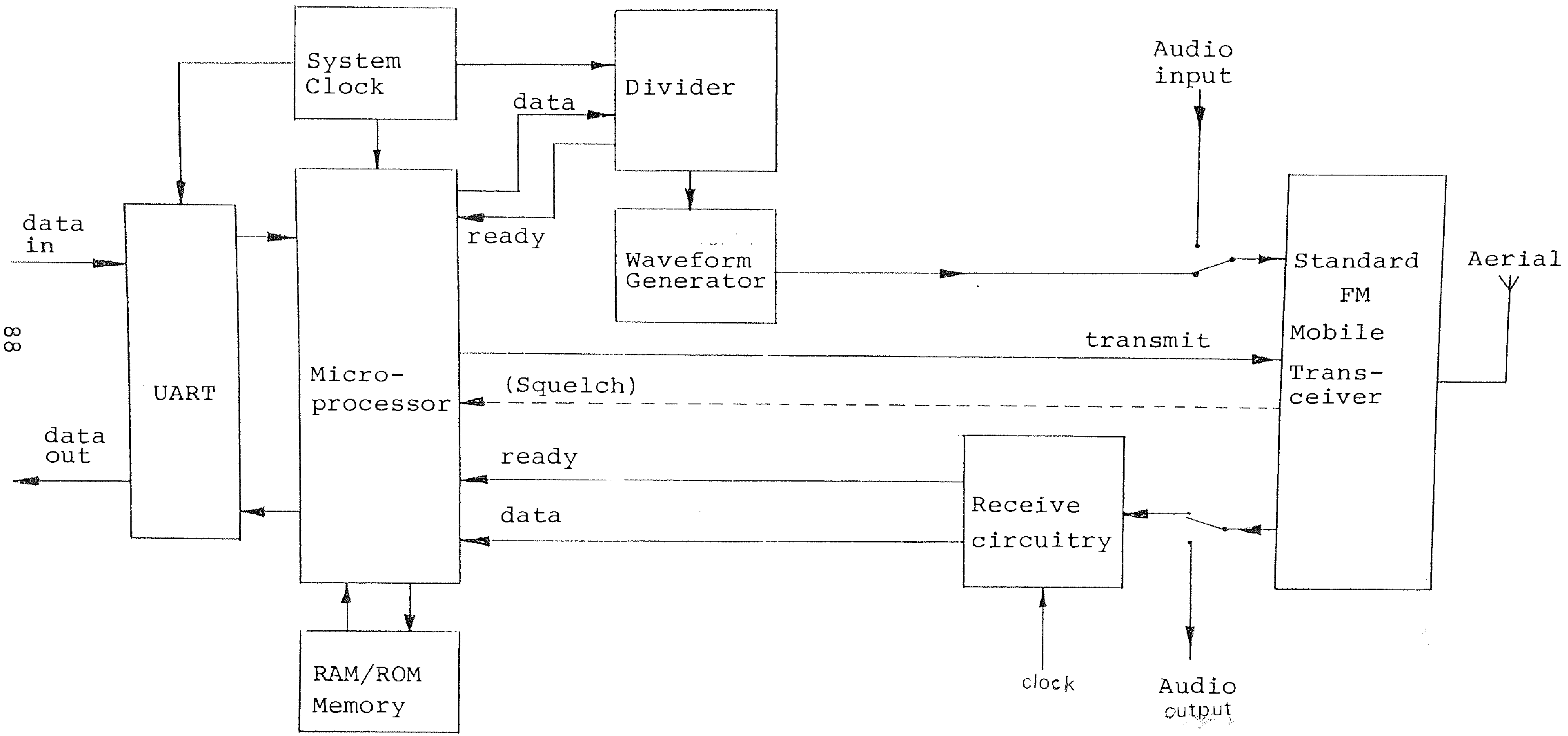


Figure 5.1 System Block Diagram

Microprocessor

The microprocessor used at the base station is a Texas TMS9900. This is a 16-bit device with excellent facilities for real-time control applications. These facilities help with both interfacing and software development but are in no way essential. The processor used in the mobile is a TMS9980. This is very similar to the TMS9900 and is software compatible. However, it is somewhat slower than the 9900, but requires less hardware support, making it more suitable for mobile use.

Both systems have their programs in EPROM, together with a set of initial system conditions which are read into RAM. As communication proceeds, these values are changed by the processor to reflect the current state of the channel. Only two error-detection codes are provided in EPROM, one using the CRC that protects all the blocks and the other using a BCH code[19] giving a system BER of not more than 1 in 10^{10}. Any other error-detecting code or FEC code that may be required by the system can be transmitted to the mobile and stored in RAM.

Transmit hardware

A block diagram is shown in figure 5.2. The transceiver is switched to transmit by a signal from the microprocessor, earthing the transceiver's press-to-talk line via a latch and transistor. The transmit FSK signal consists of complete cycles of either 1 kHz or 2 kHz tones. The data signal determines the frequency of each cycle which is changed only at positive zero-crossings. Each cycle is formed by sequentially scanning a weighted resistor network, this produces a stepped sinewave whoose frequency is determined by the rate at which the network is scanned. This scanning frequency is derived by division of the 1 MHz system clock, a factor of two in the divider is made controllable by an external logic signal. Thus the network can be scanned at either of two rates, one twice the other. The data from the microprocessor is used to control this extra division, binary '0' was chosen to give the lower rate.

From the above it will be clear that the transmitted bit rate is 1000 bits/sec for binary '0's and 2000 bits/sec for binary '1's. The overall bit rate will, therefore, depend on the actual data to be transmitted. Assuming random data, this will be around 1330 bits/sec. Should the channel be capable of supporting a greater bandwidth than the one used, then the bit rate can be increased accordingly. Extra bits added to the data for formatting and for the CRC reduce the average data bit rate to around 1200 bits/sec. The figures just mentioned are not to be confused with the data throughput which will depend on the channel. All the transmit timing is governed by the 1 MHz system clock which is derived from the microprocessor's clock of 3 MHz, see figure 5.3.

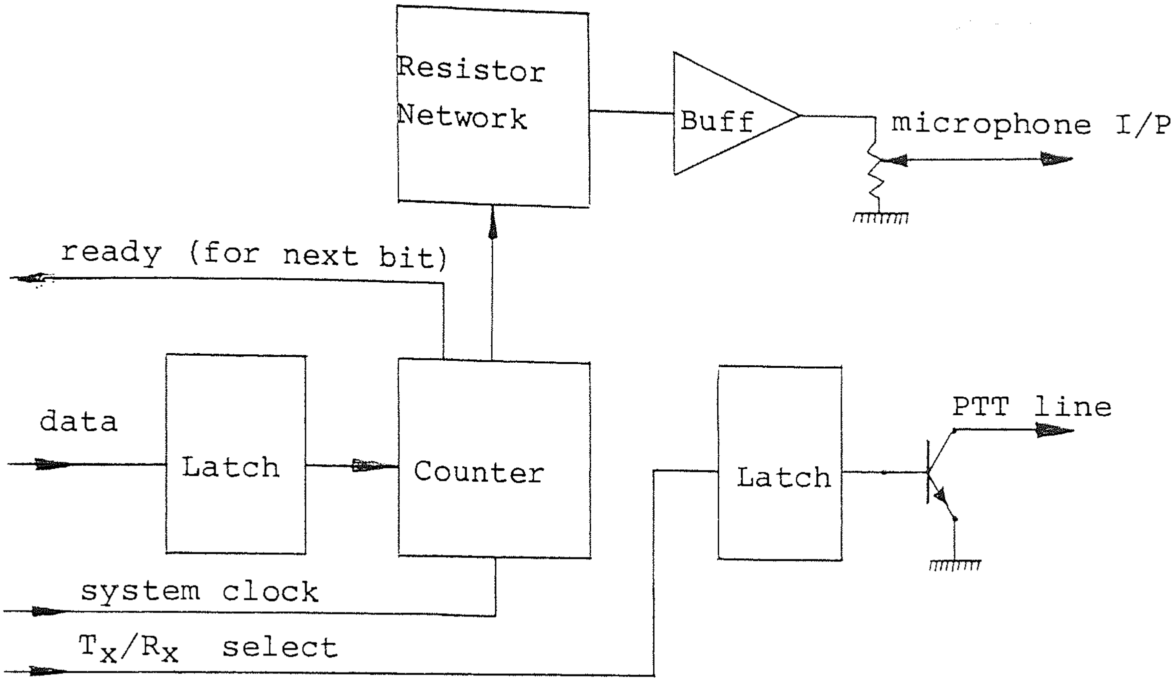
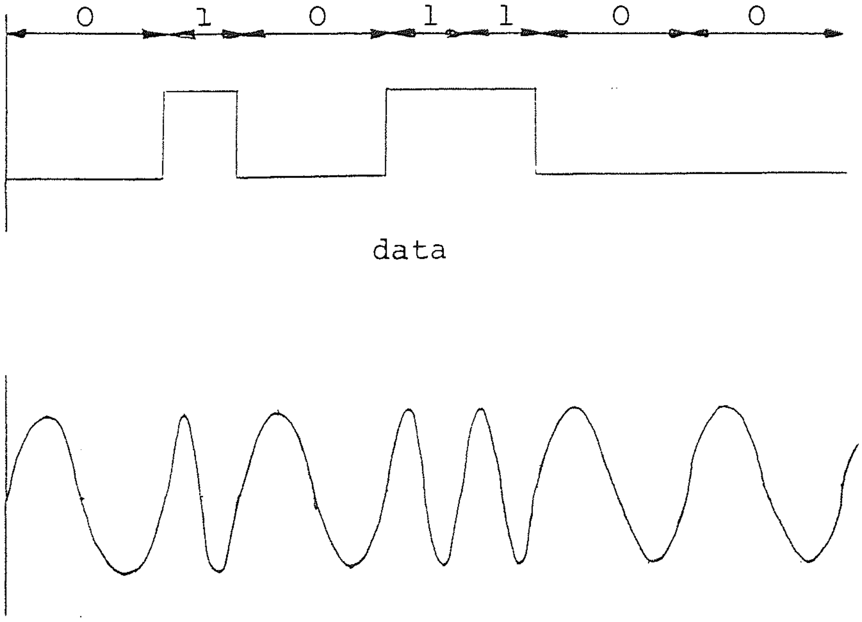


Figure 5.2 Transmit Section



FSK Output

Figure 5.3 Transmit Timing

Receive hardware

A block diagram and associated timing diagrams are shown in figures 5.4 and 5.5 respectively. The audio signal from the receiver is buffered, filtered (if necessary), limited and sliced about zero volts. The positive zero crossings of this signal trigger a monostable. This generates a short pulse which latches the previous output of the counter into a tri-state buffer and triggers a further monostable which re-sets the counter. The counter is then clocked by the 1 MHz system clock until the next positive zero crossing of the input signal. The value of the count at this point is directly proportional to the period of the last cycle of the input. The 'counter reset' monostable also informs the processor that there is a new count in the latch. The processor has until the next positive zero crossing to read the data from the latch.

The receiver's internal squelch line can, if desired, be brought out to interupt the processor should the signal be lost. However, in this system the absence of any audio output from the receiver when the squelch operates is detected by the software.

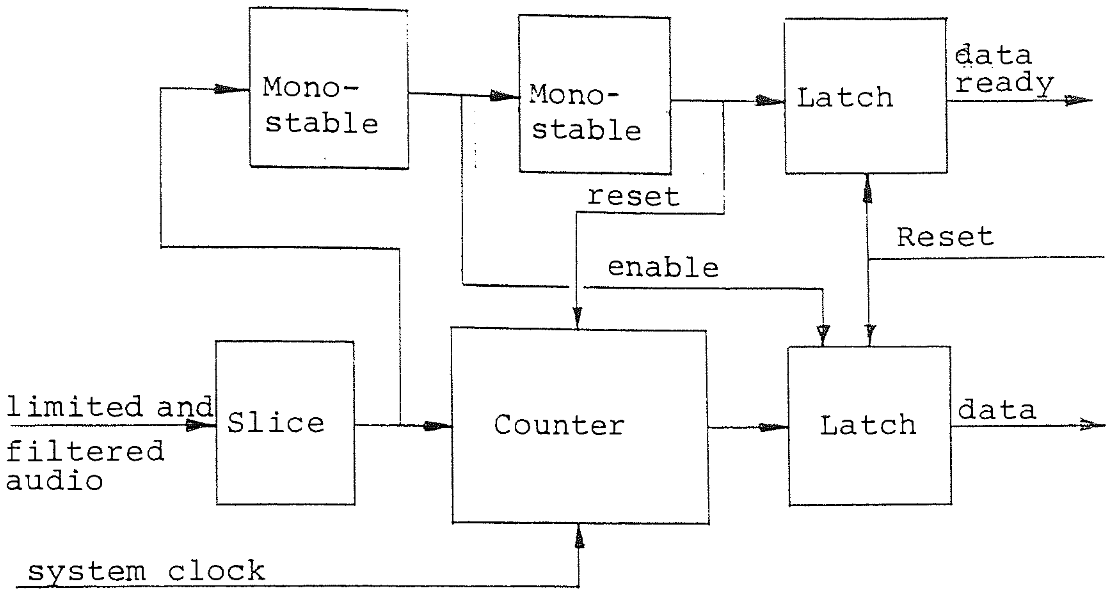


Figure 5.4 Receive section

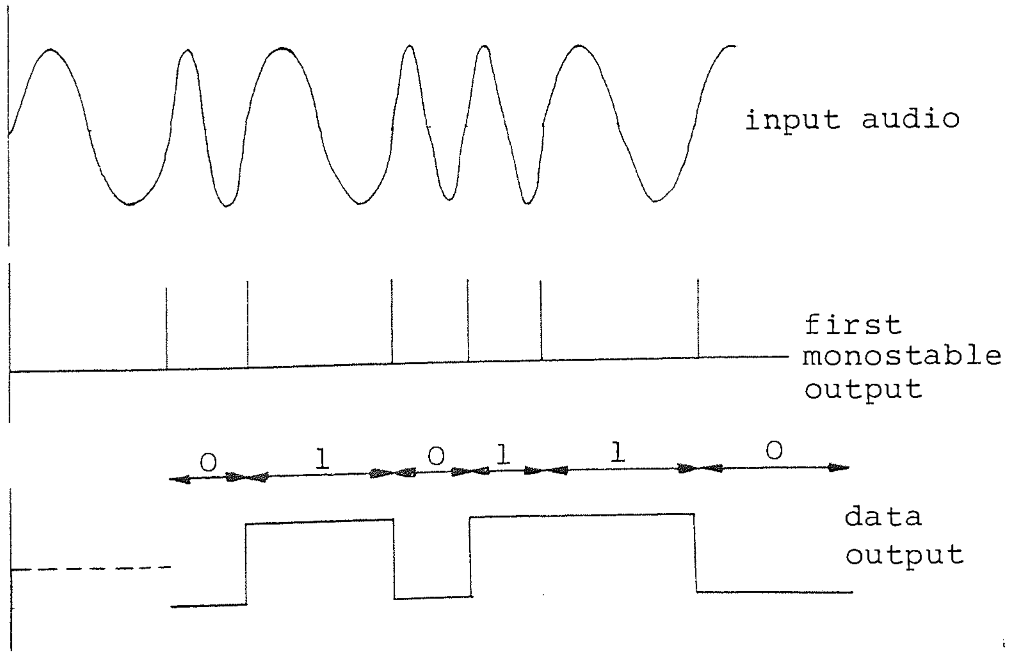


Figure 5.5 Receive timing

5.3 SOFTWARE

Source code listings can be found in appendix D. They show the basic I/O routines used to transmit and receive individual bits and words of data. Briefly, the software must take care of the following:

(1) The off-air signal from the receiver must be constantly monitored when data is not being transmitted. The four states to cater for are: (a) 'channel free' signal present, (b) 'channel in use' signal present, (c) data present, (d) signal lost.

(2) The current state of the channel should be recorded at regular intervals.

(3) When data is being received, this task should take priority over everything else.

(4) Received data should be checked for errors.

(5) If extra coding is incorporated in the blocks, the data will need to be decoded.

(6) Choosing the coding when transmitting data and acknowledgements.

(7) Encoding the data before transmission.

(8) Transmitting the data.

(9) The operator's controls should be checked periodically.

The next section on operation will expand on some of these tasks.

5.4 OPERATION

This is the most important aspect of the entire system. The operation of the system is very similar for both single and multi-user systems operating over a single channel. Multi-user, multi-channel arrangements, while basically the same as single channel systems, require a more complex high-level protocol. This will be covered briefly at the end of this section.

The description that follows relates to a single base station and one or more mobiles/portables operating over a simplex or half-duplex channel. Detailed diagrams of the various block formats referred to in this chapter can be found in appendix C.

Transmission from base station to mobile

The base station must first ascertain whether the mobile for which the data is intended is receiving an acceptable signal. An 'inquiry' block is, therefore, transmitted first. This consists of a preamble, the code corresponding to the mobile, the inquiry code and the length of the message to be passed. A CRC will terminate the block, as it does all blocks. The length of the message is most important as will be explained later, and should be transmitted as early in the communication as possible. The inquiry code distinguishes this transmission from a normal data or acknowledgement transmission which will have different codes at this point. If the mobile receives this inquiry correctly it will respond by simply sending two code words back to the base station. One of these will inform the base station of the current channel conditions. Providing the base station transmits continuously, the mobile will always have an up to date evaluation of the channel for this purpose. Should the base station not receive an acknowledgement, or it is corrupted, it will continue to send the inquiry block at regular intervals. If communication is not established within a certain time, the base station processor will inform the operator, who can then take appropriate action.

Having received an acknowledgement from the mobile the base station will then send the message. The size and coding of the minor blocks that will comprise the message is indicated in the major block header. If the message is quite long the base station will send it in more than one major block. The mobile must know the total length of the message as the information preceeding each minor block relates only to the position of the data within the message. The mobile will not send an 'all message received' signal until the entire message has been correctly received. This is the reason for sending the message length at the first opportunity.

After the first major block has been transmitted by the base station it can expect to receive a response from the mobile. In addition to giving the base station information about any minor blocks that were missed, the acknowledgement will also contain an update on the current channel conditions. Should the base station then decide to change the length of subsequent minor blocks, the mobile has to receive enough information to enable it to associate these new minor blocks with any that may have already been received. It is this variability in the size and coding of the minor blocks which requires extra information to be sent indicating the position of every minor block in the message. In this system each minor block is labled with its absolute position in the message. When the message is almost complete, the format of the acknowledgement block may change to indicate directly parts of the message that still have not been received. All the block positions sent by either the base station or the mobile require only 13 bits, even though the system can handle single messages up to 64 Kbytes in length. This is because the smallest minor block is 8 bytes long. It is, therefore, the smallest unit of the message that needs to be identified. Intermediate positions cannot legally occur and so need not be catered for. If the base station misses the acknowledgement it will transmit a 'repeat last message' signal, repeating it several times if conditions are poor. This same signal can be used by the mobile in appropriate circumstances.

When The channel becomes free, the base station transmits a 'channel clear' block continuously, informing any mobile that wishes to communicate that it can do so at the correct time. If a mobile is transmitting and the base station has no data to send, it will transmit a series of '1's. This informs other mobiles that their common transmit frequency is in use. When the base station is sending data, the state of the mobiles' transmit frequency is assumed to be the same as it was immediately prior to the start of transmission.

The beginning and end of every message is basically the same. The preamble always consists of at least 33 consecutive '0's followed by the sequence '101'. This sequence synchronises the receive circuitry after the run of '0's. It was chosen because the system usually changes a '0' into a pair of '1's when an error occurs. The distinction between an error and the synchronisation sequence is thus apparent to the system on the second bit. The receiver can, therefore, reset as quickly as possible after such an error. The four bits following the synchronisation sequence always indicate the number of the mobile (or portable) to which the transmission is directed, or from which the transmission is coming. Four bits allow for 15 mobiles, the 16th state is used as a broadcast indicator meaning that the following data is intended for all mobiles. The next four bits indicate what type of transmission it is, the same code sequences are used by both the base station and the mobiles. If all these four bits are '0's this signifies that a further four bit sequence is to follow. The interpretation of this new sequence can be entirely different from the usual sequence so allowing a further 15 codes. An example, which is used in this system, is when the mobile is being re-programmed by the base station. The all '0's code indicates immediately that the transmission consists of machine code instructions for the processor and not normal data. These instructions are further indentified by the additional four bit code just mentioned.

The system assumes nothing about the data to be transmitted other than it is in multiples of eight bits. Precautions have, therefore, to be taken to ensure that valid data sent to one mobile is not interpreted as a preamble and synchronisation sequence, or as a 'channel clear' indication, by another mobile that may have lost the signal for many bit periods. Any run of more than 32 '0's can be mistaken for a preamble, and a certain pattern of data could be mistaken for a 'channel clear' block. This latter mistake could cause two or more mobiles to transmit simultaneously. To prevent this happening, any consecutive pair of bytes that are both all '0's are always followed by a '1'. This is inserted at the transmitter and removed at the receiver. Should a long run of '1's occur it may prevent a mobile transmitting if it is mistaken for the 'channel in use' indicator. Clearly this is not particularly damaging to the system and will not cause interference. It will, in any case, be corrected following the base stations current transmission.

Transmission from mobile to base station

When the mobile is the originating station the operation of the system is similar to that previously described for base station to mobile transmission. The differences are as follows:

Unlike the base station when it wishes to transmit, the mobile has to wait for the base station to signal that the mobiles' common transmit frequency is not in use by another mobile. The mobile, therefore, waits for the 'channel clear' sequence to be transmitted. The priority code is compared to the priority of the message to be transmitted and, if it is less or equal, then the mobile transmits an inquiry block. The priority code transmitted by the base station varies from high to low priority thereby allowing mobiles with high priority messages access to the channel in preference to those with low priority messages. A random element can also be introduced if mobiles regularly have similar priority messages and so jam one-another.

The four bit inquiry code transmitted by the mobile is interpreted by the base station as a request to use the channel. Once the channel has been granted the other mobiles cannot use the channel except for acknowledgements. The operation from this point is much the same as for base to mobile transmission.

Forward error correction

As has already been said, errors usually occur by the receiver interpreting a '0' as a pair of '1's. This effectively inserts an extra bit into the block and so destroys the formatting. With only error-detection coding this is of little consequence as the block would be rejected due to the error alone. However, if FEC coding is in use this loss of formatting is disastrous. To overcome this problem when using FEC, binary '1' is represented by two cycles of a 2 kHz tone. Now, this type of error will still cause an error in the block but it will not destroy the formatting.

Multi-channel operation

Again, operation of a multi-channel system is similar to single channel operation once initial contact has been established. Before this, the mobiles and the base station(s) have several channels to choose from. There are two ways in which the choice can be made.

The first is to give each channel the same status and let both mobile and base station look for a free channel when one is required. Mobiles that are not actually transmitting or receiving data either monitor a common pre-designated channel, or else each monitors a specific channel out of those available. The base station will 'know' in advance on which channel a particular mobile will be found.

The second method is to allocate channels for specific purposes. A calling channel is still required but it is now dedicated to that purpose. Other channels can only be used for acknowledgements, the rest for data. The system will, naturally, be able to dynamically allocate channels to suit demand. In this case the transmit and receive frequencies of the mobiles are allowed to change channel independently, whereas in the first case the two frequencies would move together when the channel was changed.

Which of the two approaches would prove to be the most efficient would depend on many factors, including such things as average message length, message frequency, number of mobiles, average channel BER, etc.

CHAPTER 6

RESULTS

6.1 INTRODUCTION

The results obtained from the particular implementation of the system described in the previous chapter are presented here. Two sets of results were obtained, these represent the performance of the system when disturbed by white Gaussian noise (WGN), and rapid fading. The results will, therefore, be divided into two sections; 6.2 covering the system's performance in WGN, and 6.3 covering its performance when subjected to fast fading. A discussion of these results will follow in section 6.4.

It should be stressed that all the results presented in this chapter relate only to the system as described in chapter 5. Due to the inherent limitations imposed by that system, the results may be regarded as those obtained under 'worst case' conditions.

Experimental conditions

Unfortunately it was not possible to perform meaningful field trials on the system. All measurements were obtained by simulation of the channel, in terms of noise and fading impairments, in the laboratory. This did have one advantage in that the performance of the system could be assessed separately for noise and fading.

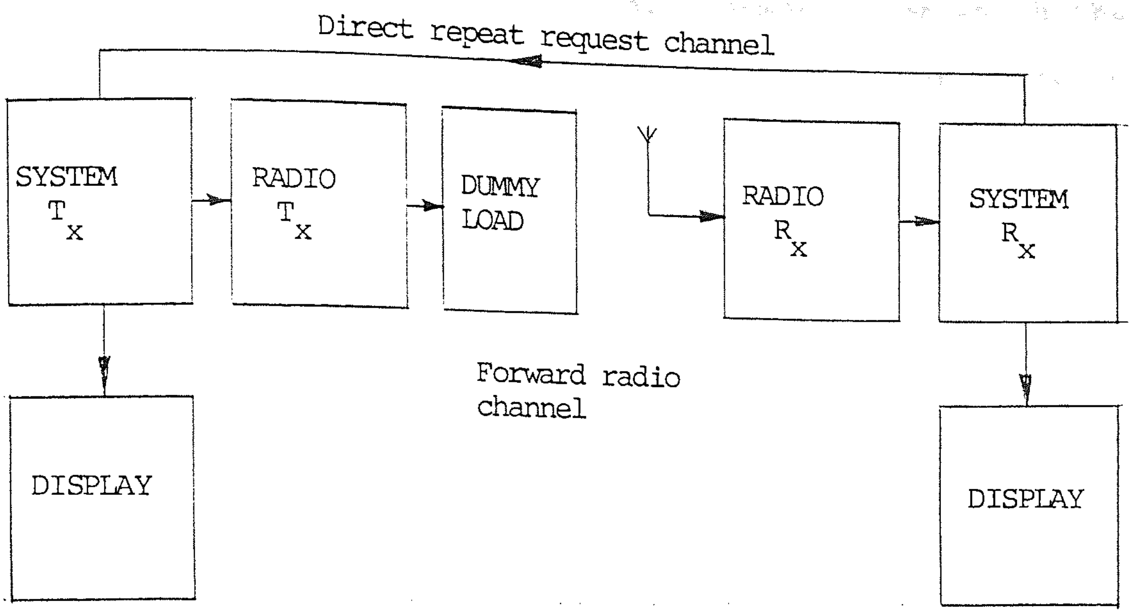


Figure 6.1 Experimental Conditions

Figure 6.1 shows the arrangement from which the results were obtained. The system was set up for forward transmission over the radio channel exactly as it would be in practice. The return channel was made by a direct connection between the receiver and the transmitter. This departure from the practical situation was felt to be acceptable as the only data passing over this channel would be request-repeat blocks. These normally have a high probability of getting through on their first or second attempt. The system itself generated and checked the data passing over the forward channel, so as to minimise the amount of additional equipment required whilst the results were being obtained. This data could either be a block of 1024 bytes of random binary data or 1024 bytes of text in ASCII format. It was found from early measurements that both block types were giving similar results. The majority of the measurements were therefore made using only the block of random data.

The radio transmitter on the forward channel was terminated in a dummy load, positioned close to this was a short whip aerial feeding the receiver. The RF field strength around the dummy load was sufficient to enable the receiver to produce a demodulated signal with a high signal-to-noise ratio, (40dB). White Gaussian noise was added to the demodulated signal, taken from after the receiver's first audio amplifier stage, by means of an adjustable noise generator. The noise generator was also used to produce the fading envelope for the fading measurements. Again, the fading was introduced into the signal path after the receiver's first audio stage. This arrangement obviated the need to simulate the complex frequency and phase response of the transmitter and receiver, whilst allowing full control over the addition of noise and fading to the signal. To generate the fading envelope the output of the noise generator was used to drive a field-effect transistor. This was biassed so that the signal would be attenuated as the output of the generator increased. This gave a characteristic not unlike 'square wave' fading but was slightly more realistic in that the envelope was reasonably smooth. The fading rate was controlled by adjusting the noise bandwidth of the generator.

What could not be checked dynamically was the ability of the system to continuously adapt to differing fading rates. However, the system did adjust as expected when the fading rate was changed manually; more will be said on this later. The Measurements obtained during each run were:

(1) Time - The length of time over which a particular set of results were taken.

(2) Blocks transmitted - The total number of blocks transmitted by the transmitter. This figure includes repeats.

(3) Blocks repeated - This is the number of blocks that were repeated by the transmitter. This is essentially a check on (8), in as much as the difference between

measurement (2) and measurement (3) should equal measurement (8).

(4) Blocks received - This is the number of block headers (preambles) that were correctly identified by the receiver.

(5) Loss of signal - This is the number of blocks that were lost through loss of signal, ie: fades.

(6) Format errors - This is the number of blocks that were rejected due to detected inconsistences in the formatting of the received block.

(7) CRC errors - This is the number of blocks that were rejected due to a disagreement between the received CRC and the CRC calculated from the received data.

(8) Good blocks - This is the number of blocks that were received without detectable errors.

(9) Actual errors - This is the number of blocks that were received with undetectable errors.

All the blocks referred to above are minor blocks transmitted by the system over the forward channel. The major block size throughout all the measurements was 1024 bytes, ie: 8192 bits. The system was not allowed to change the minor block size during each test run. The formatting of the minor blocks was slightly different from that which has been described in chapter 5. Both the two four-bit codes following the preamble were omitted, reducing the formatting and CRC overhead on each minor block from 76 bits to 68 bits. In addition, no major block header (66 bits) was transmitted. This effectively meant that a major block consisted of minor blocks simply sent in sequence. The repeat request acknowledgement, if present, from the receiver was ignored by the transmitter and a whole new major block was always sent after receiving it. If the request repeat was not received due to the receiver completely missing a header then the system waited an extra one second before 'timing out' and re-commencing transmissions. Normally the delay overhead between blocks was around 200 mS, this ensured that any loss of synchronisation in one block did not carry over into the next.

When a complete major block had been received, the receiver compared the minor blocks that had not been rejected with a copy of the transmitted data. Any discrepancies were noted as actual errors. During the course of the tests the minor block size was varied from 8192 bits down to 64 bits (plus formatting and CRC bits) for each different signal-to-noise ratio, and for each different fading rate, that could be produced by the test equipment.

6.2 NOISE PERFORMANCE

White Gaussian noise was added to the system as explained in the previous section. For each minor block size the rms amplitude of the added noise was varied giving signal-to-noise ratios of 9, 11, 13.5 and 17 dB, prior to being passed to the data demodulator. SNRs greater than 17 dB produced no errors in the received blocks, this being due to the discrepancy between the noise characteristic of the generator and the true Gaussian characteristic. SNRs lower than 9 dB produced so many errors the system throughput fell to only a few percent of its maximum value, and the undetectable error rate increased sharply. The bandwidth of the added noise was set to extend from zero to 5 kHz. This was felt to be a realistic figure as, although the FSK signal would be low-pass filtered before being passed to the data demodulator, the bandwidth of the noise produced by a conventional FM detector is wide compared to the demodulated audio. In addition, the high-frequency content of this noise increases as the RF input decreases.

As only the block error-rate of the channel is directly measured by the system, the bit error-rate has to be inferred. This was done by comparing the number of minor block headers that were not recognised by the receiver and comparing this figure with the total number of blocks transmitted. Assuming that only one error has caused the header to be missed, a close approximation to the channel BER can be obtained.

This method gave the following BERs:

SNR	BER
17 dB	$<10^{-6}$
13.5 dB	1.3×10^{-4}
11 dB	2.5×10^{-3}
9 dB	10^{-2}

The figure for 9 dB is optimistic, as it was clear that in many cases there were two or more errors in the header causing it to be lost. A more realistic figure would be between 2×10^{-2} and 4×10^{-2} .

Mathematical model

Mathematical modelling of the system is complex, except in its simplest form, due to its adaptability in the presence of fast fading. Considering white Gaussian noise and a non-adaptable implementation of the system, the model presented here can serve as a guide as to how the system should perform in practice.

Taking the major block size, K, as 8192 bits, this can be partitioned into n minor blocks, each of D data bits.

So $K = nD$ where $n = 1, 2, 4, \ldots\ldots 64, 128$

This means that the minor block size can vary between 8192 and 64 data bits, the formatting and CRC overhead on each being 76 bits. For every n minor blocks there will be a header of 66 bits, and a 0.5 second delay whilst the receiver sends the acknowledgement. This acknowledgement is assumed to be received without errors.

Taking the probability of one bit being in error to be Pe, then for a block of m bits,

probability of errors in the block $= 1 - (1 - Pe)^m$

Therefore, the average number of times the block will have to be transmitted to be received without errors

$$= 1 / (1 - Pe)^m$$

Taking $m = 76 + 8192 / n$ (the minor block size) the average number of minor blocks, Bav, that will have to transmitted to pass all the major block will be:

$$Bav = n / (1 - Pe)^m$$

For every n minor blocks there will be a header, so the number of headers, H, will be:

$$H = Bav / n$$

These headers will be subject to errors. Therefore, to ensure the required number will be passed without errors the average number that will have to be transmitted, Hav, will be:

$$Hav = H / (1 - Pe)^{66}$$

The loss of Hav - H of these headers may cause otherwise error-free minor blocks to be repeated. These extra minor blocks, Bex, are assumed to be received correctly.

$$Bex = n(Hav - H)$$

Assuming random data the transmitted bit rate will be 1333 bits/sec, and there will be an overhead of 0.5 second per header. Therefore, the total time, T, for the transmission time of one major block is:

$$T = 0.5Hav + [mBav + 66Hav + mBex] / 1333$$

The optimum minor block size will be the block size which gives the smallest T for a particular Pe. Graph 6.1 shows the optimum block size (8192 / n) against BER. Graph 6.2 shows how the throughput efficiency falls as the BER of the channel increases, assuming the optimum block size is used. 100% corresponds to the throughput which would be obtained if the channel BER = 0.

As it stands this model is not quite applicable to the test runs that were made. Making the required modifications:

$$T = 0.2Bav + 1.0(Bav - n) + mBav / 1333$$

$$\text{where } m = 68 + 8192 / n$$

Graph 6.3 shows the optimum minor block sizes predicted by this modified model. Optimum block sizes obtained from the experimental results are shown for comparison.

Experimental results - Noise

Graph 6.4 shows the transmitted data rate in bits-per-second plotted against data block size for signal-to-noise ratios of 9, 11, 13.5 and 17 dB. All the curves tend towards a transmitted bit rate of 1333 bits/second as the block size increases. However, they would not actually reach this value as the formatting and CRC bits will always result in the transmitted data rate being less than the transmitted bit rate.

The graph shows two important characteristics of the system. These are:

(1) The data rate decreases smoothly as the block size decreases for all SNRs. This is due to the progressively higher percentage overhead of the formatting and CRC bits, and is entirely to be expected.

(2) The data rate falls as the SNR decreases. This is due to lost headers causing system 'time-outs', when the transmitter waits for a pre-set time before re-commencing transmissions.

This last characteristic is again to be expected, however, the large change in data rate between the SNRs of 11 and 13.5 dB is worthy of comment. Graph 6.5, which shows data throughput plotted against data block size for the same four SNRs, makes this even more apparent. There is a fall in throughput from almost 850 bits/second to 150 bits/second between these two SNRs. To explain this large change in throughput over what is less than a 3 dB change in SNR it is necessary to relate the SNR to the channel BER. This has already been done in this section, a SNR of 11 dB corresponds to a BER of 2.5×10^{-3}, and 13.5 dB corresponds to a BER of 1.3×10^{-4}. Using the modified system model, with values of Pe between the two figures above, it becomes clear that over this same range of BERs the time taken to transmit a major block is changing most rapidly as the BER changes. Obviously, the values of Pe in this range are critical in their effect on the probability that a block will be received error-free. This is demonstrated by the effect on the block headers alone, (graph 6.4). Graph 6.2, although relating to the normal system model, also shows this.

The net result is that the system exibits a threshold around a SNR of 12 dB. Edwards found a similar threshold appearing in the system he investigated, [19]. However, the equipment and modulation technique he employed produced the threshold at a more respectable 6 dB.

6.3 FADING PERFORMANCE

It is not intended to derive an expression for the theoretical performance of the system. This would be difficult, primarily due to the way in which the fading envelope was derived, and of doubtful use. However, as the arrangement used produces an envelope similar to squarewave fading, the general result that the data rate should start to fall significantly if the fading rate exceeds one-seventh of the transmitted block rate can be used, [19]. The results will show that the data rate begins to fall around one-fifth of the transmitted block rate, but the precise point is not distinct. The apparent better performance of the system over theory is not significant in view of the approximation of the simulation.

Experimental results - Fading

Graph 6.6 shows data throughput plotted against data block size for different fading rates. As in the previous graphs, formatting and CRC bits are not included. A smooth fall in data throughput is evident as the fading rate is increased. There is no threshold as in the case of WGN. Graph 6.7 shows the information in graph 6.6 as data throughput plotted against fading rate for different block sizes. Examining this graph, several points are worthy of comment.

(1) Data throughput when using data blocks of 64 bits (total minor block length of 64 + 68 bits) is almost constant up to a fading rate of 50 Hz. Only when the fading rate is increased to 150 Hz and beyond do blocks of this length become seriously affected. It was this result that led to the choice of block lengths of around 60 bits for major block headers, inquiry, 'quick' repeat and 'channel clear' blocks. All these blocks have a high probability of successful reception with fading rates up to 100 Hz. As an example, this is close to the maximum fading rate that would be experienced at 1000 MHz and at a vehicle speed of 30 mph. An increase in the transmitted bit rate would raise the system's tolerance to fast fading proportionately, so allowing higher RF carrier frequencies and higher vehicle speeds.

(2) Data blocks of 128, 256 and 512 bits offer increased data throughput at lower fading rates, while still retaining a throughput comparable to 64-bit data blocks even at the higher fading rates.

(3) Data blocks of 1024 bits and above begin to be seriously affected by fading rates above 5 Hz. However, even slow fades (0.15 Hz) can be detrimental to data throughput, this being due to the duration of the fade increasing as the fading rate reduces, making the channel unuseable for quite long periods.

(4) If there is no fading at all then the larger block sizes, 4096 and 8192 bits, give much improved throughput. When this happens the system is, of course, limited by noise alone.

Operation of the experimental system in the field would be slightly different to its operation in the laboratory where the results were obtained. However, the only difference worthy of mention is the decrease in SNR that would accompany a 'real' fade. Due to the characteristics of conventional FM demodulators the transmitted sub-carrier would not simply vanish during a severe fade, and so be correctly interpreted as a loss of signal, but would be replaced by noise. The data demodulator looks for zero-crossings which would still be present, but would not contain any useful information. It is advisable, therefore, to prevent this situation occuring. This can be achieved by adjusting the receiver's squelch circuit to operate at a pre-determined SNR (such as 12 dB in this case) thus cutting off the output completely. The data demodulator now detects a loss of signal instead of random noise. If this is not done the probability of undetectable errors in the blocks, particularly small blocks, increases, making the system less reliable for little increase in throughput.

The system as described produced no undetectable errors when using data block sizes greater than 64 bits, ie: 64 + 68 bits in total. The undetectable block error-rate with 64-bit blocks did not exceed 10^{-4}. This was not unexpected due to the simple modulo-2^{16} addition CRC which was used throughout the test runs. The apparent departure from theory for the larger block sizes can easily be explained, and is one of the hidden advantages of the system.

It was noticed that when errors occured, binary '0's were often mistaken for a pair of binary '1's. The sub-carrier modulation scheme employed is prone to this. Indeed, no data block greater than 64 bits that was investigated for individual errors ever showed a '1' mistaken as a '0' unless several '0's had been mistaken for pairs of '1's. Extra bits inserted into blocks in this way completely destroy the formatting of the block. As such these blocks are rejected because of inconsistences in their formating before the CRC is checked. With very short blocks, particularly at high fading rates, it is possible to have undetectable errors without the formatting being destroyed.

6.4 DISCUSSION

It should be made clear, at this point, that the results presented in this chapter were not obtained solely to confirm that the operation of the system was as expected. The prime use of these results was to influence the design of the system once the basic 'hardware' decisions of frequency, modulation technique, bit-rate, etc, had been made. Although these results have been included at this point in the thesis for convenience, they pre-date much of the work described chapters 4 and 5. This is the major reason why the experimental system is rather basic in operation, and why there are slight differences in the block formatting. The results also form the operational foundation on which the adaptive system can build.

Impulsive interference

Little effort was made to determine the performance of the system when subjected to impulsive interference. The difficulty of producing controlled impulsive interference in the laboratory and the questionable validity of any results that may have been obtained, were the two primary reasons for this omission. However, it is not difficult to estimate the probable effects of impulsive interference on the system.

Without FEC, the system is not tolerant of any interference which is strong enough to produce errors in the received blocks. Modulation techniques which are resilient to any amplitude changes in the received signal (such as FM and PM) will offer best protection under these circumstances. The use of a FEC code would improve matters considerably. Only a simple code is required to give moderate protection as errors caused by impulsive interference can be regarded as single isolated errors at bit-rates less than about 5000 bits/second, [22].

Occasionally, interference can take the form of bursts of noise which can cause several consecutive bits to be in error. When this happens it is better to let the system operate in its normal SRT-ARQ mode rather than use FEC. In cases where the bursts are frequent and their characteristics are known, FEC could be used. However, it is unlikely that this will occur in practice.

Adaptive performance

Here, as in the case of impulsive interference, meaningful results are difficult to obtain in the laboratory. The system was run on several occasions during which times the fading rate and SNR were changed manually. In the main the system did adapt in the way it was expected to. The following points arose from these tests.

There is a trade-off between the speed of response and the 'stability' of the system. The system makes the decision on whether to change the minor block size by comparing the number of blocks that are received with no detectable errors to the number of block headers that have been recognised. A low ratio indicates blocks are being lost, so a decrease in block size is needed. An increase in the block size would be permissable if the ratio was equal or close to unity. In the case of FEC, the number of errors in recoverable blocks can also influence the decision. The number of blocks on which the decision is based is important. If the number is small then the response time is short but the system continually changes block size even though the channel parameters are constant. If many blocks are examined then the response time becomes excessive.

It became clear that a more complex method of deciding on the minor block size was necessary. The software was modified to provide both long and short term channel measurements. In addition, a record of past changes was kept by the system to determine any trends in the channel parameters. A real-time clock is useful to further increase the significance of this last piece of information. The distinction between those blocks which are rejected due to loss of signal and those rejected due to format or CRC errors can also be useful. This result can determine whether the channel is predominantly impaired by noise or by fading. The runs made using the modified software did show a slight improvement in the system's ability to adapt.

Receiver tracking

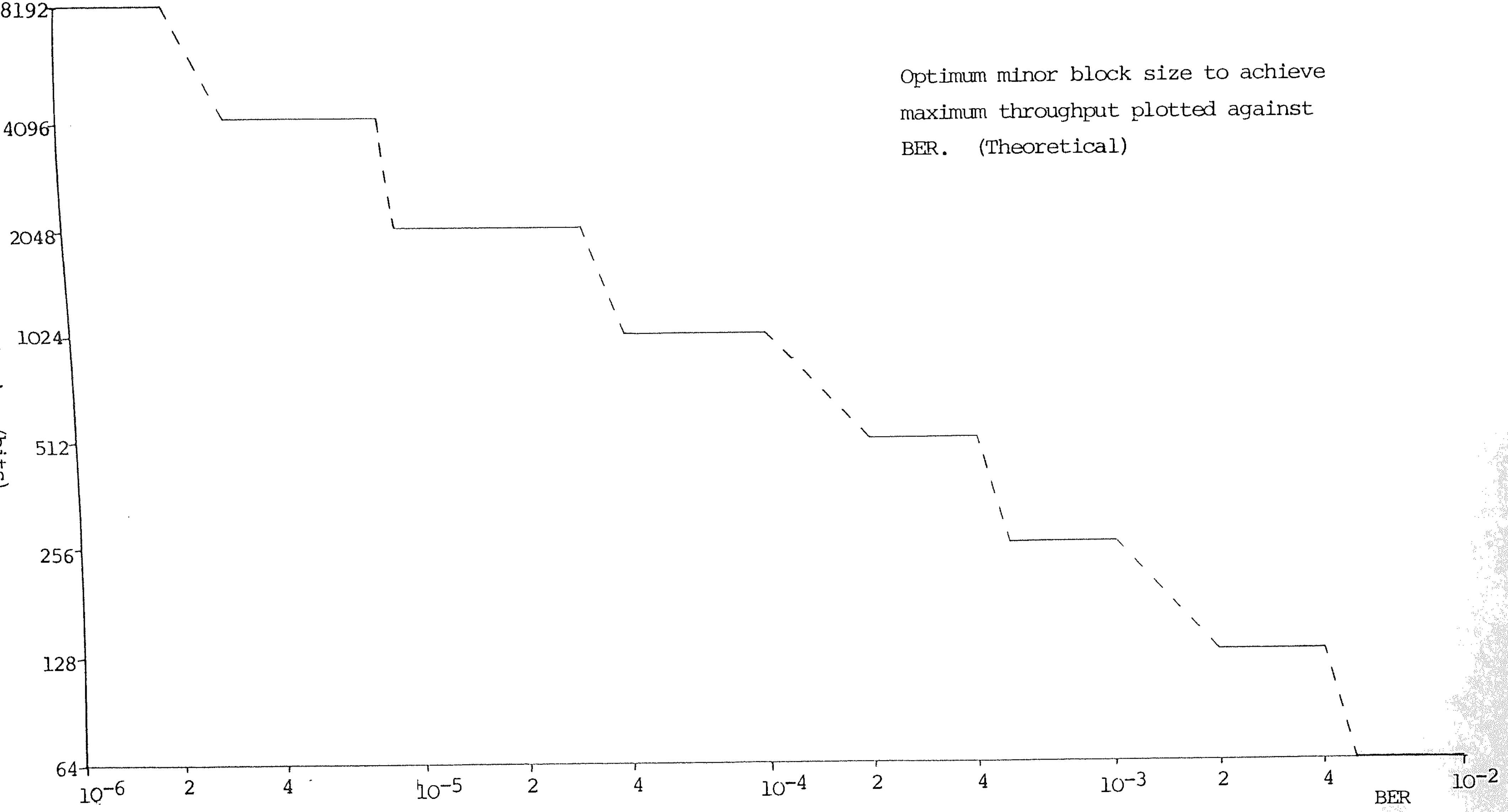
Although this is not directly applicable to the experimental system as it stands, a change from subcarrier modulation to direct modulation of the carrier would necessitate some form of frequency control to lock the receiver onto the transmitter's frequency. A simulation of the arrangement shown in figure 4.4 was set up and tried. In the absence of noise the 'receiver' remained co-channel with the incoming frequency. This was true providing the rate of change of the input frequency was slow compared to the bit-rate. The addition of noise produced increasingly erratic changes in the VCO frequency until the system failed at a SNR of some 6dB. This figue is worse than can be expected in practice as the data demodulator was still basically that of the subcarrier system. Providing corrective action is only taken based on data obtained from blocks in which there are no detectable errors the receiver will not lose lock. In the event of the receiver being too far off frequency, such as at system start-up, the VCO can be made to progressively scan over its range until the microprocessor can recognise incomming data.

Conclusion

The results show that a SRT-ARQ system designed to choose its block lengths automatically will give superior performance, in terms of data throughput, over one whoose block length is fixed. The basic experimental system produced an undetectable block error-rate of better than

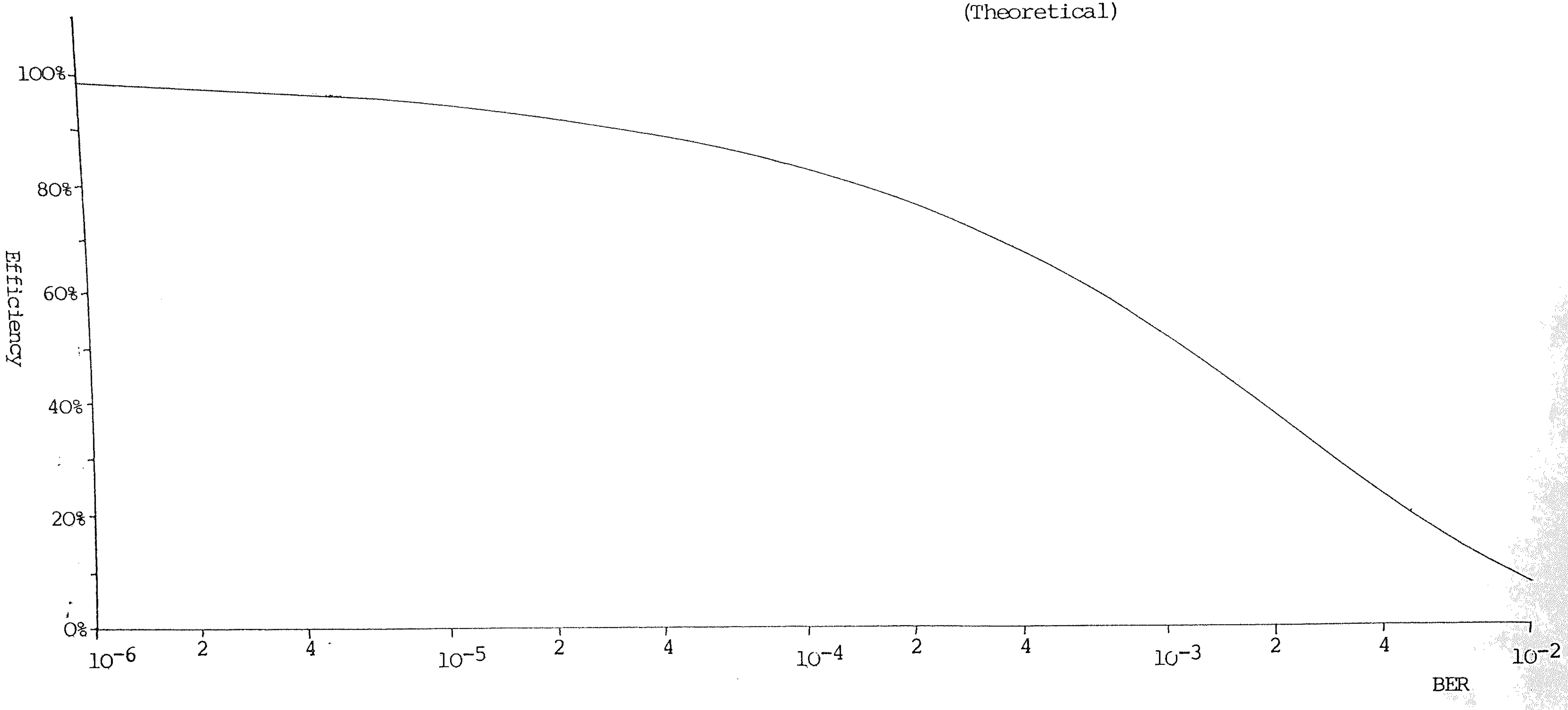
10^{-4} with small blocks. With larger blocks the error-rate was several orders of magnitude better. This is due to the additional security provided by the formatting and loss of signal checks. Unfortunately, this suprising performance cannot be expected to be quite as good when using FEC blocks or when using systems where bit errors are unlikely to destroy block formats. The basic system provides a low enough error-rate to enable even lower error-rates to be achieved with the addition of extra redundancy in the data block itself.

Examining graph 6.7 once more, it would seem that all the minor block sizes that are available are not necessary, particularly the larger sizes. However, when the mobile is stationary these larger blocks do provide the best throughput, and it is most likely that this is the time when most information will need to be passed. Comparing graphs 6.1 and 6.3 also shows why a complete range of block sizes is desirable. The only difference between these two graphs is the protocol of the system, there are no hardware changes. The optimum block size is a function of transmitted bit rate, signal-to-noise ratio, fading rate, and the protocol of the system. Any change in one of these is likely to produce a significant change in the optimum block size.

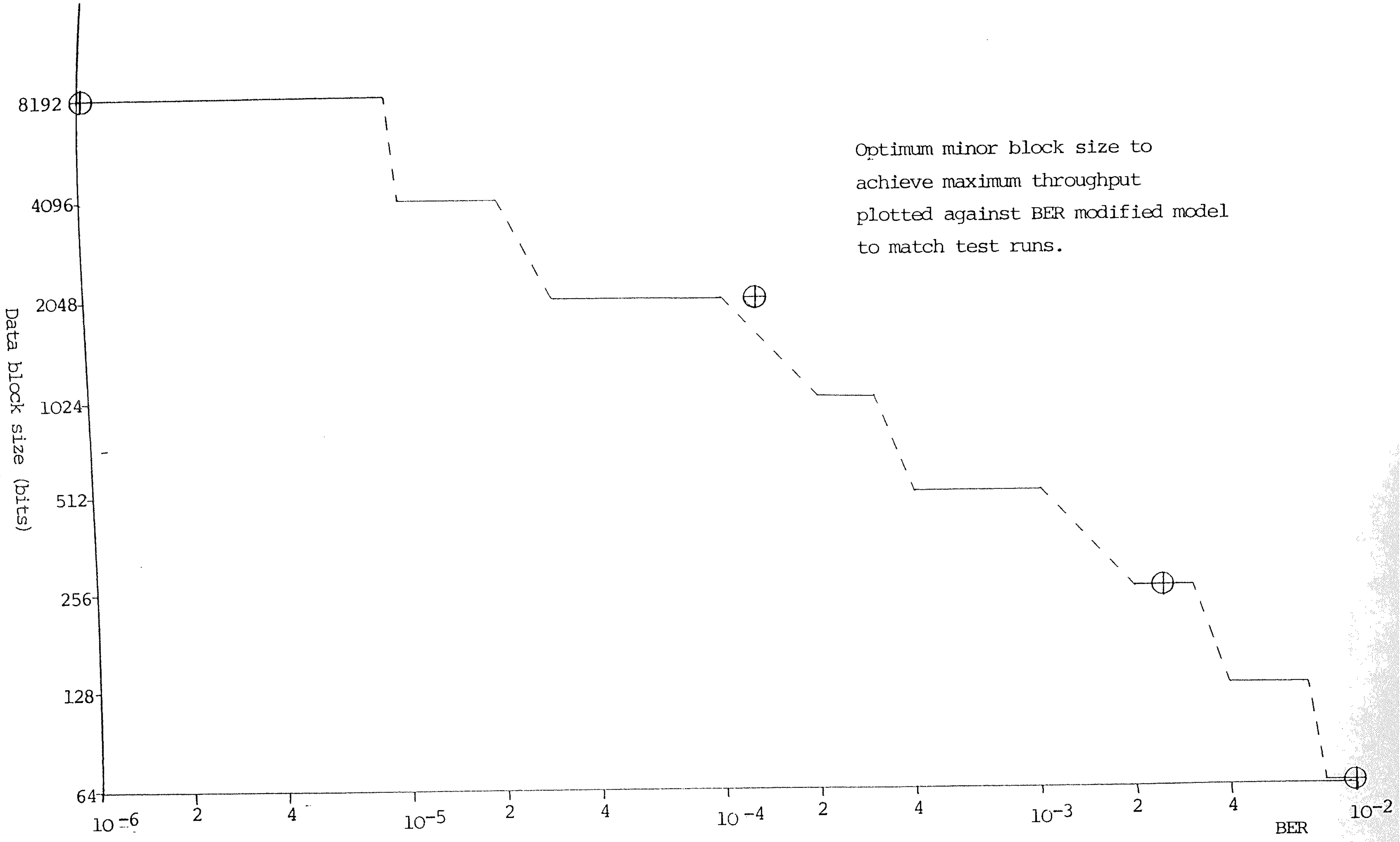

Optimum minor block size to achieve maximum throughput plotted against BER. (Theoretical)

Graph 6.1

Throughput efficiency
plotted against BER
100% corresponding to zero BER.
(Theoretical)

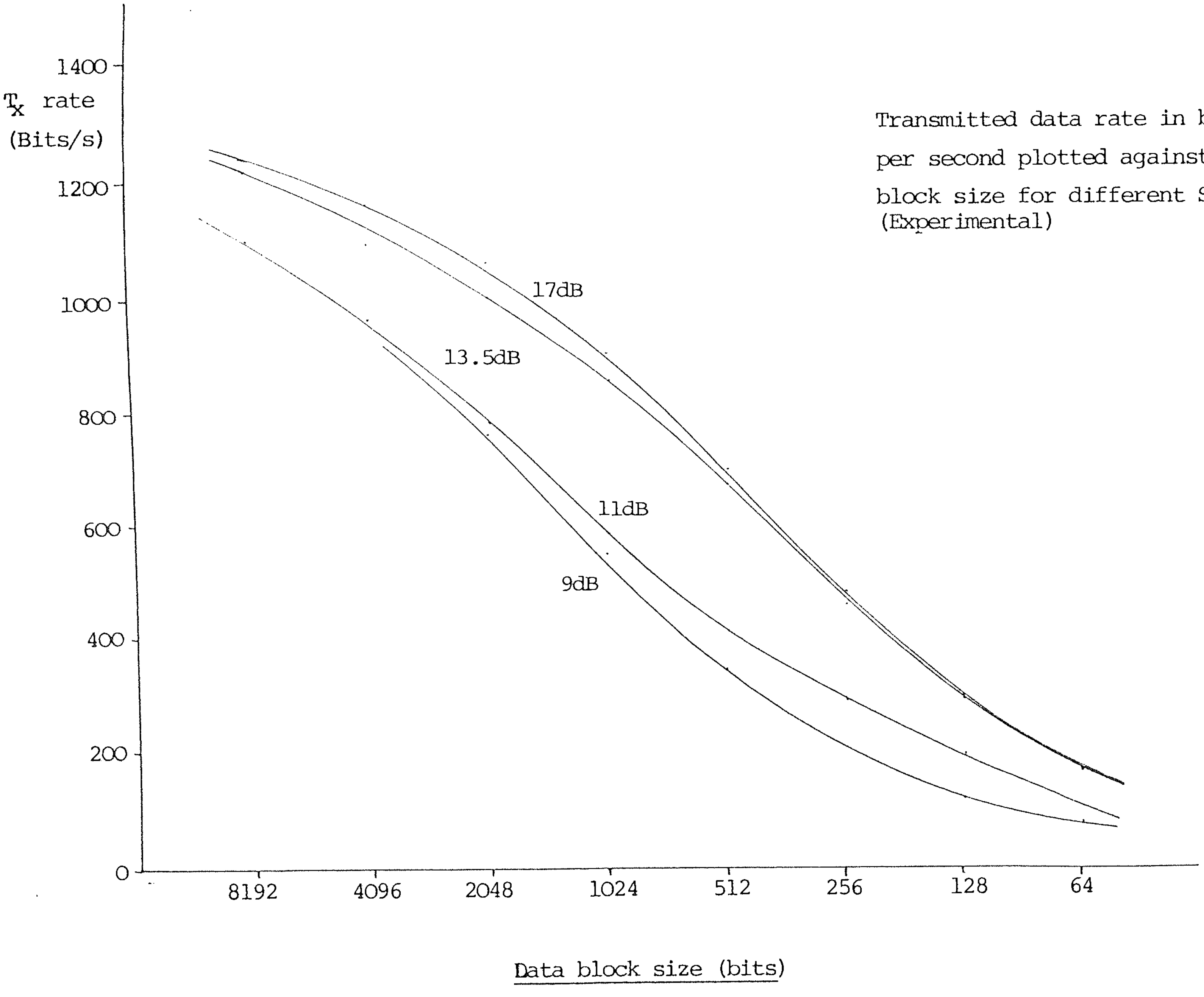


Graph 6.2



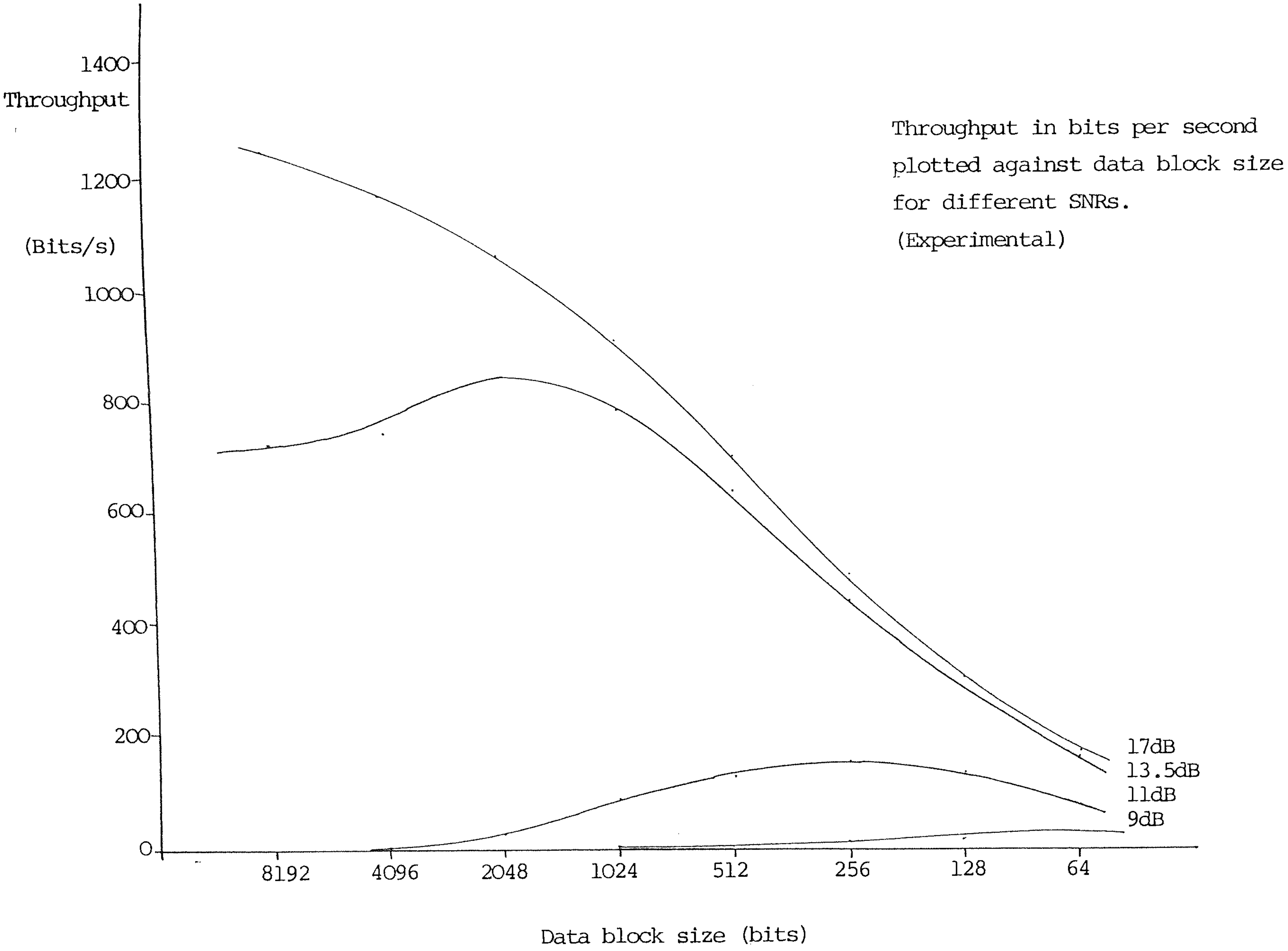
Optimum minor block size to achieve maximum throughput plotted against BER modified model to match test runs.

Graph 6.3



Transmitted data rate in bits per second plotted against data block size for different SNRs. (Experimental)

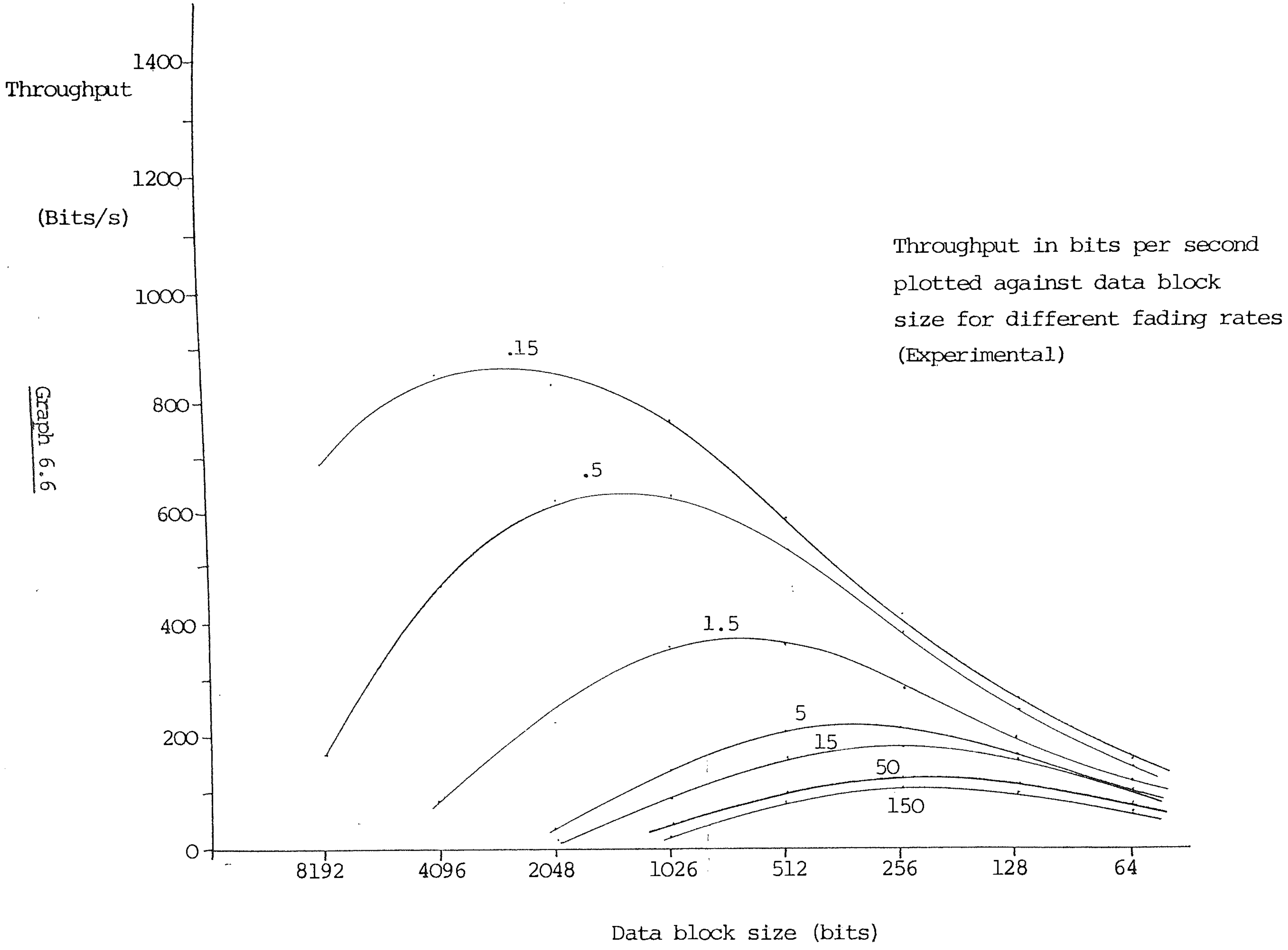
Graph 6.4



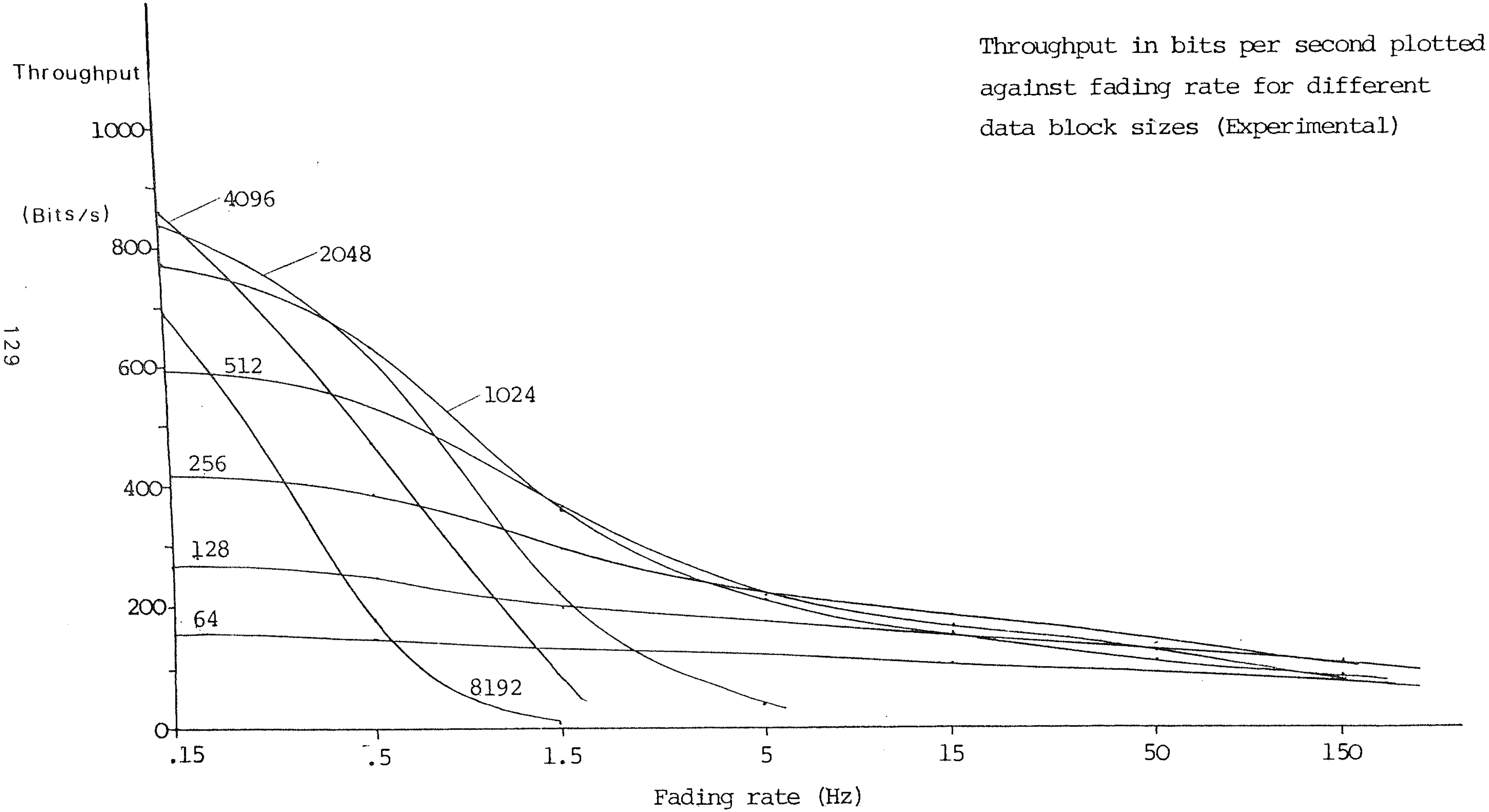
Throughput in bits per second
plotted against data block size
for different SNRs.
(Experimental)

Graph 6.5

Graph 6.6



Throughput in bits per second plotted against data block size for different fading rates (Experimental)



Throughput in bits per second plotted against fading rate for different data block sizes (Experimental)

Graph 6.7

CHAPTER 7

CONCLUSIONS

7.1 CONCLUSION

The aim of this work was to provide a highly reliable data communications system for use over channels impaired by multipath interference. The channel which suffers most from this problem is the VHF/UHF land mobile channel. For this reason, and because data communication to mobile vehicles is increasing rapidly, the work predominantly relates to this channel. However, the system has been designed so that it can operate sucessfully over any duplex channel. It can also be configured for single or multi-user and for single or multi-channel operation.

An experimental version of a simple implementation of the system was built and tested early in the design period. The results obtained were used in the final system design. It was not possible to test the system in its final form as a single unit. However, all the novel parts of the system were tried separately in the laboratory prior to their inclusion in the final design.

Throughout the research, considerable emphasis was placed on producing a practical system capable of being installed by current mobile radio users at low cost. To this end subcarrier modulation has been suggested, this allows the use of existing mobile radio hardware with little or no modification. The more efficient direct modulation of the carrier wave can be adopted immediately by new users who have been allocated new channels. Users initially employing subcarrier modulation can change to direct modulation with no change in the operation of the system, at least as far as the operators are concerned. Making the system as flexible as possible by the use of software rather than hardware enables radio sets to be mass produced to a common standard. Customisation is then achieved through the software.

System capabilities

The system possesses several features that are not available together on any current commercial system. A summary of the important features follows.

(1) Selectable error-rates - The system can operate with a low enough error-rate to provide safe transmission of computer data. Its normal error-rate (without additional coding) is acceptable for text and similar types of information.

(2) Single or multi-user and single or multi-channel - As the system is entirely under software control it can easily be changed to accommodate extra users or frequencies. The system remains transparent to both the data and the operator irrespective of its configuration.

(3) Adaptability - The system automatically adapts to current channel conditions so as to maintain maximum data throughput. It is also capable of 'learning' about the

channel. This last feature depends solely on the software.

(4) Remote programming - The mobile or portable may be re-programmed by the base station via the channel. This allows flexibility in the coding of the data and can give security against third parties.

(5) Uses existing hardware - The system can use existing hardware either with or without modifications. The existing radio users' investment is thus protected when data communication is required.

(6) Fully automatic operation - If required the system can be fully automatic, even to the point where the system takes charge over the operators. This is an extreme, but it would be possible.

(7) Retains voice operation - This allows users to take advantage of data communication without losing voice facilities.

(8) Broadcast capability - This is not available with normal SAW-ARQ schemes, the use of SRT-ARQ and FEC coding allows a limited broadcast capability.

7.2 SUGGESTIONS FOR FURTHER WORK

It is hoped that this work will lead to further development of the system. There are three areas in which further work is required.

Software

This is where most work is needed, particularly in the case of multi-user, multi-channel systems. Very little software exists for this type of application so any development in this area would be of use. Self-modifying software should be developed allowing the system to 'learn by experience' of new situations.

Hardware

More work needs to be done on direct modulation techniques, principally regarding FSK. The possibility of using analogue processors in the data modulator and demodulator stages could also be examined. The use of such processors would enhance the flexibility of the radio transceivers, putting more 'hardware' characteristics under software control. The automatic selection of binary or multi-level FSK would then become possible.

The simultaneous use of high transmitted bit-rates (>10,000 bits/second) and complex error correction / detection coding necessitates the use of more than one processor. Development of multi-processor systems would be useful, particularly if the microprocessors are not from the same manufacturer.

Simulation

It is unlikely that the practical testing of a multi-user, multi-channel system will be possible, at least initially. This being the case the simulation of such a system would probably be the only way it could be evaluated. Although the simulation of a large system would be a major undertaking, the results obtained should be invaluable in producing an efficient practical system 'first-time'.

The Future

One means of radio communication which is rapidly gaining popularity is spread spectrum, [30,51]. Its use for both fixed and mobile links is being investigated, [44,48]. As most means of producing spread spectrum signals use digital techniques, the incorporation of some of the ideas described in this thesis into a spread spectrum communications system would be possible.

APPENDIX A

Land Mobile Frequency Allocations
in the U.K.

The allocations given on the following pages include only private mobile radio, public radio-telephone and public safety allocations. The list is not exaustive, it is merely presented to give some idea of the frequencies and bandwidths in use. It reflects the situation in 1979, and so does not include any changes made by WARC '79.

N.B. B/W = channel spacing in kHz.

VHF

Allocation (MHz)		Use	Type	B/W
71.500 to	72.800	PMR	duplex	12.5
76.950	78.000	PMR	duplex	12.5
80.000	84.000	public safety		12.5
85.000	86.300	PMR	duplex	12.5
86.300	86.700	PMR	simplex	12.5
86.950	88.000	PMR	duplex	12.5
98.500	104.500	public saftey		12.5
105.000	108.000	PMR	duplex	12.5
138.000	141.000	PMR	duplex	12.5
146.000	148.000	public safety		12.5
154.000	156.000	public safety		12.5
158.525	159.925	public r/tele		25
163.025	164.425	public r/tele		25
165.050	168.250	PMR	duplex	12.5
168.250	169.850	PMR	simplex	12.5
169.850	173.050	PMR	duplex	12.5

UHF

Allocation (MHz)		Use	Type	B/W
425.500 to	429.000	PMR	duplex	25
440.000	443.000	PMR	duplex	25
445.500	446.000	PMR	duplex	25
446.000	446.500	PMR	simplex	25
452.000	453.000	public safety		25
453.000	454.000	PMR	duplex	25
456.000	457.000	PMR	duplex	25
459.500	460.500	PMR	duplex	25
461.500	462.500	PMR	duplex	25
466.000	467.000	public safety		25
468.000	470.000	PMR	simplex	25

APPENDIX B

Typical Mobile Radio Transceiver

Specification

GENERAL

Frequency Range	Dependant on requirements (see appendix A)
Channel Spacing	50/25/12.5 kHz to suit (up to 10 channels can usually be fitted)
Power Supply	12 volts dc
Power Consumption	Receive - 200 mA Transmit - Dependant on power o/p of Tx
Temperature Range	-30°C to $+60^{\circ}$C ambient
Dimentions	24 x 28 x 8 cm (approx)
Weight	3 kg (approx)

RECEIVER

Sensitivity	Depends on mode and frequency but 0.5 µV provides an acceptable signal
Audio Output	2 watts with 10% distortion
Spurious Response	Better than 85 dB except for image at -70 dB.
Squelch	Operates down to 0.3 µV
Frequency Response	±3 dB over 300 Hz to 3 kHz reference 1 kHz

TRANSMITTER

Power Output	Depends on requirements, usually between 5 and 25 watts
Spurious Outputs	Better than 60 dB down on the carrier
Frequency Response	±3 dB over 300 Hz to 3 kHz reference 1 kHz

APPENDIX C BLOCK FORMATS

Inquiry

Base to mobile and mobile to base

33'0's + 101	4 bits	4 bits	16 bits	16 bits
Preamble	Code corresponding to mobile	Code for inquiry	Message length	CRC

Reply

Preamble	4 bits	4 bits	16 bits	CRC
	Code corresponding to mobile	Code for inquiry ack.	Channel	

Data

Base to mobile and mobile to base

Preamble	4 bits	4 bits	3 bits	3 bits	CRC
	Code corresponding to mobile	Code for data header	Code for minor block size	Coding	

Preamble	4 bits	4 bits	16 bits	Data	CRC
	Code corresponding to mobile	Code for data	Start of block in message	64 to 8192 bits	

ARQ Replies

Normal Reply

Preamble	4 bits	4 bits	13 bits	64 bits	16 bits	CRC
	Code for mobile	Code for reply	Position of 1st missing block	Bits corres-ponding to blocks	Channel info.	

Alternate Reply

Preamble	4 bits	4 bits	13 bits	13 bits	16 bits	CRC
	Code for mobile	Code for alter-nate reply	Up to 5 pairs of locations of missing blocks		Channel info.	

'Quick' Repeat

Preamble	4 bits	4 bits	CRC
	Code for mobile	Code for 'quick' repeat	

Channel Clear

Preamble	4 bits	4 bits	4 bits	CRC
	All zeros (only time when all 8 bits are zero)	All zeros	Channel priority code	

APPENDIX D

SOURCE CODE LISTINGS

THE FOLLOWING SOURCE CODE LISTINGS ARE INCLUDED TO SHOW THE BASIC I/O ROUTINES USED BY THE MAIN PROGRAM. THEY ONLY RELATE TO THE TEXAS INSTRUMENTS TMS 9900 AND TMS 9980 MICROPROCESSORS.

```
◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆
◆LABEL DEFINITIONS AND INITIALIZATION◆
◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆
COMCRU EQU   >0000        COMMUNICATIONS CRU BASE
VDUCRU EQU   >1FC0        VDU CRU BASE
◆
RXDI    EQU   >0000       RECEIVE DATA INPUT (BIT)
TXBRF   EQU   >0001       TRANSMIT BUFFER READY FLAG
RXDRF   EQU   >0002       RECEIVE DATA READY FLAG
TXDO    EQU   >0010       TRANSMIT DATA OUTPUT (BIT)
CTXBRF  EQU   >0011       CLEAR TRANSMIT BUFFER READY FLAG
CRXDRF  EQU   >0012       CLEAR RECEIVE DATA READY FLAG
RXLED   EQU   >0015       RECEIVE (SIGNAL PRESENT) LED
TXLED   EQU   >0016       TRANSMITTER ON LED
PTT     EQU   >0017       RADIO TRANSMITER ON/OFF
◆
VTXON   EQU   16          9902 TRANSMITTER ON/OFF
VRRXBF  EQU   18          9902 RESET RECEIVE BUFFER FLAG
VRXBF   EQU   21          9902 RECEIVE BUFFER FLAG
VTXBF   EQU   22          9902 TRANSMIT BUFFER FLAG
◆
TIME    EQU   100         TIME BEFORE ASSUME LOS
◆◆◆◆
INIT    LIMI  0           MASK INTERUPTS
        LWPI  WSP         WORKSPACE
        LI    R12,COMCRU  COMMUNICATIONS CRU BASE
        SBZ   PTT         TX OFF
        SBO   TXDO        TX DATA BITS TO ZERO
        SBO   RXLED       TX LED OFF
        SBO   TXLED       RX LED OFF
        SBO   CTXBRF      CLEAR FLAGS
        SBZ   CTXBRF        "     "
        SBO   CRXDRF        "     "
        SBZ   CRXDRF        "     "
◆◆◆◆
```

```
◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆
◆TRANSMITTER SUBROUTINES◆
◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆
◆LABELS-TXBRF,TXDO,CTXBRF
◆R12=CRUBASE=0000
◆R0=COUNTER/NUMBER OF BITS TO BE TXD
◆R10=WORD/PART WORD TO BE TXD
◆◆◆◆
◆TRANSMIT A ZERO
TXBIT0 TB   TXBRF       TX BUFFER READY?
       JNE  $-2         LOOP IF NOT
       SBZ  TXDO        TX A ZERO
       SBO  CTXBRF      CLEAR-
       SBZ  CTXBRF      FLAG
       RT
◆◆◆◆
◆TRANSMIT A ONE
TXBIT1 TB   TXBRF       TX BUFFER READY?
       JNE  $-2         LOOP IF NOT
       SBO  TXDO        TX A ONE
       SBO  CTXBRF      CLEAR-
       SBZ  CTXBRF      FLAG
       RT
◆◆◆◆
◆TRANSMIT ALL THE WORD IN R10
TXALWD LI   R0,16       TX ALL WORD
◆TRANSMIT WORD IN R10, NUMBER OF BITS IN R0
TXPTWD SLA  R10,1       TEST BIT TO BE TXD
       JOC  BITONE      JUMP IF A ONE
       TB   TXBRF       TX BUFFER READY?
       JNE  $-2         LOOP IF NOT
       SBZ  TXDO        TX A ZERO
       SBO  CTXBRF      CLEAR-
       SBZ  CTXBRF      FLAG
       JMP  NEXBIT
BITONE TB   TXBRF       TX BUFFER READY?
       JNE  $-2         LOOP IF NOT
       SBO  TXDO        TX A ONE
       SBO  CTXBRF      CLEAR-
       SBZ  CTXBRF      FLAG
NEXBIT DEC  R0          ONE BIT GONE
       JNE  TXPTWD      NEXT BIT IF MORE
       RT
◆◆◆◆
```

```
◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆
◆RECEIVER SUBROUTINES◆
◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆◆
◆LABELS-RXDRF,CRXDRF,RXDI,TIME
◆R12=CRUBASE=0000
◆R10=WORD/PART WORD TO BE RXD
◆R9=COUNTER/FLAG TO INDICATE LOSS OF SIGNAL (FFFF)
◆R8=CONSECUTIVE ZEROS COUNTER
◆◆◆◆
◆RECEIVE A BIT-PUT IN EQ FLAG
RXBIT  LI   R9,TIME        LOOPS TILL LOS
RX1    DEC  R9             LOS LOOP DONE
       JEQ  NOTIME         NO TIME LEFT?
       TB   RXDRF          BIT ARRIVED YET?
       JNE  RX1            LOOP IF NOT
       SBO  CRXDRF         CLEAR-
       SBZ  CRXDRF         FLAG
       TB   RXDI           TEST BIT
       RT                  (RXD BIT IN EQ FLAG)
◆◆◆◆
◆RECEIVE A WORD
RXALWD LI   R0,16          RX ALL WORD
◆RECEIVE A WORD, NUMBER OF BITS IN R0
RXPTWD CLR  R10            CLEAR INPUT REGISTER
RX2    LI   R9,TIME        LOOPS TILL LOS
RX3    DEC  R9             LOS LOOP DONE
       JEQ  NOTIME         NO TIME LEFT?
       TB   RXDRF          BIT ARRIVED YET?
       JNE  RX3            LOOP IF NOT
       SBO  CRXDRF         CLEAR-
       SBZ  CRXDRF         FLAG
       INC  R8             INC ZEROS COUNTER
       SLA  R10,1          MAKE ROOM FOR RXD BIT
       TB   RXDI           TEST RXD BIT
       JEQ  RX4            JUMP IF ZERO
       INC  R10            PUT ONE IN R10
       CLR  R8             RESET ZEROS COUNTER
RX4    DEC  R0             ONE BIT RXD
       JNE  RX2            NEXT BIT IF MORE
       RT
◆◆◆◆
NOTIME SETO R9             R9=FFFF AS FLAG
       RT
◆◆◆◆
```

```
**********************************
*THIS LOOP SENDS A BLOCK OF DATA*
*  POINTER IN R3, LENGTH IN R4  *
**********************************
NEWBLK INC  R7             CURRENT BLOCK NO.
AGNBLK MOV  R1,R3
       MOV  R2,R4
       CLR  R6             CHECKSUM
       CLR  R10
       BL   @TXALWD        SEND 16 ZEROS
       BL   @TXBIT1        SEND A ONE
       BL   @TXBIT1        SEND ANOTHER ONE
       BL   @TXBIT1        SEND THIRD ONE
       MOV  R7,R10
       A    R10,R6         CHECKSUM
       BL   @TXALWD        SEND BLOCK NO.
WDLOOP MOV  *R3,R10
       A    R10,R6         CHECKSUM
       BL   @TXALWD        SEND DATA WORD
       MOV  *R3,*R3+       INC R3, AND WAS WORD=0000?
       JNE  SKIP
       BL   @TXBIT1        SEND A ONE AS WORD=0000
SKIP   DECT R4
       JNE  WDLOOP         LOOP IF MORE DATA TO SEND
       MOV  R6,R10
       BL   @TXALWD        SEND CHECKSUM
       CLR  R10
       BL   @TXALWD        SEND 16 ZEROS
       INC  @TOTAL         INC TOTAL BLOCKS COUNTER
*******************************
*WAIT FOR REPLY FROM FAR STATION*
*******************************
       LI   R0,1000        TIME TO WAIT
*LOOK FOR HEADER
HEAD1  CLR  R8             CONSECUTIVE ZEROS COUNT
HEADER DEC  R0
       JEQ  TIMOUT         OUT OF TIME
       BL   @RXBIT
       JNE  SKIP0          ONE?
       INC  R8             ZERO
       INC  R9             LOS?
       JNE  HEADER         NO, CONTINUE
       JMP  HEAD1          YES, START AGAIN
SKIP0  INC  R9             LOS?
       JEQ  HEAD1          YES, START AGAIN
       CI   R8,33          33 ZEROS YET?
       CLR  R8             ONE SO RESET COUNT
       JLT  HEAD1          NOT ENOUGH ZEROS
*FOUND HEADER
       BL   @RXALWD
       LI   R0,>8000       DELAY BEFORE TX
       DEC  R0
       JNE  $-2
       C    R10,R7         BLOCK NUMBERS THE SAME?
       JEQ  NEWBLK         JUMP IF YES
TIMOUT INC  R5             REPEAT BLOCK
       JMP  AGNBLK
****
```

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