Wireless Local Loop and Packet Radio Technology for Developing Communities

by

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Submitted to the Department of Electrical Engineering
in partial fulfillment of the requirements for the degree of

M.Sc.(Elec. Eng.)
at the
UNIVERSITY OF CAPE TOWN
May 1998

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Abstract

In developing countries there is an increasing demand for telecommunications and Internet access. The major hurdle for most of these countries is the lack of adequate telephone network infrastructure. The current problem, especially in Africa, is to find solutions which can provide underdeveloped countries with a satisfactory network. This requires a relatively fast roll-out at low-cost. Modern wireless system claim to offer quick and easy deployment. However, in many cases the wireless system deployment is delayed because adequate radio plans have to be designed, and they are not optimized for Internet and computer communications. Multihop packet radio can offer computer communication in an easily deployable ad-hoc wireless network which does not require frequency planning. However, voice traffic is not easily implemented on packet radio networks due to the bursty and delayed traffic characteristcs. A multihop packer radio network is proposed which offers voice and Internet communications.

Acknowledgements

I thank my parents who have supported me all my years, through thick and thin. Their love, efforts and hard work have helped me complete, amongst many other things, a masters degree in electrical engineering, something I did not think I would achieve. Dear mom and dad, sincerely, thank you very much (ahava ve'todah).

I also thank my brother Dan for all his suggestions. He always let me bounce ideas off him. His careful listening to my questions and problems, and his answers contributed significantly to my thesis. I must also acknowledge his help with latex type setting without which this thesis would not be as well laid out. He also contributed to some of the research by forwarding any papers that he found on the Internet that were significant to my thesis.

I thank my sister for letting me use her computer which she also needed at the same time. She put aside her needs to let me finish my thesis. Thank you.

Thank you to my supervisor, Dr. Robin Braun, for introducing me to packet radio and the help he gave me during my studies.

Praven Ready and I did a small investigation into packet radio and its possible application to voice communications. He helped me with many of the initial ideas that got me enthusiastic about the use of packet radio for voice communications.

I am grateful to Kevin Butchard, Daniel Mashao, and Candace Rennie for the help they gave me with computer and printing related problems that I needed to be solved.

Thank you to the 1997/1998 Digital Communications Radio Research Group masters students for making the studies fun and enjoyable.

Thank you to Jose Pinto and Yon Rosenthal for being my undergraduate study partners. I don't think I would have made it through undergraduate studies in fours years without them.

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Chapter 1

Introduction

In developing countries there is an increasing demand for telecommunications and Internet access. The major hurdle for most of these countries is the lack of adequate telephone network infrastructure. The current problem, especially in Africa, is to find solutions which can provide underdeveloped countries with a satisfactory network. This requires a relatively fast roll-out at low-cost because the demand is large yet these countries do not necessarily hold large foreign exchange reserves to purchase and maintain expensive technologies. Many developing countries use their foreign exchange to purchase these technologies abroad as they lack the technological capabilities to design and manufacture the equipment themselves, although many have the potential.

Wireline telephone networks (or Plain Old Telephone System - POTS) are not suitable for deployment in urban township or rural communities because of the expense of the wire and the work required to lay wire underground or on support structures overhead. Network planning in urban townships is difficult because of the system complexity that results from the close proximity of houses due to the high density population in these areas. In addition to this, deploying the network is difficult logistically because there is little space for equipment in the spacing between houses. It is difficult hard to prevent theft of the wire. Also, roll-out of wireline systems take considerably more time than wireless systems. A wireless system overcomes these problems, except for radio planning which can take a long time to perform. Many cases wireless systems result in a delayed deployment because suitable radio sites have to be located in order to achieve good radio coverage of the proposed area.

Many communities in developing countries tend to be scattered in small low density rural communities and/or concentrated in high density township-like communities. This poses a problem to current standard cellular technology. Current state-of-the-art cellular telephone network technology is designed to work in city-like environments and along major highways and national roads. City environments usually contain various small domains of high density within a larger mainly low density domain.

Township domains tend to be large high density domains containing hundreds of thousands of people in a few square kilometers. For this reason it is difficult to achieve a good radio plan in these areas, especially if more than one provider intends to compete in the same domain. Designing the radio plan requires many measurements of the radio site and takes a long time to achieve thus creating delays in providing the local loop. In addition, even with the use of very small cell sizes the networks may have overcapacity traffic demand. Even if very small cells could be used and two or three cellular providers can compete in the same domain, the cost of the equipment and installation would be expensive because of the large number of base stations that would be required. It would also not be extremely cost effective to provide every family with one or two cellular phones.

Cell phones use voice codecs which in many cases prohibits the use of the Internet due to there information loss characteristics. However, new standards have been made available to overcome this problem, for example cellular digital packet data [CDPD] makes packet data connectivity possible using existing networks [42].

Recently many new systems have been introduced into the market place to provide solutions to some of the above problems. A discussion of some of the systems considered in Cape Town, South Africa, by the previously government owned telecommunications company Telkom, are discussed in Chapter 4. Most of these systems require line-of-site (LOS) between the base station and the user terminal within the township area. This requires the use of tall base station antenna structures. It also usually requires that the user's telephone be attached to a raised antenna on a pole. This results in public dissatisfaction which usually comes from environmentalists or environmental institutions, and often results in deployment delays until a suitable compromise is achieved.

Other factors which affect the practicality and useability of current wireless technologies, such as power sources and suitable areas within the house for placing equipment, are discussed in more detail in Chapter 4.

The new wireless technologies use various channel access protocols which enable communications between the user terminal and base station. These access protocols are usually TDMA, FDMA or CDMA. For these multiple access schemes to function they rely on pre-allocated channel assignments. This means that a channel (in time and/or frequency) is reserved for a particular user throughout the duration of the call. The channel exists between the user and the base station, which has the result that that channel is not available to any other user in the coverage area between the user and the base station. This is a limitation for these systems, and also has the requirement that large antennae be used to cover the distance between user and base station. These systems require a signal to cover the distance between user and base station, thus increasing the power requirement of the equipment. This may be problematic when power breakdowns occur, and the backup batteries used (discussed in Chapter 4).

The emphasis of Chapter 3 is placed on the channel access schemes used and there overall effect on the outcome of current system design.

As a different approach to providing a similar service like the new wireless systems, packet radio used in a multihop network is proposed for use in a wireless local loop system. Various aspects of such a hypothetical system are presented, such as: physical connectivity, channel access, data link control, traffic throughput, routing, acknowledgement schemes, etc. Packet radio offers some advantages over the current systems. For example: reduced power requirements for user equipment, smaller antennae, removes need for direct line-of-site of the base station, almost completely removes the need for radio planning, enables quick deployment. However, multihop network has inherent traffic delays and does not guarantee a high grade of service (GOS).

Thesis development

The original aim of this thesis was to investigate self-organising wireless networks. But the application of the networks were to provide digital voice and Internet access to rural and township communities with previously little or no communication infrastructure. The networks proposed were to be local networks only serving the "last mile" connections to a wireline or wireless local loop system. The reason that self-organising networks were originally of interest is that they were thought to be quick and easy to deploy, requiring little management due to their self-organising ability. It was thought that a multihop packet radio network would be ideal for providing Internet capability because packet radio information transport is based on digital packet routing and transmission.

There were two problems with the original ideas. First, wireless telecommunications systems aimed at developing rural markets are only recently emerging markets. They are specifically optimized for providing digital voice communication, with little direct support for Internet traffic.

Thus, their computer networking ability is limited. In addition, they are not self organising, although some "last mile" DECT systems avoid this problem with dynamic channel allocation. Secondly, most packet radio and multihop packet radio systems were design to cover relatively long distances of the order of kilometers (ground based) or tens of kilometers (satellite based), we are interested in radio over a distance of the order of a hundred meters, say between houses in a local rural community.

Their function was specific to computer networking for carrying data traffic (which is generally not delay critical) and not digitised voice traffic which is delay critical.

Multihop networks have been investigated. They can be made easily deployable and self-organising. However, they are subject to delay, throughput, and routing constraints. These characteristics and others have been researched for the past decade, and are now better understood. The multiple access protocols of multihop networks have also matured, and have been applied to wireless Internet gateways[2], and community computer networks such as the Rooftop Community Network [18].

With the advent of Internet phones enabling people to communicate across continents, it may be possible, in the future, for *localised* wireless internets to carry prioritised digitised voice traffic at an acceptable grade of service (GOS) as well as provide global Internet access. The convergence of these two technologies seems more probable as radio and microprocessor prices decrease throughout the industry and processing speeds increase.

Midway through this investigation a decision had to be made between concentrating on the technical aspects of self-organising multihop networks or researching aspects of providing wireless telecommunications to developing communities which have low teledensity. A substantial amount of literature research on multihop packet radios and technical aspects has been covered. Most published work on multihop packet radio state that performance of these networks is very difficult to study due to there being few closed form solutions to the various network and protocol derivatives, and because they demand substantial time to simulate because of all the issues that have to be taken into account [29] [47].

Already, solutions are being marketed for the problem of providing communications to rural and developing communities. Telkom, a privatised telephone company in South Africa, has began a project to provide a wireless local loop DECT phone system (produced by Alcatel) to residents of Khayelitsha, a densely populated township community in Cape Town. Vodacom have developed the "phone shop" concept using a prepaid GSM system.

Many developing rural and township communities lack a robust electrical power infrastructure. Power supply and electrical safety considerations are briefly discussed.

Another important aspect of wireless voice communication networks is billing. Vodacom, a local cellular network provider, has developed a new prepaid billing system to deal with cultural issues that affect payment.

This thesis is specific to the subject of providing township and rural communities with telecommunications. Issues are approached from various angles, providing the reader with a more holistic view to understanding the current state-of-the-art, and the many technical issues related to the problem at hand. The thesis can be seen as a distillation of knowledge gleaned from various published works, applied to aspects of rural and township wireless communications.

Thesis Chapter Structure

Chapter 2 describes the thesis objectives. Chapter 3 describes fundamental principles used in radio channel signalling. It introduces duplexing and multiple access techniques. Chapter 4 describes the GSM and DECT wireless networks used for providing telephone services to a township community. Some of the problems related to these networks are discussed. It also mentions the possible use of LEO satellites for communication in rural communities. Chapter 5 discusses the fundamental principles of multihop packet radio networks. These principles are used in Chapter 6 to propose a multihop packet radio network, which is applicable to developing township and rural communities, for providing Internet and voice service. Chapter 7 highlights conclusions from the research and mentions recommendations for future study.

Chapter 2

Issues Relating to Wireless Local Loop

The term 'developing communities' is a broad term. Here it is used to refer to rural and township communities, which are highly prevalent in South Africa. Insufficient teledensity is a general characteristic of these communities. In many cases these communities also have underdeveloped water, sewage and electrical power facilities. Studies have shown that increasing teledensity in these communities by rolling out telephone and related communication systems has resulted in increased commercial activity with the result that the communities financial income increases allowing other necessary services such as water and electricity to be further financed and developed.

The prerequisite to fast rollout is ease of deployment, and therefore wireless technology is the first choice.

Most of this thesis serves as a distillation of knowledge from various published works on wireless communication. It is an objective of this thesis to introduce the fundamental radio and channel access techniques which are used throughout the text. The techniques are used to describe GSM and DECT wireless networks which are used for the purpose of providing communications to developing communities which have been previously underdeveloped. Various problems are associated with these systems. A multihop packer radio network is proposed to solve some of the problems that are associated with the GSM and DECT networks.

The main objective of this thesis is two fold: 1) to describe two connection-oriented wireless network technologies that are used in South Africa to provide telephone service, and 2) to introduce the reader to packet radio and propose a multihop packet radio network that can be used to provide Internet and/or telephone service.

We can break the problem into several aspects:

- Technology
- Scenario
- Services
- Economics
- Radio requirements
- Timing requirements

These aspects must be considered when designing a telecommunications system as they have far reaching consequences on capacity, network application, and services the network can render.

2.1 Technology

The decision of which appropriate wireless technology to use in the design of a modern wireless network can be either to use state-of-the-art technology or a mature an well know technology. The technology may be required to be manufacture independent such as a wireless DECT handset (Chapter 4).

Usually wireless communications systems which offer a telephone service require that the system offer voice quality output. This is not always true, the GSM system favours relatively high-speed handset mobility in favour of excellent voice quality (Chapter 4).

2.2 Scenario

The scenario for wireless networks are generally categorized into rural, urban and business (city), or suburban communities. Each scenario will have significantly different traffic capacity requirements.

Rural communities present difficulties because of their location relative to a telephone circuit switching center. For example rural communities can be found more than 100 km from a switching center. This presents a problem to network planners. It is expensive to supply a microwave link or wire based link to these communities which are characteristically small and of low income. The process of laying the wire would require a relatively long time compared to a wireless option and would slow down network deployment. This means that the traffic offered by these communities is small and not sufficient to pay (over a long period of time) for the extended connection from the rural community to the switching center. One method to over come this problem may be the use of low Earth orbit satellites discussed in Chapter 4. There are two feasible choices for wireless connection of rural communities to an exchange. Either a terrestrial microwave link or a satellite link can be used.

Many African rural communities tend to have informal housing. This can be a problem for mounting equipment and may require special mounting platforms to be built, adding time and cost to installation.

In densely populated township communities, like Khayelitsha, minimal space is available for allowing trenches to be dug for laying wire for wire line services. This is an obstacle to quick deployment which results in a logistics problem. Wireless services overcome this problem by removing the need to have the whole network connected by wire.

2.3 Services

Various communications services are available, the most common is POTS (plain old telephone service). Fax and modem (Internet) service are increasing in demand in many businesses and homes. ISDN service is used by many businesses. The public pay phones is another important service provided in many cities. The most modern service available is the mobile telephone.

When designing a system the type of service has to be determined in order to specify technical requirements of the system. It is the objective of this thesis to describe the wireless telephone service supplied to a township community. Its objective is also to propose a packet radio network which can provide a telephone service and an Internet service.

2.4 Economics

The economics of a network are affected by the network reliability, operation and maintenance costs, and installation costs, as well as payment for service which involves billing.

We are primarily interested in installation costs, and operation and maintenance costs. The objective in this regard is to design the proposed packet radio network very simply, with a minimal amount of support hardware, in order to reduce system complexity, and operation and maintenance costs.

2.5 Radio Requirements

A radio network is usually designed to work in a specific part of the radio spectrum. GSM uses the 800 MHz and 900 MHz part of the spectrum. Many new systems are designed to use the 1.8 GHz and 1.9 GHz part of the spectrum.

For a system to become internationally used it should have a part of the spectrum dedicated exclusively for it. This was the approach used in Europe when the DECT system was designed. The whole of Europe assigned the 1880 - 1900 MHz part of the spectrum solely for DECT usage. Therefore all DECT systems built for Europe will work in this frequency band.

In the design of the proposed packet radio network we are more concerned with the concept of the network (i.e. design concept) and less with the actual physical aspect of the operating frequency of the system. The system will need to work at least in the 1 GHz to 10 GHz region of the spectrum.

Wireless Internet gateways (WINGs) [2] use the 905 MHz to 925 MHz frequency band. They have been demonstrated to deliver a channel bit rate of 298 kbps.

2.6 Timing Requirements

Timing requirements relate to the network deployment time, the time for the network to adapt to demand, the planning time, and network penetration forecasts.

We are primarily interested in the network deployment time and planning time. Rapid network deployment is necessary because of the large populations that need to receive basic telephone services. If network deployment is slow then it means that many people will have to wait months maybe even more than a year to receive service. We propose a self-organising network that can set itself up within a day. Anyone who wants to get access to the service would have to buy a network node (radio terminal) and they can connect almost immediately if they have no problems with power supply or mounting. The fact that it is self-organising means that minimal planning is required for frequency plans.

Chapter 3

Multiple Access Techniques

In todays world of telecommunications the radio spectrum is increasingly being used as a result of growing markets and new technologies, resulting in increased demand of a finite amount of usable radio resourse. The result is increased cost for radio spectrum usage. Multiple access schemes are used to allow many users, mobile or stationary, to simultaneously share a common part of the radio spectrum. Manufactures of communication equipment, especially wireless telephone systems, need to utilize the radio spectrum more efficiently, thereby "squeezing in" as many usable channels as possible into as little radio spectrum as possible. Multiple access schemes enable sharing the spectrum to achieve high utilisation by simultaneously allocating available bandwidth to multiple users.

This chapter explains the fundamental concepts of multiple access techniques. It serves as a distilation of information about common wireless access techniques. It contains detailed information about how these techniques are used in a network to achieve multiple access. The way in which these techniques are used has a significant effect on the capacity and easy of deployment of wireless networks. We discuss GSM and DECT networks in Chapter 4, and see how there multiple access schemes affect there ability to offer Internet and voice communications. Before looking at multiple access some fundamental ideas need explanation.

3.1 Physical and Data Link Layers of Radio Networks

This section serves to introduce the reader to the basic practical considerations of a radio network. In particular, we focus on layer one and layer two of the OSI network model, namely the physical layer and the data link layer, respectively.

3.1.1 Physical Connectivity of Radio Connections

The physical layer in any network establishes the link (or connectivity) among network nodes so that information (data) paths can be established between traffic sources and their destinations. In any network making use of wireless technology to transport information link connectivity must be robust in order to transfer error free information at a satisfactory data throughput rate. What follows is a general discussion of frequency considerations in networks using radio links.

The *link connectivity* from node A to node B is B's ability to correctly receive data transmitted by A at a specified minimum rate. The quality of a radio link depends on radio propagation parameters, such as the radio frequency, the distance between nodes, the type of terrain traversed, and the transmission power (signal strength).

The radio frequency is usually regulated by governmental institutions. The distance between nodes determines the propagation time which directly contributes to *network delay*. The type of terrain can be, for example, the sea, or the forrest, or a heavily built-up city, or a densely populated township, and can give rise to signal fading such as Rician fading which is the most common in static networks and terrains.

The transmit power falls off as the radio signal propagates making the information contained in the signal more vulnerable to intereference as distance is traversed. The stronger the transmit power the better the signal-to-noise ratio (SNR) at the receiver.

As stated above, link connectivity is also qualified in terms of a minimum data rate which is directly affected by the channel radio frequency (RF) bandwidth, and the data encoding and modulation schemes. For example, a 16 QAM modulation scheme will be less effective than a GMSK scheme in a typical gaussian radio channel due to its amplitude modulation characteristic which is more susceptable to noise than frequency modulation orientated systems, such as GMSK.

The physical layer defines the bounds of the network in terms of data rate and data accuracy, which ultimately defines the networks possible application in a broad sense. We see that given the type of environment and terrain the network is to operate in, the users location, their requirements in terms of traffic rate and noise immunity, the main design parameters are:

- radio frequency band position
- RF bandwidth

- signalling, encoding, and modulation schemes
- network topology

Usually connectivity between nodes is represented in graph form. The vertices represent nodes and the edges represent link connectivity. Due to fading effects, nodal mobility, wheater conditions, etc., variations in link connectivity may render the network topology graph representation as a probabilistic one [17] due to the fact that the probability of a particular link existing under specific channel conditions may vary.

Lower frequencies results in better propagation because low frequency signals can travel longer distances with less disruption from the terrain. However, lower frequencies restrict the channel to lower data rates. If the user data requirement is satisfied by relatively slow channels then full link connectivity may be achieved and network equipment designe simplified. For higher data rates, higher frequencies and larger bandwidths are required. The power of high frequency RF signals falls off quickly relative to lower frequency signals and they usually requires line-of-sight (LOS) propagation between antennas.

As mentioned before, data rate and accuracy broadly defines the networks range of applications. The higher level network functions define the network application. Initial network design may proceed with an estimate of the range for the required link data rates given user traffic and delay requirements, independently of higher level protocols. This is the approach to initial network design, as design proceeds, the higher levels are further developed.

Frequency Allocation and Bandwidth Issues

When making these estimates, the first issue addressed is the *frequency allocation* to be used. The decision of using low or high frequency determines the bound on data rate, propagation effects due to terrain, and the LOS requirement. The decision determines the degree of connectivity between the network nodes and additional functional support requirements. With high frequency applications radio signal repeaters may be required to achieve good link connectivity over distances.

The other important consideration mentioned above is *signal bandwidth*. The required bandwidth is determined by the user data requirement. Also, spreading the signal, as in spread spectrum or CDMA systems, may be required for improved network performance and operation, and ease of network deployment. The cost of high frequency equipment is decreasing due to improved mass production. This enables the higher end of the radio spectrum to be more usable, and is attracting the attention of many wireless systems manufacturers. Already we see the use of the 1.9 GHz spectrum for personal communication systems (PCS) [14] use in the United States of America, and other locations throughout the world. CDMA and other systems using spread spectrum technology are becoming widely used. It is forecast that by the year 2000, CDMA subscriber units (wireless terminals) will comprise 31 percent of the total subscriber unit sales [1] in the United States of America.

The dynamics of the network topology, such as in mobile terminal networks, which requires network traffic control, also sets a requirement on data bandwidth and therefore the minimum frequency band. Networks with slow changing connectivity, for example, fixed-site amateur packet radio can afford limited control traffic. However, cellular modile networks, such as the GSM and mobile DECT networks, require considerably more control traffic and therefore increased control data bandwidth, due to the rapidly changing connectivity of subscriber units to base stations. This is important in cellular networks that provide a roaming facility which require handoffs [42] between the base stations and subscriber units (mobile terminals) that move between cells.

Propagation and Interference Issues

Link connectivity also depends on the quality of the radio channel. This is significantly affected by terrain type, antenna type, antenna directionality and focus, transmit power and distance between nodes (a node can be a base station and subscriber unit, radio transceiver station in a packet radio network, etc.). Given the terrain, antenna parameters, and node locations, link connectivity is achieved by appropriate selection of data-encoding scheme and transmitter power. For example, in a city environment characterised by high multipathing it is expected that some sort of forward error correction (FEC) be used in the data-encoding scheme, and that the tranmitter power be relatively strong to propagate through buildings. When the radio channel is shared by many nodes, as in a GSM network, multiuser interference and the near-far problem [42] [17] are introduced.

3.1.2 Bandwidth-Time-Space Management

Once frequency allocation and RF bandwidth have been selected, the method for managing the frequency spectrum must be decided upon. The four most common techniques available for allocating bandwidth in time and space are: Frequency division, time division, code division, and space division (or spacial reutilisation) of the bandwidth (section 3.3). Figure 3.1 illustrates this in terms of frequency, time and code division multiple access.

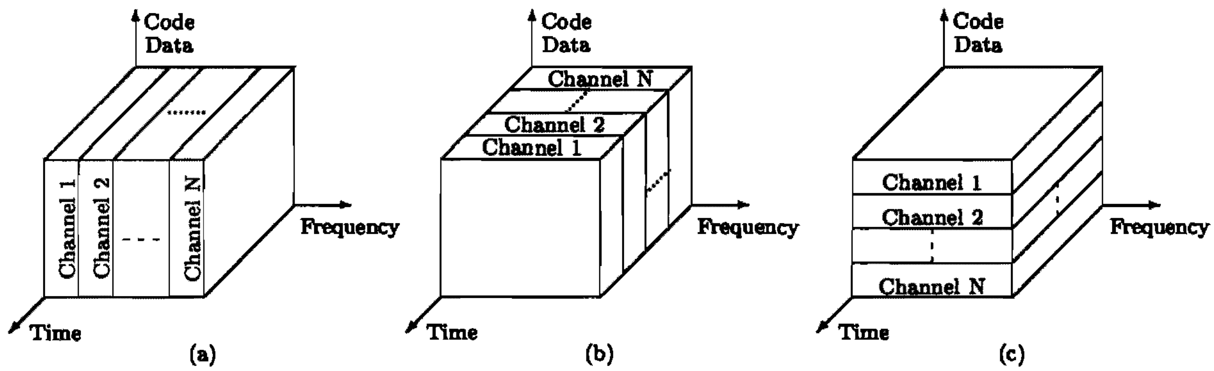


Figure 3.1: (a) Frequency Division, (b) Time Division, and (c) Code Division

Frequency division refers to the partitioning of the bandwidth into sparate radio channels, orthogonal in the frequency domain. Time division refers to the allocation of a given radio channel to different users at different time slots. Code division relies on orthogonal spreading codes, each user of the channel uses a different code, so that users can use the channel simultaneously. Spacial reutalisation of the bandwidth-time resource enables users in different parts of the network to use the same frequencies without interference. This is achieved by keeping users of the same frequencies far enough apart so that their signal strengths are low enough to prevent significant interference from other users. In GSM networks a frequency reuse factor describes the number of times a frequency is reused between base stations.

Frequency Design Considerations and Tradeoffs

The radio spectrum is an important resource which is increasingly in demand. We therefore want to use it efficiently by minimising unused bandwidth. The use of frequency division for providing a channel to be used by a node or node pair may be adaquate if the channel is used for sustained periods of time and the channel is used often and regularly, i.e. the channel utilisation is high resulting in non-bursty and predictable demand. If this is not the case then the bandwidth is being underutilised resulting in low bandwidth efficiency.

Frequency division is useful because it enables provision of several channels which can be shared in the time and code domains by many users (section 3.2.1 and section 3.3). In general however, network deployment, resource allocation and management are easier if all users are tuned to the same frequency band and time or code division techniques are used. Radio channel sharing in the time and code domains is discussed in section 3.2. Spacial reutilisation of the bandwidth-time resource achieves a higher overall utilisation by allowing multiple transmissions to take place simultaneously and with the same code in different areas of the same network. Spacial reutilisation is commonly employed in cellular mobile networks, packet radio multihop networks, and satellite networks which use directed antennae. The antennae at many GSM base stations are sectored in three using directional antennas to increase capacity via frequency reuse.

3.2 Duplexing

In ordinary wireline telephone systems it is possible to hear the person at the other end of the line and to be heard simultaneously. This results from using a full-duplex channel. A half-duplex channel allows two-way communication, but uses the same radio channel for both transmission and reception, thereby allowing one person to be heard at a time. Full-duplexing allows simultaneous two-way traffic while a half-duplex channel only enables traffic flow in one direction at any one time. Duplexing is achievable using frequency or time domain techniques.

3.2.1 Frequency Division Duplexing

Frequency division duplexing (FDD) is achived using two simplex channels. Each user is assigned two frequency bands, a forward band (or uplink) and a reverse band (or downlink). The forward band provides traffic from the base station to the subscriber unit, while the reverse band provides traffic from the subscriber unit to the base station. Each users wireless telephone unit has a device called a duplexer which allows simultaneous reception and transmission on the duplex channel.

Basically, a simplex channel allows communication in one direction. For example a paging system, where the pager receives a broadcast but cannot acknowledge the reception, uses simplex channels. The frequency split (or separation) between the forward and reverse simplex channels is equal for all channels. Each transceiver has to simultaneously transmit and receive radio signals which can vary by more than 100 dB [42], therefore frequencies used for the forward and reverse channels must be carefully allocated to minimize interference from out-of-band users that use the frequency spectrum near the forward and reverse bands. Also, the frequency separation allocation must permit the use of inexpensive radio frequency (RF) technology.

3.2.2 Time Division Duplexing

Time division duplexing (TDD) shares time instead of frequency to provide full-duplexing. If the time split between the forward and reverse channels is small enough then transmission and reception of data appears to be simultaneous and continuous to the user. In TDD, communication is achieved on a single channel and does not necessitate a duplexer as in FDD. However, there is a time latency in TDD because it is not full-duplex in the truest sense [42].

3.3 Multiple Access

Multiple access techniques enables users to share the radio spectrum to achieve high capacity. Frequency division multiple access (FDMA), time division multiple access (TDMA), code division multiple access (CDMA), and space division multiple access (SDMA) are the most widely used multiple access techniques that share available radio bandwidth in a wireless communication system. Each multiple access technique uses the radio spectrum differently. These techniques can be used in narrowband or wideband systems.

Packet radio (PR) networks is another multiple access system used in wireless communications. Packet radio achieves multiple access in a significantly different way to TDMA, FDMA, CDMA and SDMA. Packet radio is discussed in Chapter 5 below.

3.3.1 Narrowband Systems

The bandwidth of a signal can be described with respect to the complete radio spectrum. In relative terms, a narrowband signal occupies much less bandwidth than a wideband signal. Thus many narrowband signals can occupy a wide radio band. Narrowband communication systems divide the available radio spectrum into a number of narrowband channels. This is the reason that narrowband systems usually use FDD. To minimize interference between the forward and reverse duplex channels, the frequency spacing between these channels is made as large as possible without requiring expensive duplexers in the subscriber units. FDMA and TDMA systems exist that make use of narrowband signalling. For example, the GSM system uses TDMA, and the DECT system uses FDMA.

In narrowband FDMA, users are allocated separate channels respectively, each user is assigned one duplex channel. The TDMA equivalent assigns each user to the same channel but this time they are each allocated a respective time slot. This is done in a cylic fashion. For example, if a time frame contains 32 time slots and user A is allocated time slot 3 of the frame, user B is allocated time slot 4 of the frame, etc. Every time the frame cycles, the user knows which time slot it must use, until the link is terminated. The users of a single channel are therefore separated in time. For narrowband TDMA, channels are assigned using either TDD or FDD, where each channel is shared using TDMA. These systems are refered to as TDMA/FDD or TDMA/TDD multiple access systems.

Although modulation techniques are not the subject of this thesis a brief discussion, which follows, has been included for completeness. Since spread spectrum is being increasingly used in modern cellular and wireless communication systems the subject of spread spectrum modulation also mentioned in section /refspread. This paper is more concerned with digital systems as will be seen later when discussing packet radio multihop networks in Chapter 5, and GSM and DECT wireless networks in Chapter 4

Analogue Modulation Techniques used in Narrowband Systems

Modulation is a method of converting a message source to a form suitable for transmission. This usually means translating a baseband signal to a passband signal at frequencies orders of magnitude higher than the baseband signal itself. The baseband signal is called the modulating signal and the passband signal is called the modulated signal. Demodulation is the process of extracting the baseband message from the carrier so that it can be processed by the receiver.

Most analogue narrowband radio systems use frequency modulation (FM) or amplitude modulation (AM), FM being the most popular modulation technique. In FM the amplitude (or envelope) of the carrier is kept constant while the frequency is varied by the modulating message source. The signal information is contained in the phase or frequency of the carrier. In AM schemes the amplitude of the carrier is varied in accordance with the modulating message source. Thus AM signals have all their information in the amplitude of the carrier. With AM signals there is a linear relationship between the quality of the received signal and the power of the received signal, since the signal-to-noise (SNR) at the receiver will determine the distinction between the signal and the noise.

Digital Modulation Techniques used in Narrowband Systems

Modern wireless communication systems use digital modulation techniques. Advancements in very large-scale integration (VLSI) and digital signal processing (DSP) technology have made digital systems smaller and more cost effective than analogue systems. They enable digital error-control codes which detect and/or correct transmission errors. Furthermore, they support complex processing techniques such as source coding, encryption and channel coding.

In digital wireless communications systems, the modulating signal (containing the message) may be represented as a time sequence of symbols or pulses, where each symbol has m finite states. Each symbol represents n bits of information, where $n = \log_2 m$ bits/symbol. Some digital modulation techniques have differences between one another, and each technique belongs to a family of related modulation methods, each with its own advantages and disadvantages.

A good modulation scheme provides low bit error rates at low received signal-to-noise ratios. With this in mind they also have to perform well under multipath and fading conditions. In additon they must make efficient use of bandwidth and be easy and cost-effective to implement.

Bandwidth efficiency describes the ability of a modulation scheme to accomodate data within a limited bandwidth. Increasing the data rate requires reducing the pulse width of the digital symbols, which then increases the bandwidth of the signal. Thus there is a relationship between data rate and bandwidth of binary systems. Each modulation scheme is designed to achieve a trade-off between data rate and bandwidth. The bandwidth efficiency tells how well the bandwidth is utilised. It is defined as the ratio of thoughput data rate per Hertz in a given bandwidth. Bandwidth efficiency η_B is expressed as

$$\eta_B = \frac{R}{B} bps/Hz$$

where R is the data rate in bits per second, and B is the bandwidth occupied by the modulated RF signal. The capacity of a wireless system is limited to the efficiency of the modulation scheme, since a higher η_B implies that more data can be transmitted in a given bandwidth allocation. The fundamental upper bound on achievable η_B is described by Shannon's channel coding theorem [44]. The theorem states that for an arbitrary small probability of error, the maximum possible η_B is limited by the noise in the channel, and is given by the channel capacity formula

$$\eta_B = \frac{C}{B} = log_2(1 + SNR)$$

where C is the channel capacity, in bits per second (bps), B is the RF bandwidth, and SNR is the signal-to-noise ratio.

Digital communication systems often make a trade-off between bandwidth efficiency and power efficiency. For example, adding error control coding to a message increases the bandwidth occupancy which, in turn, reduces the bandwidth efficiency, but also reduces the required receiver power for a particular bit error rate. A good example of this is the codes used for space communications between the earth and distant space probes sent to the reaches of the solar system. On the other hand, some modulation schemes decrease bandwidth occupancy but increase the required received power, and hence trades power efficiency for bandwidth efficiency.

The performance of the modulation scheme under various types of channel fading such as Rayleigh and Rician fading and multipath time dispersion is another key factor considered when choosing a modulation scheme. In cellular systems where interference is an important consideration, the performance of a modulation scheme in an interference environment is extremely important for mobile use. In this thesis the packet radio network suggested is a non-mobile network.

Sensitivity to timing jitter, caused by time-varying (mobile) channels, is also another important factor to consider. In general, the modulation, interference, and time-varying effects of the channel and the performance of the demodulator are analysed via simulations to determine the relative performance of a digital modulation scheme. These considerations may seem unimportant in a non-mobile wireless network, however, these networks may span across highways and other areas where moving vehicles travel, and thereby produce time-varying channel effects.

The most common digital modulation schemes rely on line coding, pulse shaping techniques and intersymbol-interference (ISI) cancellation techniques to achieve the above mentioned tradeoffs. These subjects will not be discussed here; a detail discussion can be found in [42].

A detailed discussion of modulation schemes is describe in [42]. Some common digital modulation schemes are:

- offset quadrature phase shift keying (OQPSK)
- $\pi/4$ quadrature phase shift keying ($\pi/4$ QPSK)
- Gaussian minimum shift keying (GMSK)
- direct sequence spread spectrum
- frequency hopped spread spectrum

The features of some commonly used modulation schemes found in modern wireless multiple access systems is briefly presented below. Direct sequence spread spectrum and frequency hopped spread spectrum modulation is discussed in section 3.3.2 below.

Gaussain minimum shift keying (GMSK) is widely used in GSM cellular networks and in DCS 1800 PCS networks. GMSK is a binary modulation scheme which may be seen as a derivative of the minimum shift keying (MSK) modulation technique. The sidelobe levels of the spectrum of a GMSK signal are reduced by passing the modulating NRZ (non-return to zero) data waveform (message information) through a premodulation Gaussian pulse-shaping filter. The effect of baseband Gaussian pulse shaping is to smooth the phase change between successive pulses of the MSK signal. This reduces the sidelobe levels in the transmitted spectrum due to phase continuity. However, it also introduces ISI into the transmitted signal, but degradation is not severe if the 3 dB bandwidth product (BT) is larger than 0.5 [42]. GMSK has good power efficiency (due to its constant envelope) and spectral efficiency. This is important in wireless systems where interference must be small and bandwidth used efficiently, thereby allowing higher capacity, satisfactory signal quality and minimising cross-talk.

$\pi/4$ shifted QPSK ($\pi/4$ quadrature phase shift keying) is used in the PHS system. $\pi/4$ shifted QPSK is a constant envelope modulation technique. It can be noncoherently detected, which significantly simplifies receiver design. It performs well in the presence of multipath spread and fading. $\pi/4$ QPSK signals are often differentially encoded to facilitate easier implementation of differential detection or coherent demodulation when the signal contains phase ambiguity. Differentially encoded $\pi/4$ QPSK is called $\pi/4$ DQPSK, and is used in the Japanese PHS system, IS-54, and PHS (Personnal Handy-Phone) digital cellular standards.

3.3.2 Wideband Systems

In narrowband and wideband systems users share the total available transmission bandwidth. We saw earlier that narrowband systems divide the transmission bandwidth into many narrowband channels. In wideband access systems the signal bandwidth is spread over the entire transmission bandwidth. The advantage of the wideband technique is that it offers improved performance in channels affected by multipath fading than systems using narrowband techniques. This is also true for frequency selective fading channels.

In wideband multiple access systems, a user transmits over a large part of the spectrum, hence the term wideband. At the same time, many users can transmit on the same channel. Spread spectrum CDMA allows many transmitters to access the channel at the same time. Like TDMA, CDMA systems may be implemented using FDD or TDD multiplexing techniques.

Four common spread spectrum modulation techniques exist. These are: direct sequence, frequency hopped, chirped, and time hopping. Hybrids of these four spread spectrum techniques also exist. Direct sequence and frequency hopped techniques are the most commonly used in modern wireless multi-user communication systems.

Spread Spectrum Modulation Techniques used in Wideband Systems

Narrowband modulation techniques try to achieve increased power and/or bandwidth efficiency in a stationary additive white Gaussian noise (AWGN) channel. Since radio spectrum is a limited resource, one of the primary design objecives is to use it efficiently. Narrowband schemes try to obtain this objective by minimizing the required transmission bandwidth. Spread spectrum modulation techniques achieve efficient utilization of the spectrum in a way that, at first, seems contradictory to the proposed objective.

The transmission bandwidth of a spread spectrum scheme is orders of magnitude larger than the minimum required signal bandwidth (sometimes called data bandwidth for digital modulation schemes). We can say that the scheme is very bandwidth inefficient for a single user, but this does not mean that the spectrum cannot be further utilized. The advantage of spread spectrum is that many users can simultaneouosly use the same bandwidth without significantly interfering with each others signals.

The signal bandwidth of the baseband message source is *spread* by a spreading waveform which is usually controlled by a pseudo-noise (PN) sequence. A PN sequence is a binary sequence that appears random but can be reproduced in a deterministic manner. Crosscorrelation of the received spread signal with the PN sequence effectively *despreads* the signal and restores the modulated message in the narrow band. Crosscorrelation of the PN sequence with the signal from an undesired user or with narrowband interference results in a very small amount of wideband interference at the receiver output. Effectively the noise of other users signal is spread thoughout the received frequency band while the required signal is recovered through crosscorrelation (despreading).

Spread spectrum modulation is well-suited to wireless communication systems because of its inherent interference rejection capabilty. Each user is assigned a unique PN code which is approximately orthogonal to the other codes used. The receiver can separate each user based on their codes, even though they occupy the same bandwidth simultaneously. This means that the interference between spread spectrum signals occupying the same frequency band is negligeble until a certain noise threshold, determined by the number of users, is reached.

Passing the noise threshold can cause a catastrophic breakdown of the channel information rate. Spread spectrum also permits the full recovery (or capture) of a signal even when it is corrupted by narrowband noise. Since all users are able to share the same spectrum, frequency planning is made easier. This requires dynamic PN code assignment, not dynamic frequency allocation in mobile systems.

There are two commonly used spread spectrum techniques, namely, *direct sequence spread spectrum* and *frequency hopped spread spectrum*. A direct sequence (DS) spread spectrum system spreads the baseband data by directly multiplying the baseband data pulses with a spreading sequence that is produced by psuedo-noise generator. Figure 3.2 shows a simplified functional block diagram of a DS system transmitter stage. Data symbols are added in a modulo-2 fashion to the chips of the PN sequence before being modulated. This results in the baseband signal being spread to a wide bandwidth.

Figure 3.2 shows a simplified functional block diagram of a DS spread spectrum receiver. Assuming synchronisation with the spreading sequence has been achieved at the receiver, the received signal passes through the wideband filter and is multiplied by a local replica of the PN code sequence (crosscorrelation). Figure 3.4 shows the received wideband signal with interference at the output of the wideband filter of the receiver, and figure 3.5 shows the correlation output of the receiver after despreading. The received signal bandwidth is reduced to the original baseband bandwidth, as desired. The interference energy is spread over a much wider bandwidth and is consequently many orders of magnitude lower in power than the desired signal in the signal bandwidth. The filtering action of the demodulator removes most of the interference. Thus most of the interference energy is removed and does not significantly affect signal fidelity. The effect of spread spectrum techniques is that processing gain is increased, and thus the interference rejection capability of the system is enhanced.

Frequency hopped (FH) spread spectrum involves a periodic change of transmission frequency. The change is determined by the PN sequence. A frequency hopped signal may be thought of as a sequence of modulated data bursts with a time-varying, psuedorandom carrier frequency. The data bursts have a bandwidth called the instantaneous bandwidth. The set of

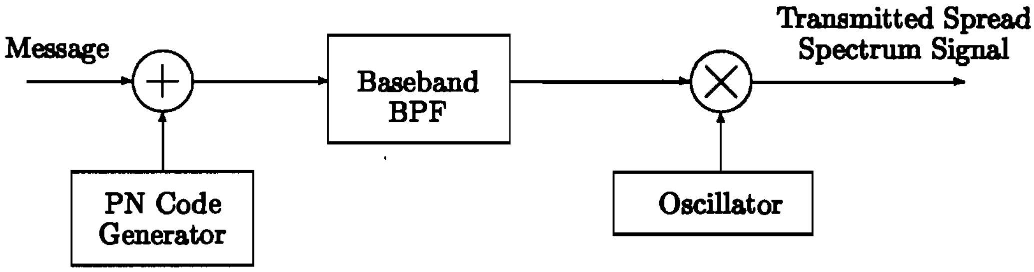


Figure 3.2: Simplified block diagram of a DS spread spectrum system: transmitter

possible carrier frequencies is called the hopset. The hopping takes place within a frequency band that is divided into a number of channels. Each channel is defined as a spectral region with a central frequency in the hopset and a bandwidth that can accommodate the instantaneous bandwidth burst of the hopped signal. The bandwidth of the spectrum over which the hopping occurs is called the total hopping bandwidth. Information is sent by hopping the transmitter carrier to random channels which are known only to the desired receiver. So small bursts of data are sent in the randomly selected channel before the carrier hops to the next channel.

Figure 3.6 shows a simplified functional block diagram of a single channel FH spread spectrum system transmitter, and Figure 3.7 shows the receiver. If the frequency hop pattern of the receiver synthesizer is synchronized with the same frequency pattern of the received signal, then the mixer output will be the dehopped signal. The mixer output is then fed to a conventional receiver to extract the baseband signal. It is possible to have collisions in a FH system where an undesired user transmits in the same channel at the same time as the desired user.

3.3.3 Frequency Division Multiple Access (FDMA)

As the name implies frequency division multiple access (FDMA) uses frequency division to manage the bandwidth resource. Each user is allocated a unique frequency channel. These channels have to be assigned on demand to users requiring service. This requires the network nodes to have management protocols to dynamically allocate the channels. During the call connection period the network protocols do not allow any other user to access the channel. FDMA systems use FDD. A channel is assigned

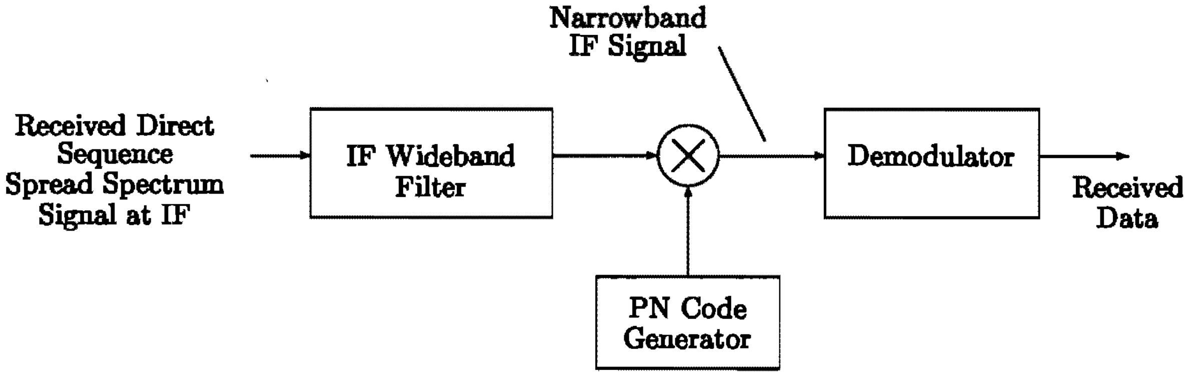


Figure 3.3: Simplified block diagram of a DS spread spectrum system: receiver

as a pair of frequencies, one frequency for the forward channel, and the other for the reverse channel. These channels complete the local loop or circuit.

FDMA becomes wasteful of the bandwidth resource when a channel remains idle because the channel cannot be used by other users to increase or share network capacity. The bandwidth of an FDMA channel is (usually between 25 kHz and 30 kHz), and usually implemented in narrowband systems. The amount of intersymbol interference (ISI) is relatively low due to orthogonality in the frequency domain, therefore requiring little or no frequency equalisation.

FDMA is relatively uncomplex when compared to TDMA and CDMA mobile systems. FDMA requires less control data than TDMA and CDMA systems once the call connection has been established because it is a continuous transmission scheme, and does not require overhead data such as synchronisation and framing bits.

Each channel in an FDMA system requires a carrier; this requires the base stations to have costly bandpass filters to eliminate spurious radiation in order to reduce signal distortions. This results in FDMA systems having higher cell site costs compared to TDMA systems [42].

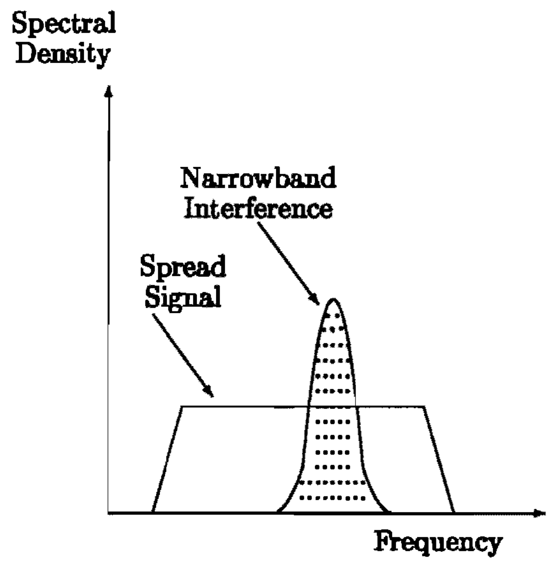


Figure 3.4: Spectra of a desired received spread spectrum signal with interference: wideband filter output

3.3.4 Time Division Multiple Access (TDMA)

Time Division Multiple Access (TDMA) systems divide the radio spectrum into time slots which are arranged into a frame that periodally cycles. One user is allowed to either transmit or receive in each time slot at any one time. A channel can be thought of as a particular time slot that repeats every frame of time slots. In other words, a channel corresponds to a specific time slot of the repeating frame. The number of time slots per frame is usually defined in a standard or for a particular system. GSM uses TDMA with a time frame consisting of eight time slots. Due to the fact that TDMA systems use time slots to manage data a TDMA system is non-continuous. The implication is that digital data and digital modulation are required to enable TDMA systems to operate. The various transmissions are interlaced into a cyclical frame structure, shown in the figure 3.8.

As mentioned above each frame is made up of a number of time slots. In addition to the time slots, overhead information is neccessary for frame boundary definition and synchronisation. Each frame consists of a preamble, information message, and trail bits. TDMA/TDD systems use half the time slots of a frame for forward link channels and the other half for reverse link channels. TDMA/FDD systems also exist. They use a

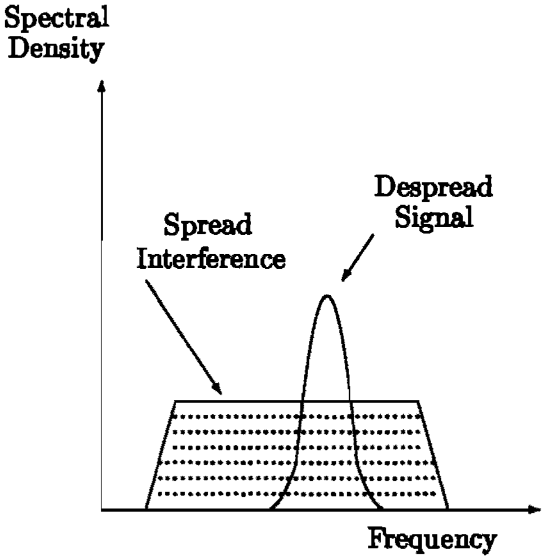


Figure 3.5: Spectra of a desired received spread spectrum signal with interference: correlator output after despreading

similar frame structure, but now the forward and reverse simplex channels use different carriers. In order for the base stations and subscriber units to identify each other, address information is placed inside the preamble.

Guard times are utilised to enable the receivers at the base station and subscriber unit to achieve synchronisation between the different time slots and frames. In order to increase capacity guard times should be minimised. If the guard times are shortened by sharply suppressing the transmission at the edges of the time slot then an expanded transmission spectrum will be induced thereby causing interefernce with adjacent channels. Various TDMA frame structures exist for different applications. The fact that TDMA transmissions are slotted results in TDMA systems having larger overheads compared to FDMA systems. The number of time slots per frame and the size of each time slot depends on several factors such as the modulation technique used and the available bandwidth.

In TDMA systems users are allocated non-overlapping time slots. As a result data transmission for users is not continuous, but occurs in burst. This reduces battery consumption since the subscriber unit transmitter can be turned off when the channel is idle, unlike in FDMA systems where the transmission is continuous requiring the subscriber unit transmitter to always be active during a call connection.

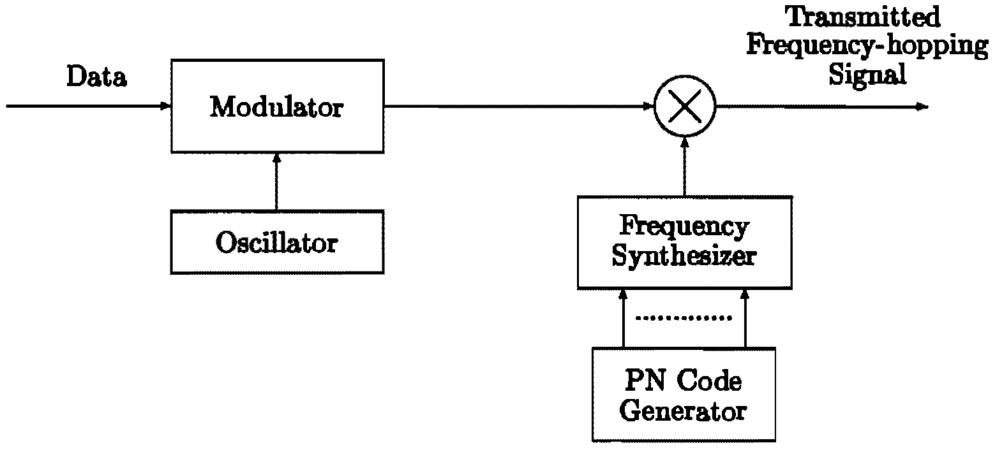


Figure 3.6: Simplified block diagram of a frequency hopping (FH) system with single channel modulation: transmitter

If a handoff process is required TDMA also enables subscriber units to listen, during idle time slots, for other base stations, thereby making the handoff process simpler than in FDMA systems. Some TDMA systems use mobile assisted handoff (MAHO) in subscriber units by listening for idle time slots in the frame.

It has been mentioned that different time slots are used for transmission and reception. This excludes the use of a duplexer in TDMA systems. If TDMA/FDD is used then a switch can be used inside the subscriber unit to switch between transmission and reception time slots, rather than using a duplexer as in FDMA systems. This results in less expensive radio equipment.

Another advantage of TDMA slotted transmissions is that it is possible to allocate different numbers of time slots per frame to different users. The effect is that bandwidth can be supplied on demand to different users by concatenating time slots based on priority.

The efficiency of a TDMA systems is a measure of the amount of transmitted data containing information compared to the overhead data of the access scheme. In TDMA systems, information is in binary form. Therefore the measure will include the binary overhead control data compared with

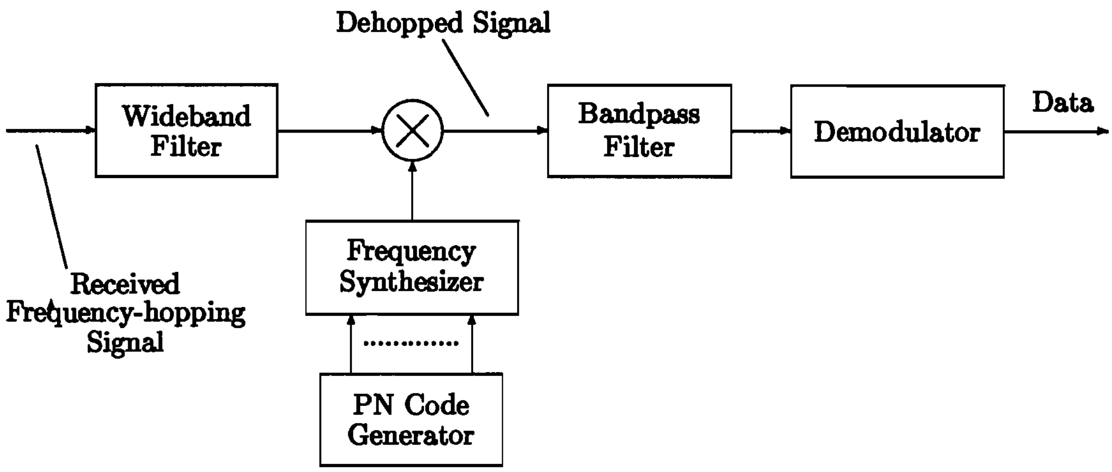


Figure 3.7: Simplified block diagram of a frequency hopping (FH) system with single channel modulation: receiver

the digitized voice information in the time slots. The frame efficiency is the percentage of bits per frame which contains transmitted data. The transmitted data may contain source and channel coding bits to improve data accuracy, so the overall end-user efficiency of the system will in most cases be less than the frame efficiency.

3.3.5 Spread Spectrum and Code Division Multiple Access (CDMA)

Up to this point we have looked at FDMA and TDMA applied in a narrow band sense. We now look at spread spectrum techniques which allow for wideband multiple access techniques. Spread spectrum multiple access techniques make use of signals with bandwidths several orders of magnitude larger than those used in narrowband systems. Code division multiple access (CDMA) will be presented followed by a brief discussion of frequency hopped multiple access (FHMA).

Code Division Multiple Access (CDMA)

The modulation of spread spectrum signals has been described above. This spreading signal is a psuedo-noise (PN) code sequence. All users of a CDMA system use the same carrier frequency and thus may transmit simultaneously in the same bandwidth. Each user is assigned a different psuedorandom codeword which is approximately orthogonal to all other

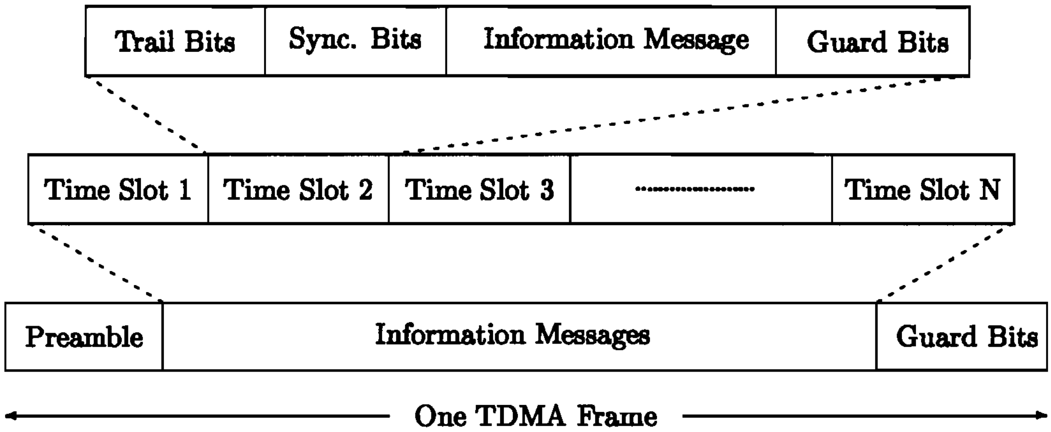


Figure 3.8: TDMA frame structure

codewords. In order for the receiver to detect a specific codeword it has to perform a correlation operation. All other codewords will be decorrelated and appear as noise. If the receiver wants to detect a specific message signal, then it as to have a priori knowlege of the signals codeword. Due to the fact that all codewords are approximately orthogonal, each user operates independently. The system is designed so that a maximum number of users can be serviced without high interference levels.

CDMA systems are affected by the near-far problem. The noise floor after decorrelation at a receiver base station is determined by the signal power of multiple users. The near-far problem occurs when many mobile users share the same channel. The strongest received signal will be *captured* by the demodulator at the base station. The result is that stronger received signal levels raise the noise floor for the weaker signals at the base station. This decreases the probability of weaker signals being received properly. To overcome this problem, CDMA systems use power control to regulate the transmission power of signals. Power control is performed by each base station and insures that each subscriber unit provides similar signal power levels at the base station. However, out-of-cell subscriber units provide interference which is not under the control of the receiving base station, and therefore limits the capacity of the system.

CDMA has a soft capacity limit due to its interference characteristics. The noise floor increases with increasing number of users and decreases with decreasing number of channel users, in a linear manner. The effect is that system performance and voice quality is determined by the noise level. Therefore, ideally there is no absolute limit on the number of users in CDMA. Usually the number of users is limited by timing and synchronization problems with radio and digital equipment.

CDMA can be used inconjunction with FDD and TDD. CDMA is versatile because it can be used to improve the operation of many existing communication system designs which implement TDD and FDD.

The effects of multipath fading is reduced because the signal is spread over a wide bandwidth. Usually the spread spectrum bandwidth is much larger than the baseband signal bandwidth and any small-scale fading has little effect on the information accuracy of the demodulated signal.

CDMA is also very convenient for frequency planning and network deployment. In narrowband systems special dynamic frequency allocation is required or time consuming frequency planning is required to prevent frequency channel clashes. In CDMA systems frequency planning is not an issue because all users use the same frequency band. Once users are assigned codewords they can use the channel without significantly affecting adjacent coded signals.

Frequency Hopped Multiple Access (FHMA)

The modulation of spread spectrum signals has been described in section 3.3.2. Frequency hopped multiple access (FHMA) is a digital multiple access system in which the carrier frequencies of the individual users are varied in a pseudorandom fashion within a wideband of radio spectrum. The digital data is divided into uniform bursts which are transmitted on different time-varying carrier frequencies. The varying carrier frequencies of the multiple users randomizes the channel occupancy of the channels at any given time. This effectively allows multiple access over a wide frequency band. In a FH receiver, a locally generated PN code is used to synchronise the receivers instantaneous frequency with that of the transmitted signal. The PN code describes the carrier frequencies included in the hopset. Each user is separated in the code domain by having different hopping sequences. The result is that they each mostly use different parts of the spectrum at any one time. Frequency hopped signals can be made very immune to deep fading by using error control coding and bit interleaving. Error control coding and bit interleaving can also be combined to protect against collisions which occur when more than one user simultaneously transmits on the same channel.

Frequency hopping provides a certain level of security to the information in the FH channel, since an intercepting receiver which does not know the psuedorandom sequence of the frequency slots would need to rapidly search for the signal it tries to intercept, or uses some other advanced technique to acquire the information. This may reduce complex encryption techniques.

3.3.6 Space Division Multiple Access (SDMA)

Space division multiple access (SDMA) increases network capacity by controlling the radiated energy for each user in space. Directed antennas are used to serve different users. TDMA, FDMA and CDMA systems can be used inconjunction with SDMA to serve different coverage areas. Sectorised antennae use SDMA in a static sense. Research activity has recently increased in the field of adaptive arrays, which will be able to perform beam steering in the direction of many users at once. This will bring about an increase in network capacity. In the mobile environment this is necessary due to the movement of subscriber units.

The reverse link (signals from the subscriber unit to the base station) present many difficulties in cellular SDMA systems. Different users will generate different radio propagation paths to the base station, each with its own power level. In order to prevent high interference levels at the base station receiver the power levels in the reverse link need to be dynamically controlled by the base station, adding complexity and cost to the system.

Another difficulty is that transmit power is limited to the battery capacity and consumption at the subscriber unit. This will limit the amount that power at the subscriber unit can be controlled. Two techniques can be used to increase battery life. Base stations can spacially filter each desired user so that more energy is detected in the reverse link of each subscriber unit, then less power is required from the subscriber unit. The other method, though more theoretical at this time, would be to use adaptive antennas on the subscriber unit to direct transmit energy and therefore prolong battery usage.

In a non-mobile network spacial filtering at receivers, and directed antennae at transmitters counld increase system capacity, and reduce power requirements. However in a multihop packet radio network a multiple of directed antennae would need to be used to connect to multiple neighbouring nodes.

Chapter 4

Wireless Local Loop Technologies

In this chapter we look at wireless local loop (WLL) technologies and some of the systems used in South Africa. Wireless local loop is sometimes called radio in local loop (RLL); the terms are used interchangebly. Two types of systems which are currently being used in South Africa are presented. The systems are explained in detail, followed by information detailing the success achieved and problems encountered using these systems.

The purpose of providing wireless communication systems in South Africa is to bring about an information based society. "The vision of the department is to improve the quality of life of all our people, make South Africa's future generations a knowledge-based society and help create an information economy. This will be achieved by establishing a networked information community to empower the way people work, live and play and to make South Africa globally competitive." [8].

The objective of an information based society requires providing both voice and computer networking infrastructure in order to provide multimedia applications. Rural communities can benefit from these technologies which can be used to provide tele-medicine and tele-education services. These services are provided by a combination of telephone and Internet services, which most South Africans currently do not have direct access too.

The original aim of this thesis was to propose a packet radio system which would be able to provide both voice and internet access in one network. Research and analysis of published works has been used to suggest such a system in Chapter 5. However, before looking at this packet radio proposition we look at the current WLL systems in use today, and highlight some of their disadvantages. The concepts relating to multiple access described in Chapter 3 are used here for this purpose.

4.1 Wireless Local Loop Issues

The purpose of WLL technology is to replace the service that the traditional wireline POTS systems have provided in the past or to extend existing public switch telephone networks (PSTNs). The local loop traditionally consists of a twisted copper wire pair, either underground or overhead on poles. Wire based services are slow to roll out due to the use of wire connections. Recently WLL has been used to rapidly and cost effectively provide local loop service in countries with low telephone penetration [6].

This has come about due to advances in technological developements, such as digital and RF circuit fabrication improvements, new large-scale circuit integration (VLSI), and miniturisation technologies reducing size of portable radio equipment and making them cheaper and more reliable. Digital switching technology has also facilitated the deployment of cheaper, easier to use and managable radio communication networks.

Market deregulation has also played a major role in the development and deployment of wireless local loop technology. Increased competition between suppliers has resulted in driving prices down, improving network services, and operation and maintainance monitoring.

Wireless local loop is seen as an initial platform for the delvelopment of multimedia and news services. Multimedia applications require high data rates which therefore require the use of high frequencies and large bandwidths. 2 GHz and 10 GHz wireless local loop has been proposed by the British government for this purpose. High frequencies are suited to last mile wireless local loop multimedia because of their data carrying capacity. The short propagation distance at these types of frequencies is not a factor because they can accomodate last mile coverage.

Wireless local loop will stimulate the introduction of competition between last mile local loop providers. Radio technologies can be used effectively by smaller providers to compete with the historical sole telephone operator that exists in most countries to gain access to the subscriber without having to first bear the cost of investing in wireline networks. The use of radio infrastructures make it possible to establish networks rapidly and more cost effectively than wireline networks.

The multiple access capability of WLL systems makes it simpler to plan the distribution network because it is not necessary to know the exact location of a subscriber. Radio networks can increase capacity according to increasing subscriber demand by sectoring antannae or using microcells. This allows for a flexible network investment plan based on subscriber numbers and demand, rather than in the case of wire based solutions where networks are fixed according to predicated subscriber demand. This means that capital investments for radio based networks can be recouped quicker than wireline networks.

Wireless local loop contributes to reducing the cost of a telephone service because network installation and maintenance costs are lower due to the absence of special cabling required for wireline networks. This is particularly relevant to distant rural communities where it would be expensive to install and maintain wire based networks over many kilometers. Wireless connection costs do not vary substantially with distance [6] compared to wireline costs.

Another aspect of WLL that is pertinant to developing countries is that WLL systems can be rolled out and deployed very quickly compared to wire based systems. The time needed to link a new subscriber with a wireless system is measured in hours or days not months.

It is not difficult to estimate the cost to install a WLL system. This enables service providers to develope an accurate pricing model which allows then to provide a more efficient and well managed service.

4.2 Wireless Local Loop Technologies

Manufacturers of modern radio local loop equipment offer a diverse range of systems that fall into the following categories:

- fixed radio access system
- point-to-point (PTP) system
- point-to-multipoint (PMP) system
- cordless system
- cellular system

These systems are used alone or in combination to provide local loop service to different communities with different communication requirements. Communities and their communication requirements may be the following:

- local loop in densely populated area, fixed radio access system

- local loop in densely populated area, with mobile communication service
- local loop in moderate or sparsely populated areas
- local loop in very sparsely populated areas

4.2.1 Point-to-Point (PTP) Systems

Point-to-point wireless systems have traditionally been used in fixed access microwave radio links to establish network infrastructure. They can be used to provide wireless local loop services to remote rural communities. For example, France Telecom uses UHF and VHF frequency systems developed specifically for certain rural communities.

4.2.2 Point-to-Multipoint (PMP) Systems

Point-to-multipoint systems offer fixed wireless access. They provide a single access point for multiple radios. They use directed antennae to form links to these radios, and therefore are not intended for continuous coverage. They are normally used as a backbone in network infrastructure (figure 4.1).

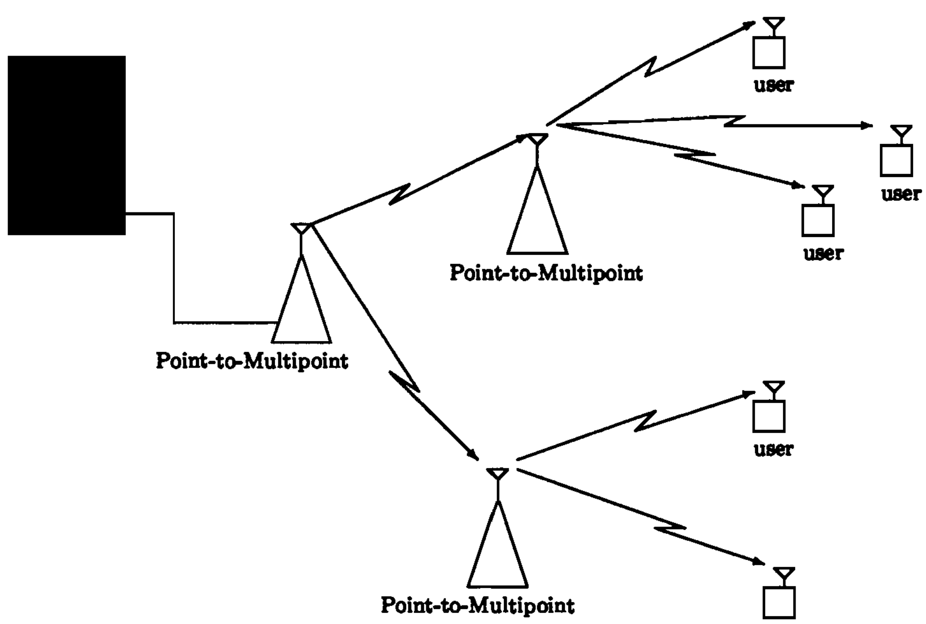


Figure 4.1: A Point-to-Multipoint Network

Usually a subscriber terminal is connected via wire to an external antenna, say on the subscribers house roof or on a pole near the house.

The point-to-multipoint system connected directly to the PSTN (figure 4.1) (is the first) in a wireless link will limit the capacity of the system, therefore determining the maximum number of simultaneous users that the wireless network can provide throughout the chain of wireless links that follow after the point-to-multipoint system at the PSTN.

4.2.3 Cordless Systems: DECT and CT2

Digital European Cordless Telephone (DECT) and CT2 are European developed systems which are currently used in South Africa by Telkom. They provide fixed subscriber access, radio telephony with varying degrees of mobility, PABX and private terminal use. DECT systems can accomodate data and voice transmissions for the office and business users. They are also being used in South Africa to deliver last mile telephone service to a township community.

DECT was developed from the need to coordinate radio spectrum usage between countries in Europe. It is widely used in Europe and Asia. DECT offers a similar service to PHS (Personal Handyphone System) in Japan. DECT is a universal cordless telphone standard developed by the European Telecomunications Standards Institute (ETSI) [39] [38] and finalised in July 1992.

4.2.4 Cellular Systems

The worlds first cellular system was implemented by the Nippon Telephone and Telegraph company (NTT) in Japan during 1979. In 1985 the Extended European Total Access Cellular System (ETACS) was deployed, and is very similar to the U.S. AMPS systems. GSM (Global System for Mobile) was developed in Europe and deployed in 1991 in the 900 MHz band which all European countries dedicated for cellular telephone service [35].

GSM is gaining world wide acceptance as the first universal digital system, and is a strong contender for personal communication system (PCS) services throughout the world.

These systems now offer access to radio local loop. Some manufacturers have developed fixed cellular systems. For example, Qualcomms QCT series offers CDMA technology based fix wireless access systems [11]. Qualcomm claim that their CDMA system offers three times the capacity of regular cellular networks, and is more easily deployable than GSM systems because it uses dynamic channel code assignment.

4.3 Global System for Mobile (GSM)

GSM access in South africa is provided by two cellular companies, namely MTN and Vodacom. They were issued with licenses in 1993 [4]. In this section a brief explanation of GSM is given. GSM is a widely used network for mobile users. We will see in this section that it has been applied to township communities in order to provide residents access to telephony, and has resulted in the development of commercial ventures.

4.3.1 System Architecture of the GSM Network

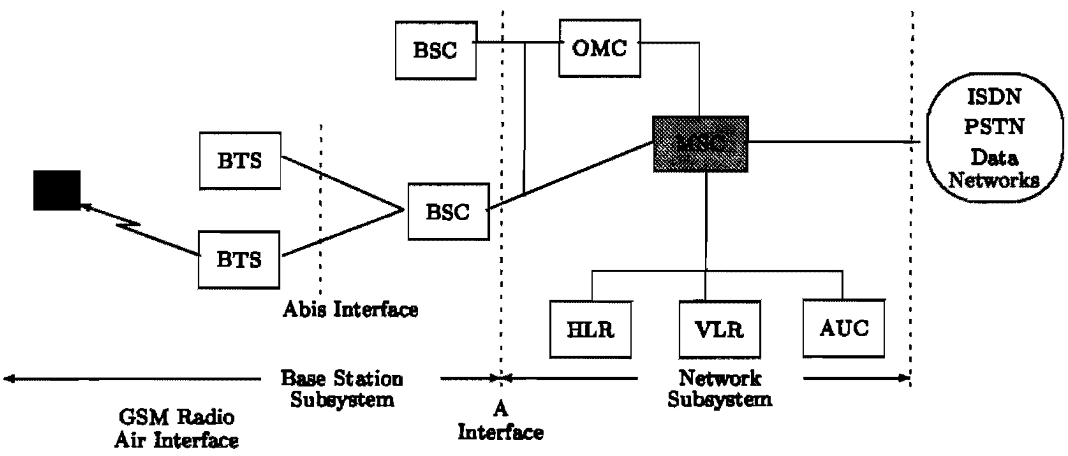
The GSM network is composed of three functional interconnected subsystems: *Base Station Subsystem (BSS)*, *Network and Switching Subsystem (NSS)*, and the *Operation Support Subsystem (OSS)*. The *mobile station (MS)* (subscriber unit) is considered part of the BSS. Figure 4.2 shows the layout of a generic GSM network. The BSS controls the radio link with the mobile station, and manages the radio interface between the mobile station and all other subsystems of GSM. The NSS performs the call switching between the mobile and other fixed or mobile network users, and performs mobility management. The main part of the NSS is the *Mobile service Switching Center (MSC)*. The *Operations and Maintenace Center* oversees the proper operation and setup of the mobile-base station interface, known as the air interface or radio link.

We will show a general description of how these subsystems work together, thereby showing the complexity of a GSM network. It will be seen that the GSM network provides functions which are not necessary in a fixed wireless network.

Mobile Station

The mobile station (MS) consists of the mobile equipment (radio terminal) and a smart card called a Subscriber Identity Module (SIM). THe SIM provides personal mobility, which allows the user to gain access to the GSM network irrespective of the mobile terminal equipment.

The mobile equipment is uniquely identified by the International Mobile Equipment Identity (IMEI) number. The SIM card contains the International Mobile Subscriber Identity (IMSI) number used to identify the subscriber to the system. The IMEI and IMSI are independent. The



Key:
MS Mobile Station
BTS Base Transceiver Station
BSC Base Station Controller
AUC Authentication Center
HLR Home Location Register
VLR Visitor Location Register
MSC Mobile service Switching Center
OMC Operations and Maintenace Center

Figure 4.2: Architecture of a generic GSM network

SIM card can be protected against unaurthorised use by using a password.

Base Station Subsystem

The base station subsystem consists of the base transceiver station (BTS) and the base station controller. The BTS and BSC interact via the standardised Abis interface, which allows system components of different manufacturers to operate together.

The BTS contains the radio transceivers that are used for communication within a cell, and handles the radio link protocols with the mobile station. In high density traffic areas many BTSs are deployed to cope with the demand. Therefore BTSs are required to be robust, reliable and low cost.

If GSM was to be used in rural communities, then the base station will not be required to handle such large traffic loads as those that arrise from GSM networks deployed in cities. Thus it may be a waste of capital to provide base station type support in some rural communities.

The BSC manages the radio resource between one or more BTSs. It performs the radio channel setup, handovers, and frequency hopping. The

BSC also connects the mobile station to the MSC. The standardised interface between the BSC and MSC is called the A interface.

Network Subsystem

The NSS handles the switching of GSM calls between external networks and the BSCs in the radio base station subsystem, and is responsible for managing and providing external access to several customer databases. The MSC is the central component of the network subsystem. It acts as a switching node for PSTN or ISDN connections, and provides all the functionality required to handle a mobile subscriber, such as roaming, handoff, authentication, call routing, etc. Signalling between functional network components in the NSS uses Signalling System (SS7). SS7 is used for trunk signalling in ISDN networks which are extensively used in GSM networks. Telkom leases ISDN networks to Vodacom and MTN for use in their GSM networks because it has a temporary control over all voice communication networks.

All call routing and roaming capabilities of a GSM network are provided by the HLR, VLR and MSC (describe in figure 4.2. The HLR contains a database with information about each subscriber and the subscribers location in the same area as the MSC.

The visitors location register(VLR) contains selected information from the HLR. The VLR temporarily stores the IMSI and customer information for each *roaming* subscriber. In other words, the VLR will contain subscriber information for each mobile currently located in the geographic area controlled by the VLR. The VLR is usually linked between multiple MSCs in a particular geographic region and contains subscription information about every visiting user in the region. The MSC itself does not contain any information about particular mobile stations, this information is stored in the location registers.

Two registers are used for authentication and security purposes. These are the equipment identity register (EIR) and the authentication center (AUC). The EIR is a database that contains a list of all valid mobile equipment on the network. Each mobile station is identified by its international mobile equipment identity (IMEI) number. IMEIs can be marked as invalid if the equipment is stolen. The AUC is a protected database that stores a copy of the secret key stored in each subscriber's SIM card, which is used for authentication and encryption over the radio channel.

The operation support system (OSS) one or more operation maintenance centers (OMC) are used to monitor and maintain the function of each MS, BS, BSC, and MSC within the GSM network. The three main functions of the OSS are:

- to maintain all telecommunications hardware and network operations within a particular market
- manage all billing procedures
- manage all mobile equipment in the system

The OMC has the ability to adjust all base station parameters and billing procedures, and allow network operators to determine the network performance and integrity of each piece of subscriber equipment [42]. While a network such as GSM requires this ability to ensure smooth operation, a rural network of non-mobile topology woulb not require such an advanced ability. At most it would require a simple billing system and network integrity check ability.

4.3.2 Multiple Access and Channel Structure

GSM uses two 25 MHz frequency bands, in the 800-900 MHz range, one 25 MHz band accomodates forward channels and the other reverse channels. GSM uses FDD and a combination of FDMA and TDMA to provide multiple access to multiple users. The forward and reverse frequency bands are divided into 200 kHz channels called ARFCNs (Absolute Radio Frequency Channel Numbers) using FDMA. Each forward and reverse band channel is separated by 45 MHz and each channel is time shared between as many as eight subscribers using TDMA.

Each of the eight subscribers uses the same ARFCN and occupies a unique timeslot (TS) per frame. GMSK modulation with a BT factor of 0.3 is used for transmissions at a channel rate of 270.833 kbps. The effective channel rate per user is 33.854 kbps. However, the GSM overhead results in data actually being sent at 24.7 kbps. Figure 4.3 shows the GSM frame structure.

Appendix A contains a tabular summary of the GSM air interface specifications.

GSM Traffic Channels

GSM has two types of logical traffic channels: the *traffic channels* (TCH) and *control channels* (CCH) [27]. The traffic channels carry digitised user speech data or user data. The control channels carry signaling and synchronisation commands between the base station and the mobile station.

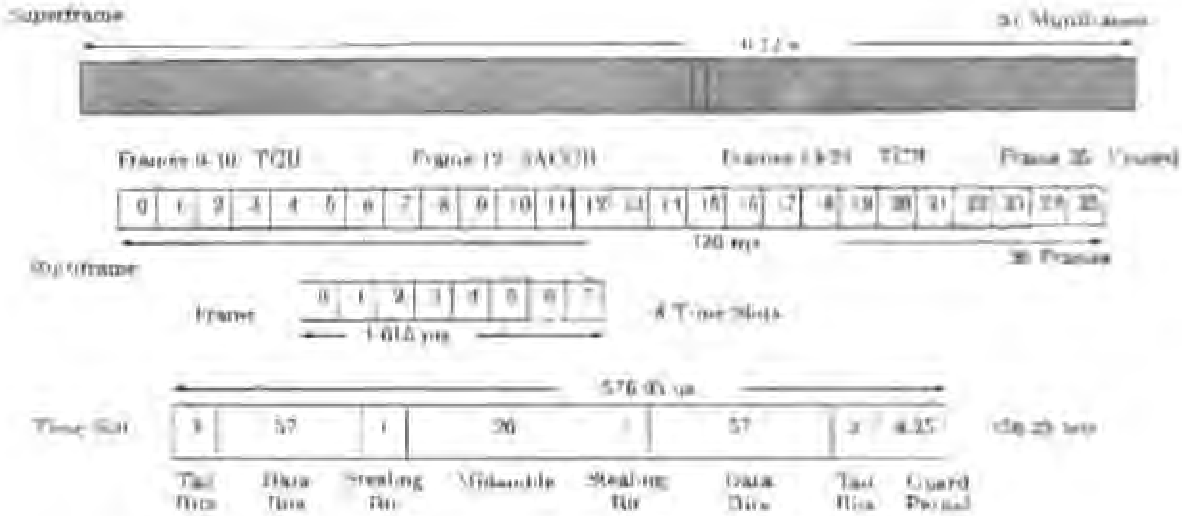


Figure 4.3: GSM Frame Structure

The TCH channels are defined using a 26-frame multiframe, or a group of 26 TDMA frames. The length of the multiframe is 120 *ms*, from which the burst period is defined (120 ms / 26 frames / 8 user periods (burst periods) per frame). Of the 26 frames 24 are used for traffic, 1 for the Slow Associated Control Channel (SACCH) and one is unused. The TCH channels for forward and reverse links are separate in time by 3 time slots. This is done so that the mobile station does not have to transmit and receive simultaneously, and simplifies the base station and mobile station electronics. The 26-*th* frame contains idle bits when full-rate TCHs are used, and it contains SDCCH (Stand-alone Dedicated Control Channel) data when half rate TCHs are used.

GSM uses three main control channels, *broadcast channel* (BCH), the *common control channel* (CCCH), and the *dedicated control channel (DCCH)*. The purpose of these channels are described in detail in [42].

The complexity of GSM makes it prohibitively expensive for rural community use, and not easily deployable because of the way it is structured. Interviews were conducted with various industry sources from MTN, Vodacom, and Telkom, and all stated that GSM does not have the capacity to support a large dense populations like Kayelitsha. Therfore GSM is prohibitive for rural application because of its high operation costs, and not appicable to township communities because it does not have sufficient capacity.

4.3.3 GSM Application in Developing Township Communities

As part of Vodacoms licensing agreement in South Africa, it had to agree to provide telephone services in developing communities. Using equipment supplied by Alcatel Telecom's local subsidiary Vodacom has franchised members of various communities to become *phone shop* operators. Calls made from initial phone shops where subsidised by Vodacom and have been paid for on a simple cash-per-call basis. The service has become part of township life, giving millions of people affordable access to telephones for the first time in their lives. The newer phone shops adopted a prepaid card system, which works on a cash-per-card basis.

At the beginning of 1998, 14000 phones have been installed in phone shops around the country, according to the marketing department of Vodacom. The phone shops are housed in industrial shipping containers sponsored by the Pick 'n Pay supermarket chain. Many of these phone shops are being developed into telecenters, and already they cater for fax and telephone usage.

This approach to providing telephone sevice is called Universal Access. Telkom have taken the approach of supplying Universal Service, explained in section 4.4. As of May 1997, Vodacom has franchised over 200 operators countrywide. More than a million calls are made from 350 outlets [9]. As a result of the high success of these phone shops thriving businesses have developed, thereby stimulating commerce in these communities. The result of the phone shop industry is that 20 million people now have *universal access* to telephones. Although they may not have access in there homes, they do have access in relatively easy reach areas.

As can be seen from the GSM network description and explaination, GSM does not provide support for deploying the network or base stations quickly, due to its complexity. Often governmental approval can take weeks or even months in some cases. Furthermore, some base stations use large antennas of 20 to 30 meters height and are in public areas were they often draw critisism for spoiling views. Many antennae have already been disguised to look like Palm trees, Cypress trees, or made to visually blend into the mountain. This added dimension to the problem of supplying cellular service further slows network deployment. Ideally small antennae would solve the problem (Chapter 5).

The other side of the story is that once the cellular network provider has installed a base station the users have to buy mobile phones. In the past a two year phone contract had to be acquired. In order to be successful in getting a contract the applicant needed a good credit history. This resulted in many applications being rejected because most of the South African population has not got any credit history at all. Approximately 40 % of all Vodacoms cellphone contract applications were rejected in 1997. 95 % of the rejected applicants bought prepaid cards thereby allowing Vodacom to accept most cellphone applicants.

Vodacom decided to use the prepaid approach to solve this problem. In this scenario, the marketing approach had to be taken into account when designing the billing system, to deal with a non-pay culture. Vodacom engineers developed the a prepaid billing system and Siemens assisted in the development of an electronic card system which was integrated with a cellular phone. This technology was then used in the phone shops. It enables customers without a credit history to prepay for a phone card. The cellular provider makes several cards available to phone shop operators who then retail the cards to the public. The advantage of the prepaid system is that service is sold on a cash basis and thus there are no problems of unpayed bills. Vodacom takes 50 % of all prepaid card sales as remuneration for their service, the rest goes to the phone shop operator.

The phone shop concept and Vodacoms prepaid Vodago service has expanded cellular sevices from a niche market into a mass market. It creates revenue for the service provider and also develops brand awareness. Vodacom prepaid phone cards are now marketed at petrol garages and retail stores. One of the advantages of prepaid cards is that they enable the subscriber to try the service before they buy the full service.

4.4 Digital European Cordless Telephone (DECT)

As much a GSM has been a success it has its limitations when it comes to supporting higher density environments for example, a township or high rise apartment block communities [24]. DECT has been choosen by Telkom to deliver quick rollout of telephone network infrastructure to meet the governments 2003 rollout plan.

DECT provides high speed short range communications for a broad range of applications and environments. It offers very good quality voice service and data application services[22]. DECT is designed around the open standard (OSI) model which makes it possible to interconnect wide area fixed or mobile networks, such as ISDN and GSM, to a portable and/or fixed subscriber population. DECT provides low power radio access between cordless subscriber unit and fixed base station over a range of up to a few hundred meters. A DECT base station also uses dynamic channel allocation based on the received signal from the portable subscriber unit and is designed to support handoffs at pedestrian speeds.

DECT is standardised, therefore any mobile terminal of any manufacture make will work with any DECT base station due to its use of the Generic Access Protocol (GAP).

Its frequency hopping multiple access ability removes the need to perform frequency planning. This contributes to the simplicity of the system and enables quick rollout. These are some of the reasons that Telkom use DECT in a township community.

4.4.1 Architecture of the DECT system

The DECT architecture is based upon the OSI (Open System Interconnect) network model principles similar to ISDN. The network can be split into four layers: the physical layer, the medium access control (MAC) layer, the data link control (DLC) layer, and the network layer. A control plane (C-plane) and a user plane (U-plane) use the services provided by the physical and MAC layers.

The Physical Layer

DECT achieves multiple access using FDMA/TDMA with TDD. The assigned DECT frequency range is divided into ten narrow frequency bands. Within a TDMA time slot a dynamic selection of one of the carrier frequencies, corresponding to the narrow frequency band, is used. The channel bandwidth is 1.728 MHz and supports a 1152 kbps data rate. The DECT frame structure consists of 24 time slots per frame, twelve time slots are used for forward link communication and twelve time slots are used for reverse link communications. The DECT frame has a 10 ms duration. The time slot structure consists of 480 bits of which 32 bits are assigned for synchronisation, 288 for data bits, and 60 bits for the guard time. The DECT TDMA time slot structure and frame structure are shown in figure 4.4.

Medium Access Control (MAC)

The MAC layer consists of a paging channel and a control channel for the transfer of signaling information to the control plane. The normal bit rate of the user information channel is 32 kbps. DECT also supports a 64 kbps data rate and multiples of 32 kbps data rates to support ISDN and LAN-type applications. The MAC layer supports the handoff of calls.

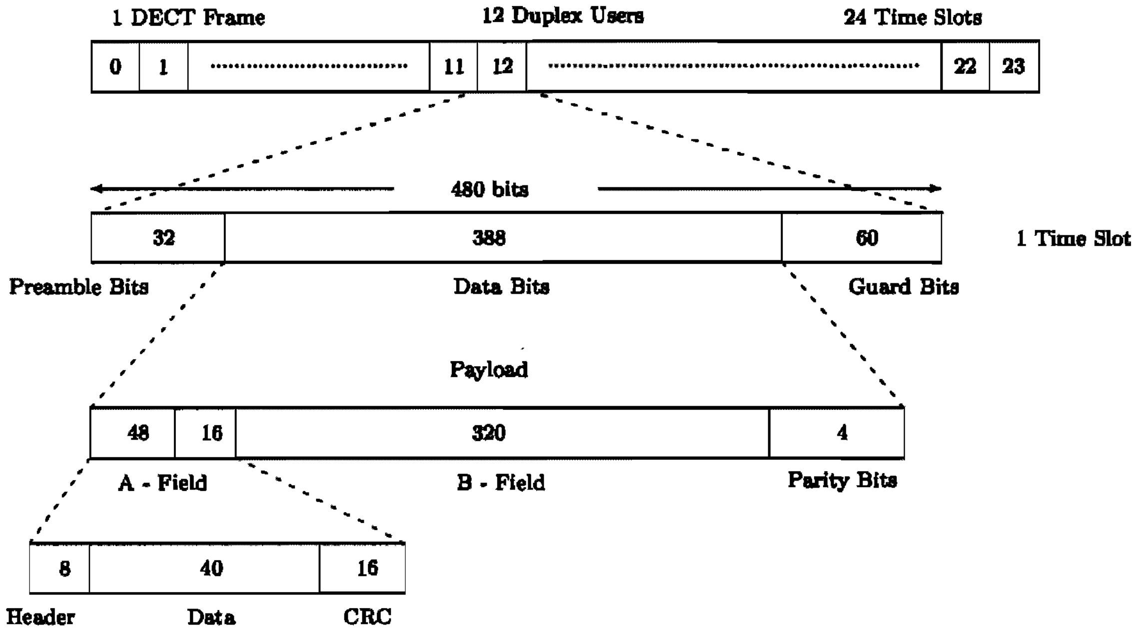


Figure 4.4: DECT TDMA Frame Structure

Data Link Control (DLC) Layer

The DLC layer divides the logical and physical channels into time slots for each user. It is responsible for providing reliable data links to the network layer.

Network Layer

The network layer is DECTs main signaling layer and is based on ISDN and GSM protocols. It provides call control and circuit-switch services that are selected from the DLCs services. It also provides a connection-oriented message services and mobility management.

4.4.2 The DECT Functional Concept

The DECT system is a microcellular cordless telephone that can be integrated with the PSTN or PABX (public automatic branch exchange). Figure 4.5 shows the five functional entites that make up the DECT systems.

The portable handset (PH) is a mobile radio station or can be a fixed terminal. Cordless adapters (CTAs) may be used to provide fax or video

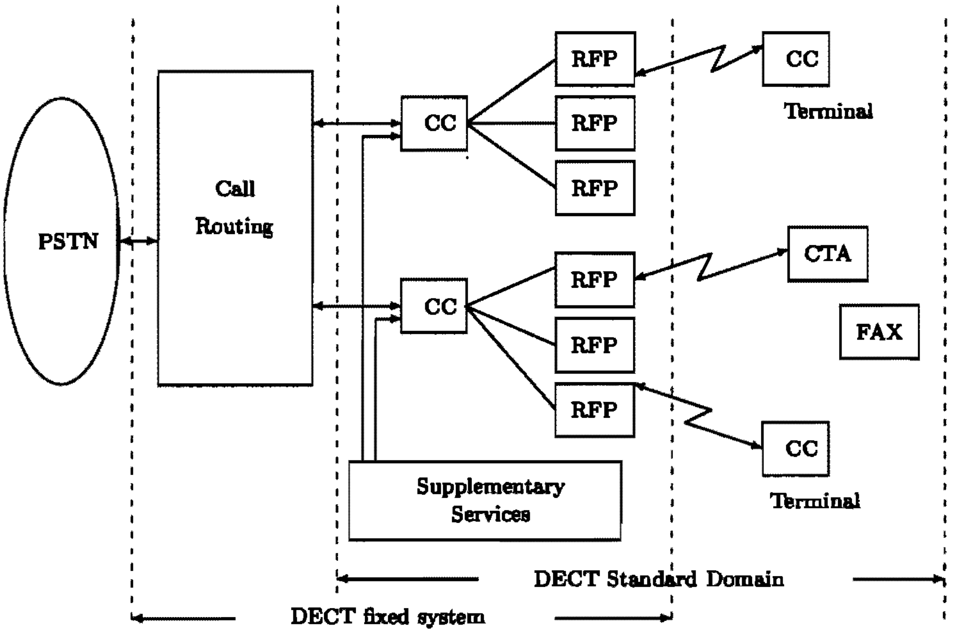


Figure 4.5: DECT Functional Concept

communications.

The radio fixed part (RFP) supports the physical layer of the DECT common air interface. Each RFP services one cell in the microcellular system. The radio transmission between the RFP and the portable units uses multi carrier TDMA. TDD is used to achieve full duplex operation.

The cordless controller (CC) handles the MAC, DLC and the network layer's one or more RFPs. It therefore forms the central control unit for DECT equipment. Speech coding is done in the CC using 32 kbps ADPCM.

The supplementary services provides centralised services and mobility management when DECT is used in a multilocation PABX network.

4.4.3 DECT Radio Link

DECT base stations support FHMA over the TDMA/TDD structure. If the frequency hopping option is disabled for each DECT base station, a total of 120 channels are provided by the DECT system before reuse is required.

DECT user data is provided in each B-field time slot. 320 user bits are provided during each time slot yeilding a 32 kbps data stream per user. No error correction is provided although 4 parity bits are used for error detection.

DECT carries it control information in 64 bits of every time slot of a call in progress. These bits are assigned to one of the four logical channels depending on the nature of the control information. Therefore the effective channel data rate per user is 6.4 kbps. DECT relies on error detection and retransmission for accurate delivery of control information. Each control word contains 16 CRC bits for error detection, in addition to the 48 control data bits. The maximum information throughput of the DECT control channel is 4.8 kbps.

The DECT system uses an 8 kHz sampling rate PCM to digitize voice. The speech samples are ADPCM encoded at 32 kbps and produce wireline quality voice signals. Therefore DECT has better voice quality than the GSM system.

A summary of the DECT radio specifications is in Appendix A.

4.4.4 DECT Application in the Kayelitsha Township Community

Telkom installed an Alcatel 9500 DECT system for wireless telephonic communication in Kayelitsha, Cape Town during the first half of 1998. This is in keeping to Telkoms policy for providing *universal service*. Universal service means that each and every user requiring a phone is given access, say in their house, as in the Site-C community of Kayelitsha.

Various problems have arisen for the service provider (Telkom). These problems relate to payment, installment, and capacity.

Telkom placed a 35 meter mast approximately in the middle of Kayelitsha. The mast uses directed antennae to provide antenna diversity. The antenna station is sectored into 12 sectors. Each sector can accomodate 120 local loop connections at any one time. Therefore the Kayelitsha DECT system installation can support 1440 local loop connections at any one time. The base station complex is connected to the Michels Plain telephone exchange via a 2X2 Mbps G.703 trunk link that is divided into 60 channels. A microwave point-to-point backup link is also provided in the event that the 2X2 Mbps link fails.

Most problems encountered in the network related to the radio terminal at the subscriber's (customer) site with the radio terminal. Telkom provided fixed access terminal stations, as is shown in figure 4.6.

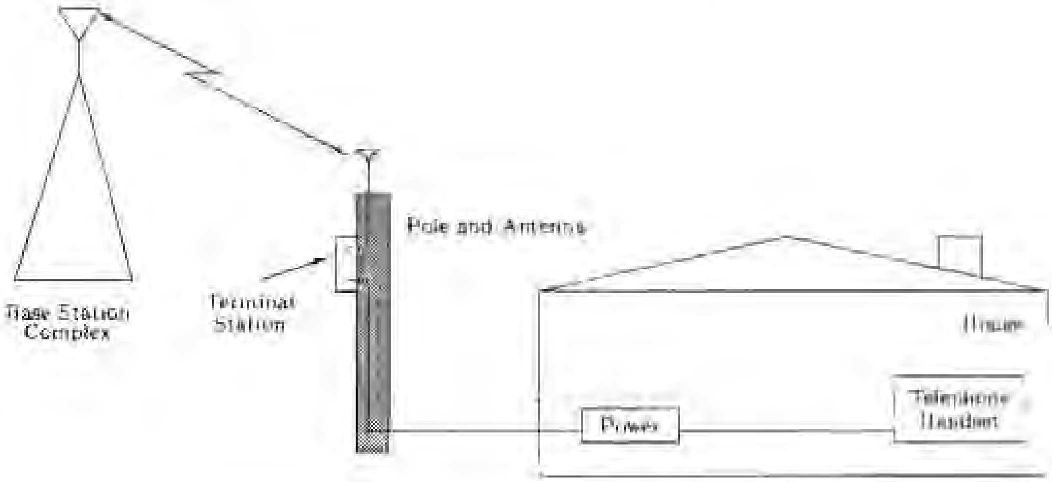


Figure 4.6: DECT house terminal station for fixed access communication with base station

Electrical Power

Due to the Kayelitsha community being underdeveloped, there is a shortage of electrical power outlets, eventhough Kayelitsha has been recently (1997) electrified not every household has got access to electricity. In the situation were no electricity supply is available the subscriber is provided with a solar powered terminal station. They effectively got free electrical power to operate their telephone. On the other hand, customers who have electrical power have to supply their own telephone electricity, and pay for it.

Telkom charges all DECT customers the same tariff and do not differential between subscribers with solar power or mains AC power access. The Kayelitsha community has made Telkom aware of this problem. Telkom are currently deciding whether to develop a subsidy system to make their billing scheme fair. Telkom are requesting that the telecommunications white paper be [illegible] so that power to wireless telephones will be required to be paid by the subscriber. Telkom say that a phone should be thought of as any other household appliance. According Telkom, the cost to supply electrical power to the DECT subscriber terminal station system should not cost more than two rand a month.

Telkom provide an AC/DC (230V AC to 16V DC) converter unit powered by the customers electricty supply via a 15 Amp plug connection. The unit also provides a backup battery having the capacity for 10 hours operation under average power drain conditions to cater for power grid failure [5]. In the case of an informal premises with a ready board (prepayed electricity supply system) the AC/DC converter unit is connected to the systems via a 15 Amp plug connection.

In the case of an isolated premises with solar power the system has a capacity for three days. The provision of backup batteries and AC/DC converters make the system more expensive, therefore the cost of a terminal station depends on the type of premises. This makes costing and tariff planning more complex for the universal service provider.

CTA Terminal Placement and Mounting

Telkom refer to a CTA as a Customer Terminal Apparatus which indicates all the equipment making up the radio terminal system of a customer (subscriber): power converter, radio, fixed handset, wiring, etc.

With formal housing an AC/DC converter and battery backup unit is placed near the electricity supply, usually on a wall. However, informal housing presents a problem. Informal houses do not always posses a sturdy and robust structure to place the terminal station equipment. This often requires a suitable structure to be built. Appendix B presents the diagrammatical plans used by Telkom to install the equipment, and serve to show the complexity involved in installing the radio terminal and auxilliary equipment.

Telkom specify that no CTAs are to be installed indoors, only the AC/DC power supply unit [7]. Therefore cabling is required from inside the house to power the CTA outside the house. A possible reason for this is to reduce antennae cabling costs and interference effects between the CTA and antenna. A radio and antenna may be mounted on a pole outside the premises, thus requiring carefull installation of the power supply cable to the pole. On some premises power and telephone cabling is layed underground between the pole and the house for safety. The pole also requires concrete bases to be cast, and lightening protection. These considerations contribute to the difficulty, expense and deployment time of the wireless system.

The lack of adequate support structures for mounting radio equipment has necessitated Telkom to devised solutions to these problems. The result is that providing universal service is not as simple as setting-up and installing a base station and supplying terminal stations. Due to the underdeveloped state of most South African township and rural community the solution to providing universal service is not easy, and requires special planning and design.

In a developed community it is easier and quicker to setup and install a wireless network because these issues do not usually arise. An example is the new wireless telephone system being installed by Telkom in Camps Bay (Cape Town).

Terminal pricing

According Alcatel Altec in Johannesburg, the wireless network terminal (WNT) used in the subscriber terminal station costs $US 252. An approximate cost per subscriber terminal station system was quoted at $US 2000. This cost does not take into account the cost of labour, pole mounting or cabling. We see that due to the problems of supplying informal premises the cost of the service is not only limited to the network cost, but must also take into account the high installation and labour cost. A true cost per line could not be acquired because Telkom would not release this information.

Traffic

The DECT Alcatel 9500 system is currently installed to provide a voice service. According to Telkom in Cape Town, Internet traffic has not been enabled on the Site-C DECT system because of insufficient capacity that long Internet call require.

DECT Site Service Market

The population of Kayelitsha as of 1 January 1997 was about 325 000 people, according to the Lelingelu Administration Board. It is reasonable to estimate that about 100 000 people of these 350 000 people may require a telephone service. The DECT installation has been designed to service 55000 people in the Cite-C area. Additional DECT infrastructure is required to service the large population of Kayelitsha. If Internet services are to be supported then additional Internet traffic carrying systems will need to be considered in order to handle the long call durations of Internet related local loop connections. This adds expense to the network, increase deployment time for rolling out service, and makes the local network more complex, resulting in additional maintenance.

4.4.5 CT2 and TDMA Telkom installations

Telkom have installed CT2 systems in various rural communities, but have stopped buying CT2 equipment because of the high cost per connection and low system capacity.

Telkom have recently bought an Alcatel 9800 TDMA system. The Alcatel 9800 system is a point-to-multipoint system which will be used to cover the Protem and Caledon areas inland of the Cape Province. This is a new project by Telkom.

4.5 Satellite: An Alternate Technology for Rural Communities

In the near future various satellite networks will be placed in orbit around the Earth, and soon thereafter become operational. Three types of satellite orbital configurations exist:

- LEO - low earth orbit
- MEO - medium earth orbit
- GEO - geostationary earth orbit

Geostationary satellites have been in use for over 35 years. They are situated at an altitude of about 36800 km above the equator, which allows them to maintain a fixed position in relation to the earth's surface. A round trip communication through a GEO satellite takes about half a second.

Modern communication systems are being designed to use satellites that move around the earth. Such systems use LEO and MEO satellites. LEO and MEO satellites offer global coverage which is an advantage that such communication systems have the terrestrial personal communication service (PCS) networks and cellular systems. The low altitudes of LEO and MEO satellites result in lower communication latency periods than that of GEO satellites.

The focus of access provisions via MEO and GEO satellite communication systems is to provide remote areas or areas of low population density. The cost per satellite circuit is almost independent of geographic location or demand density. South Africa will have large amount of bandwidth made available throughout the country from 1998 to 2003. Terrestrial operators will align themselves to satellite system operators as the market consolidates, such as has occured with Telkom and ICO.

Rural areas are traditionally the low teledensity areas in any country because of the expense of supplying terrestrial communication systems. As a result rural communities are a mostly untapped market which satellite communication companies stand to service best if their prices are competitive.

Table 4.1 and table 4.2 show a time table for commercial launch of satellite networks and their applications [12].

Table 4.1: The LEO systems compared to their terrestrial counterparts

Satellite Service	Application	Terrestrial Counterpart
Little LEOs (e.g. Orbcomm)	Paging	Paging network
Big LEOs and MEO (e.g. Iridium, Globalstar, ICO)	Cellular	Cellular networks and basic copper and fixed wireless networks
Broadband LEOs (e.g. Teledesic)	High-speed data communications	Advanced copper and fibre networks

Table 4.2: Approximate dates of commercialisation for various satellite systems

Satellite System	No. of Satellites	Approximate Date of Commercial Launch
Orbcomm	36	1998
Iridium	66	1998
Globalstar	48	1999
ICO	10	1999
Ellipso		2000
Teledesic	288	2003

The result of communication rollout on such a large scale will definately reshuffle the telecommunications market. These satellites will bring about new services and technological market places. They will also be beneficial for tele-education and tele-medicine services.

Chapter 5

Packet Radio and Multihop Networks

Various types of networks exist, fixed or mobile, wireline or wireless, which offer various services, such as enabling a telephone conversation or transfering data through a computer network. They are constructed from various electronic equipment and materials. Some networks perform the same task but achieve it with different technology. For example, a wireline telephone network and a wireless telephone network. Packet radio networks is the emphasis of this chapter.

In general, todays wireless networks offer one of two functions (or delivery services). The first type of network is a voice network which delivers a voice service at acceptable speech quality. These networks are delay sensitive. In general, they use circuit switching technology which is often used in telephony. A circuit is set up between the source and destination by the use of special signalling messages before any transmission begins. When the circuit is established, direct voice data transmission from source to destination takes place without any substantial intervention by the network.

Voice may be digitized and transported in the form of binary packets via the digital network to the destination handset. In doing so, the packets pass via various network elements, such as routers and gateways, which steer the packet to its specified end destination (sometimes called a sink). Any long delay periods in delivering a packet will result in the packet being dropped. This prevents unnatural pauses during conversation and ensures a continuous conversation, but also degrades voice quality. This is the approach used in TDMA voice communication systems.

The other common network function is to deliver data. These networks are often called computer networks and are dependent on data integrity.

Communication is achieved using relatively high-speed communications channels (in our case, wireless channels), and switching nodes.

In a multihop packet radio network the switching nodes consist of a transciever and a with processing and storage capabilities. In packet switching, the message (digital program code or raw data) to be transmitted between source and destination is transported in the form of a collection of data packets. Each packet contains a part of the program code or raw data.

The message is divided into blocks of data which do not exceed a certain maximum size. Each block of data is has a header containing source and destination address as well as control information. A checksum which is used for error detection and/or correction purposes is usually appended at the of the data. This constitutes a packet, sometimes called a datagram. A block diagram representation of a packet is shown in figure 5.1.

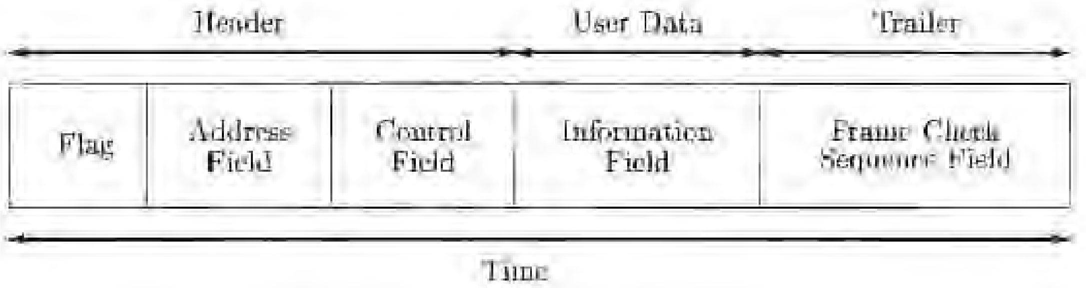


Figure 5.1: Format and fields of a typical data packet

If any packet received at the end terminal is corrupted or missing data and is beyond correction from error coding information contained in the packet then the program code will not be properly executable. This can result in system breakdown and/or unrecoverable information. In general, unlike voice services, a computer network can operate with delays, even though service speed may decrease, but cannot tolerate data corruption or packet loss. Any delays in computer networks should be minimised in order to increase channel capacity.

Todays modern mobile wireless networks also rely on control data to perform various network functions such as handoff and billing to ensure network operation and therefore are not entirely voice specific.

In the past and still today, packet switching is favoured over circuit switching for computer networks because of the bursty nature of computer traffic. Burstiness results from a message generation process and message size distribution which tend to be highly random, and users and devices requiring the communications resource relatively infrequently or seemingly randomly. For bursty traffic, packet switching using store-and-forward transmission techniques are cost effective since a packet occupies a particular link only for the duration of its transmission on the link.

New telecommunication and computer network technologies are bringing about a global trend toward technology convergence. There is increasing market demand for the two types of network services described above (voice and data) to be provided by one network. For example, we are seeing short message service (SMS), e-mail and fax capabilities built into cellular handsets, and audio and video on demand via ATM high speed computer networks. Most likely future networks incorporating voice and data service delivery functions will be hybrids comprised of different network technologies using packet switching and circuit switching at different hierarchical levels.

This chapter presents an introduction to some basic and important aspects of *multihop packet radio networks*. In particular, a discussion is presented on topics such as network architecture, packet radio multiple access schemes, and throughput results from various papers. In Chapter 6 we will use the concepts presented here to discuss the application of multihop packet radio networks in the context of telecommunication services in developing rural and urban township communities, and look at some ad-hoc multihop network technologies used in practice [2] [18].

The design of wireless multihop packet radio networks involves many issues that interact in a complex fashion. Many of these issues relate to the use of the RF channel for multiple user access. To date, no single model has been published that incorporates all the necessary parameters and provides an optimal solution that enables accurate network traffic modelling. This is evident in the continual development of packet radio models and multiple access protocols. Contrary to point-to-point networks where each channel is used by a single node pair, the radio *channel* in packet radio networks is a multi-access broadcast resource. In any area of local radio connectivity, the channel is shared by multiple contending users. Hence the need for muliple access protocols to manage the resource. In addition, radio is a broadcast medium and therefore the action of any node effects the actions taken by neighbouring nodes and their outcome.

5.1 Packet Radio Network Description

Packet radio networks represent the extension of packet switching technology into the radio broadcast environment. They provide data communication to users located over a broad geographic area, where direct radio or wire connection between the source and destination users is not practical. In general, data in binary form (digital information) is transferred. This allows for the transmission of computer data or digitized voice.

Packet radio networks operate on the basis that packet switching is applied to broadcast radio sharing a single channel. In general packet radio usually operates in a store-and-forward mode. This is explained with the aid of figure 5.2.

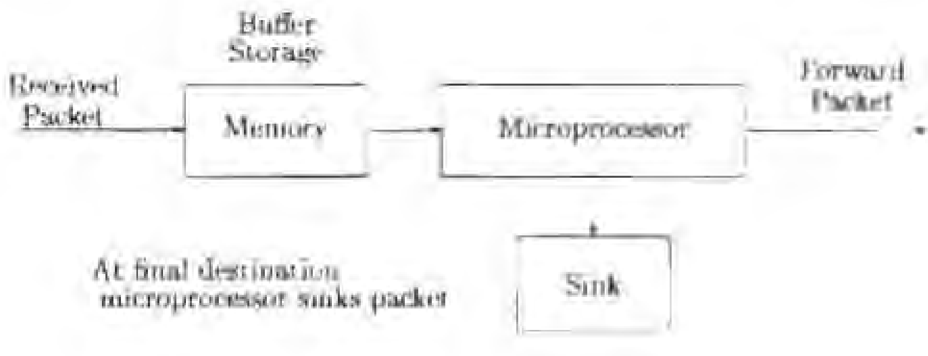


Figure 5.2. Store-and-forward operation

A typical packet radio unit (PRU) consists of a radio, antenna, and digital controller. Packet radio units will also be referred to as network nodes. In multihop networks the radio provides connectivity to neighbouring nodes, and usually cannot establish a direct radio link with all the nodes in the network.

Store-and-forward operation is provided by the controller to enable the relaying of packets to accomplish connectivity between source and destination nodes. The digital controller receives the packet, *stores* the packet in memory, analyses the packet for control and routing information, and then (if necessary) *forwards* the packet by sending it to the transmitter for retransmission. The signal should then be received by a neighbouring node on the route to the end destination node. Figure 5.3 shows typical delivery paths as packets are forwarded through a multihop ad-hoc packet radio network

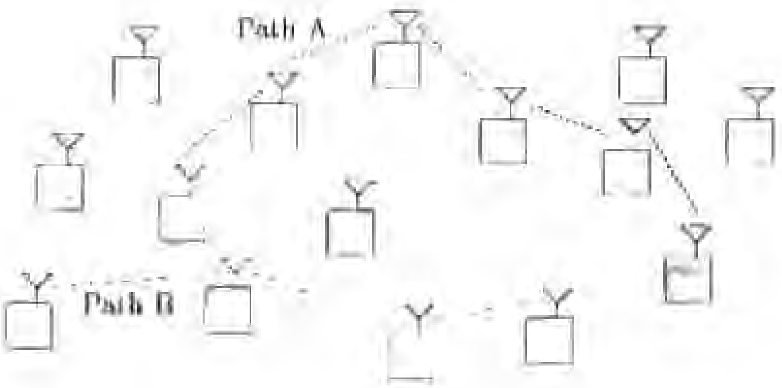


Figure 5.3: Multihop PR network data routing and delivery: path A and path B deliver packets to different nodes

A packet may travel via many nodes before it arrives at the end destination node. The source node will transmit a packet on to the radio channel to a neighbouring node. The receiving node will determine if the packet is meant for itself, else will decide which is the next appropriate node that could forward the data to the destination, and then retransmit the data to that node. If the source and destination nodes are more that one hop away, then the packet will hop between multiple nodes, hence the name *multihop packet radio network*.

In a packet radio network we want to move data as fast and as accurately as possible. In order to accomplish these two objectives various issues relating to the design of a multihop packet radio network must be understood. The most common issues are multiple access protocol, capture effects, transmission power, throughput, delay characteristics, and topology. A discussion of these subjects follow.

5.2 Channel Access

A packet radio network requires a set of protocols to operate at various network levels: from the lowest level where a packet makes a hop, to higher levels that ensure reliable data transfer and routing, as in any packet communication network. In this section we look at narrowband and wideband *channels access protocols*.

A channel access protocol defines the conditions under which a node may access a channel. For many protocols, these conditions are expressed in terms of the state of the node in question, and the state of neighbouring nodes in the network. The rules embodied in the access protocol are constrained to be defined in terms of information that can be acquired locally at the node, such as the state of the receiver at the node, and the state of the transmitters in the radio locality. The success of a protocol depends on the ability of a node to dynamically acquire knowledge about the state of the other nodes, which depends on the particular channel signalling scheme and code-assignment scheme used. The capture properties affects the choice of access protocol, as we will see later in this chapter when the capture effect is explained.

5.2.1 ALOHA: History and Description

A single hop packet radio network called the ALOHA system was pioneered at the University of Hawaii in 1970, and was developed by Abrahamson and Kuo. It was the first communication systems to use *packet broadcasting* and *random access* in sharing a multiaccess channel. The system used two radio channels: a multiaccess channel shared by the terminals (nodes) to reach the central computer/radio system, and a broadcast channel which the central system used to respond to the terminals. A *random access* scheme called ALOHA was used to share the multiaccess channel resource between the terminals. ALOHA was designed to offer service to bursty long-haul traffic. That is a message was not sent as a collection of packets, but as an entire data block, or as one packet.

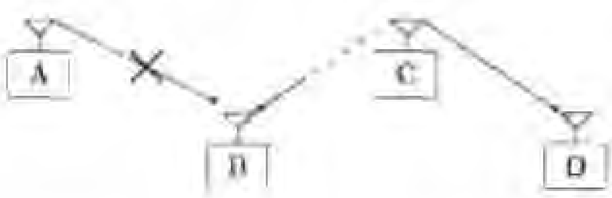


Figure 5.4: Transmission C to D collides with transmission A to B: transmission C to D succeeds, transmission A to B fails

ALOHA (also referred to as *pure* ALOHA) permits a node to transmit packets any time it desires. If part of a user's transmission overlaps with another user's transmission then the two transmissions are considered to have collided (figure 5.7) and have corrupted (sometimes

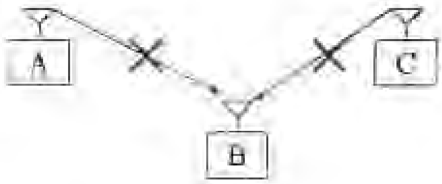


Figure 5.5: Transmission A to B collides with transmission C to B: both transmissions fail

Figure 5.6: Transmission A to B collides with transmission B to C: transmission B to C succeeds, transmission A to B fails

completely destroyed beyond correction) each others transmissions. If one of the transmissions have a significantly larger power lever than the other signal then, in general, the stronger signal will be correctly received. This effect is called *power capture*. Figures 5.4, 5.5 and 5.6 show three possible situations that cause a collision. If all the signal power levels of transmissions at all the receivers in a network are equal then we say that we have a zero capture network because there is no power capture effect. In the case of a zero power capture network a collision will always occur in the three situations described by figures 5.4, 5.5 and 5.6.

An acknowledge signal from the central system (in the case of a single hop network) is used to indicate to the nodes whether a transmission has been received successfully. Some systems actually transmit a negative acknowledgement signal (NACK) to indicate a corrupt reception. A negative acknowledgement response received by a node after a timeout period would indicate an unsuccessful transmission, and the node will retransmit. However, retransmission is delayed by a random period of time (*backoff interval*) to avoid continuously conflicting transmissions by spreading the retransmissions of packet over time. The retransmission ability of a node requires the transmitting node to store (usually in a buffer) the transmitted packet. If an ACK signal or no NACK signal is received after a certain timeout period then the packet can be removed from the buffer. Typically the data transmitted is padded with error checking bits (like cyclic redundancy checks, CRC). If the CRC does not match the CRC calculated for the received data, then the reception is deemed to be erroneous and a negative acknowledge (NACK) is transmitted. By using ACK and NACK, a packet radio system employs perfect feedback, even though traffic delay due to collisions may be high. Theoretically two transmitting nodes can continue to retransmit at the same backoff interval, but this depends on the random number generator used to determine the backoff interval. Usually the initial generator seed is different for the different transmitters.

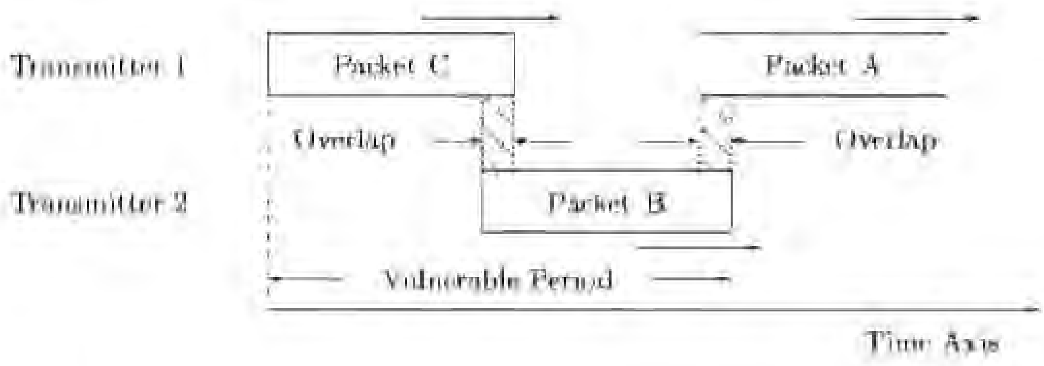


Figure 5.7: Packet collisions: packet B (transmitted by node 1) collides with packet A and packet C (transmitted by node 2)

Different channel efficiency measures have been reported in various papers. The differences are due to measures made by simulations using different network models and network assumptions, or different physical networks. The usual value reported for channel efficiency using ALOHA is about 18 percent, however [16] reports a maximum channel efficiency limited to about 18 percent, or .184 erlang [15]. This value is known as the capacity of an ALOHA channel, and it occurs for a channel traffic load of 0.5 erlang. Figure 5.8 shows channel throughput versus channel traffic for an ALOHA channel and a slotted ALOHA channel is presented further below.

If we have n users, it can be shown [15] that the normalised channel throughput S for a slotted ALOHA channel is

$$S = Ge^{-2G}$$

where G is the normalised channel traffic offered. G may be greater than 1, and S is less than or equal to 1 and $S \leq G$

As the traffic load is increased beyond 0.5 erlangs the throughput wil decrease. ALOHA exhibits low channel utilization, but it has been shown to be superior to fixed assignment schemes such as FDMA and TDMA when user traffic is bursty and low packet delay is essential. [16] [46]. ALOHA is an all end-user (one hop) traffic system. It has low overhead transmission and the system is simple, and is commonly used for ad-hoc networks.

The research at Hawaii University led to the developement of a multihop packet radio network called PRNET. PRNET was sponsored by DARPA, the Defense Advanced Research Projects Agency, in 1972. Many modern multihop packet radio network concepts and protocols were developed from PRNET.

Summary of ALOHA

ALOHA pre-dates TDMA systems, and is a technique for uncoordinated users to effectively compete for a common resource channel. It is assumed that all nodes have equal access to the channel.

- a node transmits whenever it has traffic to send/forward
- *collisions* may occur when several nodes transmit in the same time window, which is smaller or equal to the maximum transmission time for any packet
- the receiving node (central system in single hop ALOHA) transmits back an acknowledge (ACK) if the transmission was successful
- a collision requires retransmission, which the transmitter performs after a random delay

5.2.2 Slotted ALOHA

In slotted ALOHA, time is divided into equal time slots of length greater than the packet transmission duration. The time axis is considered *universal* for all nodes. The nodes each have synchronised clocks and transmit a message only at the beginning of a new time slot, thus resulting in a discrete distribution of packets. In ALOHA a node is allowed to transmit when ever data is available for transmission. In slotted ALOHA data has to be buffered until a slot is available. Since the slots are a fixed size, slotted ALOHA is best suited for fixed-sized packets which are the same size as the time slots, thereby maximising channel utilisation.

The slots prevent partial collisions, which may affect more than one packet. Collisions can still occur but they can destroy at most one packet at a time. Consider a 1 megabyte packet being transmitted using pure ALOHA. If the last 1 kilobyte of the transmission was corrupted through a collision then the whole packet would need to be retransmitted. In slotted ALOHA the 1 megabyte message would be spread over the slots in the form of small packets. The slot containing the packet with the corrupted last 1 kilobyte would have to be retransmitted, not the entire message. The result is less retransmission traffic and delay, and yields increased channel throughput. A plot comparing the throughput characteristics of pure ALOHA and slotted ALOHA is presented in figure 5.8 [26].

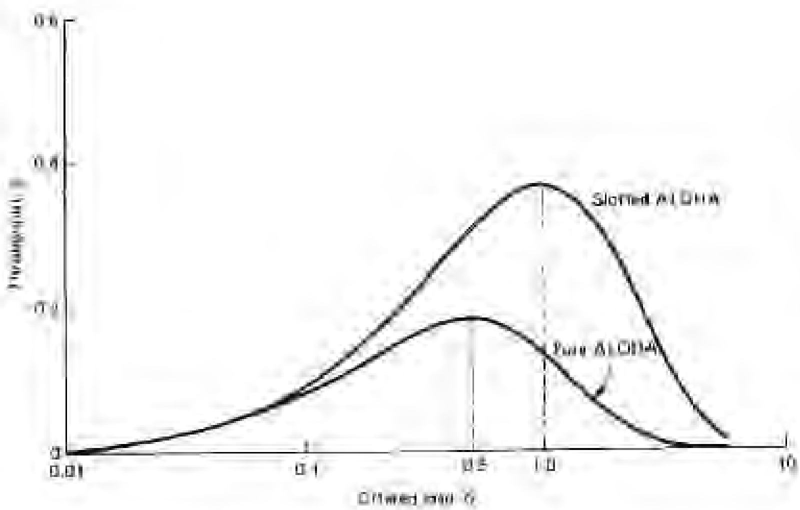


Figure 5.8: Pure ALOHA and slotted ALOHA throughput characteristics

As the number of nodes contending for the channel increases, the demand per slot increases, and delay will occur due to the complete collisions and the resulting repeat transmissions of those packets lost. If a collision occures then the packet must be retransmitted. The retransmitting node will wait a random number of slots before retransmitting. The delay characteristic is also determined by the number of slots a node must wait before retransmitting.

If we have n users, it can be shown that [15] the normalised channel throughput S for a slotted ALOHA channel is

$$S = Ge^{-G}$$

where G is the normalised channel traffic offered. G may be greater then 1, and S is less than or equal to 1 and $S \leq G$.

Slotted ALOHA throughput is 0.37 erlang as compared to .18 erlang for pure ALOHA. This is because the vulnerability time is now reduced to just the size of the time slot.

5.2.3 Carrier Sense Multiple Access (CSMA)

Carrier sense multiple access (CSMA) is another popular random-access protocol. In ALOHA no knowledge of the activity of the network is used, and nodes transmit independently of channel activity. In a carrier sense type protocol, some knowledge of transmission activity in the channel is acquired and used in the decision to transmit. Transmission is inhibited if a carrier is detected. Detecting a carrier does not only mean that a packet reception is in progress, but also that in-band RF energy is present.

In CSMA protocols, *detection delay* and *propagation delay* are important because they significantly affect protocol performance. Detection delay is the time it takes for a receiver to *sense* whether or not a carrier is present on the channel, and is generally determined by the nodes' hardware performance. A small detection time means that a node detects a free channel quite rapidly. Propagation delay is a relative measure of propagation time to packet duration. A small propagation delay means that a packet is transmitted through the channel in a short interval of time relative to the packet duration. It is still possible for a collision to take place. For example, just after node A transmits, node B may be ready to transmit. Due to a long propagation time, node A's signal has not reached node B, so node B senses that the channel is idle and transmits, resulting in a collision between the two packets. Therefore propagation delay affects the performance of CSMA protocols.

In single hop fully connected packet radio networks where all the nodes can hear each other node's transmission indiscriminately and the propagation delay is a small fraction of the transmission time of a packet (e.g. 1 percent) CSMA has a channel utilization of about 85 percent [16], significantly outperforming ALOHA. However, this is not always true and largely depends on the network topology and traffic.

In multihop packet radio networks, the *hidden node* (or *hidden terminal*) problem arises, and results in significantly reduced performance. Hidden nodes occur when a number nodes are not in range of all other nodes in the network (i.e. the network is not fully connected). Studies have shown that if a fraction $\alpha =$ % of the packet radio units making up the network are hidden from the rest of the network, all transmitting to the same destination within range of the entire packet radio population, then the channel utilization of non-persistent CSMA (described below) is about 35 percent. If α is 0.5 then channel utilization reaches a minimum of 28 percent [16].

As an example to describe the hidden node problem that is inherent in multihop packet radio networks (assuming an omnidirectional antenna is used at each node): suppose node A is transmitting a packet directed to node B, and that there exists a node C that is in transmission range of node B but not in transmission range of node A. In other words, nodes A and C are in range of node B but are out-of-range of each other. Therefore node C would not sense node A's transmission. If node C transmits a packet to any of its neighbouring nodes (one of which is node B) then it would do so without the knowledge of the presence of node A's transmission at node B. The result is that a collision occurs with node A's transmission at node B. If node C's packet was addressed to a node other than node B, say node D, then it is possible that a collision did not occur at node D.

The implementation of CSMA also requires special hardware, and the ability to switch rapidly from the receive mode to the transmit mode in order to keep the efficiency of the CSMA process high [17].

Various types of carrier sensing protocols exist for packet radio networks, and are described below.

The simplest form of a CSMA protocol is **1-persistent CSMA**. Here the transmitter listens to the channel until it finds the channel idle before transmitting. If no carrier is sensed on the channel, then it transmits its message with probability 1, hence the name 1-persistent. A problem that arrises with this strategy when two nodes, say A and B, are waiting for a channel to be freed by node C. Node A and node B continually monitor the channel and the moment the channel becomes idle they both transmit causing a collision. An improvement on this CSMA strategy is non-persistent CSMA.

An improvement of 1-persistent CSMA is **non-persistent CSMA** which operates as follows:

- the node transmits the packet if the channel is sensed idle.

- If the channel is sensed busy, then the node schedules packet transmission by delaying transmission according to a (re)transmission delay distribution. After the delay time is complete the node re-senses the channel and the algorithm is repeated.

This improves on the problem of two nodes transmitting simultaneously as soon as a third has stopped. Non-persistent CSMA is popular for wireless local area networks (LANs) where the packet transmission interval is greater than the propagation delay to the farthermost user.

p-Persistent CSMA is a generalisation on 1-persistent CSMA applied to slotted channels. The slot length is typically chosen to be the maximum propagation delay. In this protocol a node ready to transmit a packet will sense the channel. When the channel is found to be idle, the packet is transmitted in the first available slot with probability p, hence the name p-persistent, or in the next slot with probability $1 - p$.

CSMA/CD uses carrier sensing and collision detecting (CD). Carrier sensing has been described above. Carrier detecting involves a node listening (sensing) to the channel while it transmits, to 'hear' if any other transmissions take place at the same time, thus causing a collision. Therefore if two or more nodes start a transmission at the same time a collision is detected, and the transmission is aborted before completing transmission, thereby minimising the transmission of collided packets and improving bandwidth utilization. The key feature with this protocol is that a collision can be detected soon so that transmitters do not have to wait for an ACK or NACK. This requires the node to have a transmitter and receiver which support *listen-while-talk* operation. For a half-duplex channel this can be achieved by stopping transmission to listen. With full-duplex channels a full duplex transciever can be used to listen while transmitting [42].

5.2.4 Busy Tone Multiple Access (BTMA)

The *hidden node* problem can be alleviated by the use of a *busy tone* signal which is transmitted on a separate channel, say a *busy tone channel*, by a node to indicate that the node is receiving a packet and that the multiaccess channel is in use. The busy tone is used to inhibit the receiving nodes' neighbours from transmitting on the channel thereby causing interference with its reception. Any node within range of the receiving node with a packet ready for transmission, sensing the busy tone, will reschedule the transmission.

Several types of BTMA exist and they are applicable in both narrowband and wideband communication systems. In narrowband *conservative*

BTMA (C-BTMA), any node sensing a carrier, due to the presence of a transmission, will emit a busy tone. The effect is for a short period of time following the start of a transmission, all nodes within a two hop radius of the transmitting node get blocked. A certain period of time, primarily determined by propagation and detection delay, is required for setting up the busy tone signal at each node. This presents a vulnerable period where collisions may occur. If propagation and detection delay where zero then C-BTMA would be a collision free protocol.

In *destination-based* BTMA (D-BTMA) protocols the receiving node (destination) of an active link transmits a busy tone signal. In *receiving destination-based* BTMA (RD-BTMA), the receiving node transmits a busy tone only if it can synchronise on and start receiving the packet. For a narrowband systems this means that the transmission coverage area of the receiving node must be idle prior to arrival of a packet. Since a node will not know if the transmission is meant for itself until it reads the packet header, RD-BTMA is idealistic and is used for comparison purposes in [47].

In *Hybrid destination-based* BTMA (HD-BTMA) a node operates as a C-BTMA node until the header is processed, at which point it acts as a RD-BTMA node. In other words, all HD-BTMA nodes transmit a busy tone signal until the intended destination node is identifed. If a node is not the intended destination node then it removes its busy signal, and only the destination node is left transmitting a busy tone signal.

RD-BTMA is suited to spread spectrum packet radio implementation and in this context is referred to as *locked-on destination-based* BTMA (LD-BTMA). Using a receiver directed code assignment scheme (explained in a section 5.2.6), the destination node transmits a busy tone signal whenever the link is active and the destination node is locked onto the packet [47].

The use of a busy tone channel requires additional bandwidth and hardware resources for the communication system, and thus increases costs of the radio equipment in multihop networks.

In figure 5.9 [16] we see the throughput-delay tradeoff for ALOHA, CSMA and BTMA. The results were obtained from a simulation of six nodes in a ring topology. Normalized delay a is 0.01, infinite buffers, and uniform traffic at each node is assumed. [16] did not take into account the additional usage of bandwidth for the busy tone signal.

BTMA outperforms both ALOHA and CSMA. The six ring topology is not an ad-hoc topology. Different results would have been obtained for different network topologies. Figure 5.10 shows the link capacity of narrow-

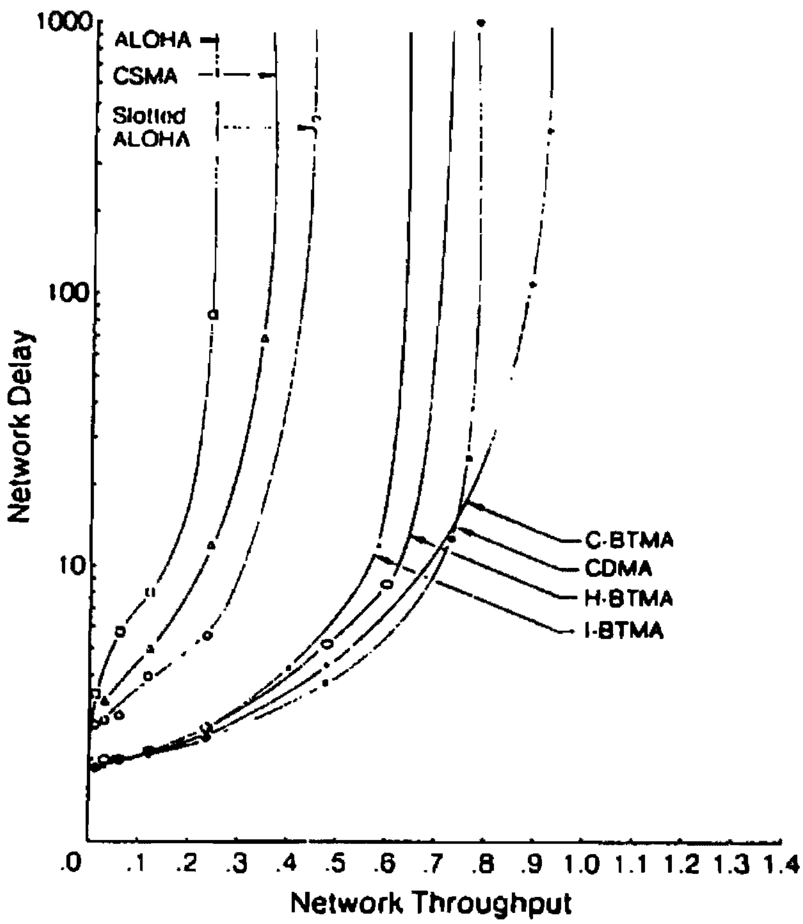


Figure 5.9: Throuhput-delay trade-offs for ALOHA, CSMA and BTMA

band systems with a ring topology using different channel access techniques. *Disciplined ALOHA* is taken into account in this figure. Disciplined ALOHA differs form pure ALOHA only in that it allows a node to transmit only if it is not already transmitting or not locked onto an incoming packet. It is suited to spread spectrum systems in which receivers lock onto packets.

5.2.5 Floor Acquisition Multiple Access with Non-persistent Carrier Sensing (FAMA-NCS)

The term *floor* is used to describe the channel in the vicinity of a receiving node. FAMA-NCS guarantees that a single transmitting node (or sender) is able to send data packets free of collisions to a given receiving node (or receiver) at any given time. FAMA-NCS is based on a three-way

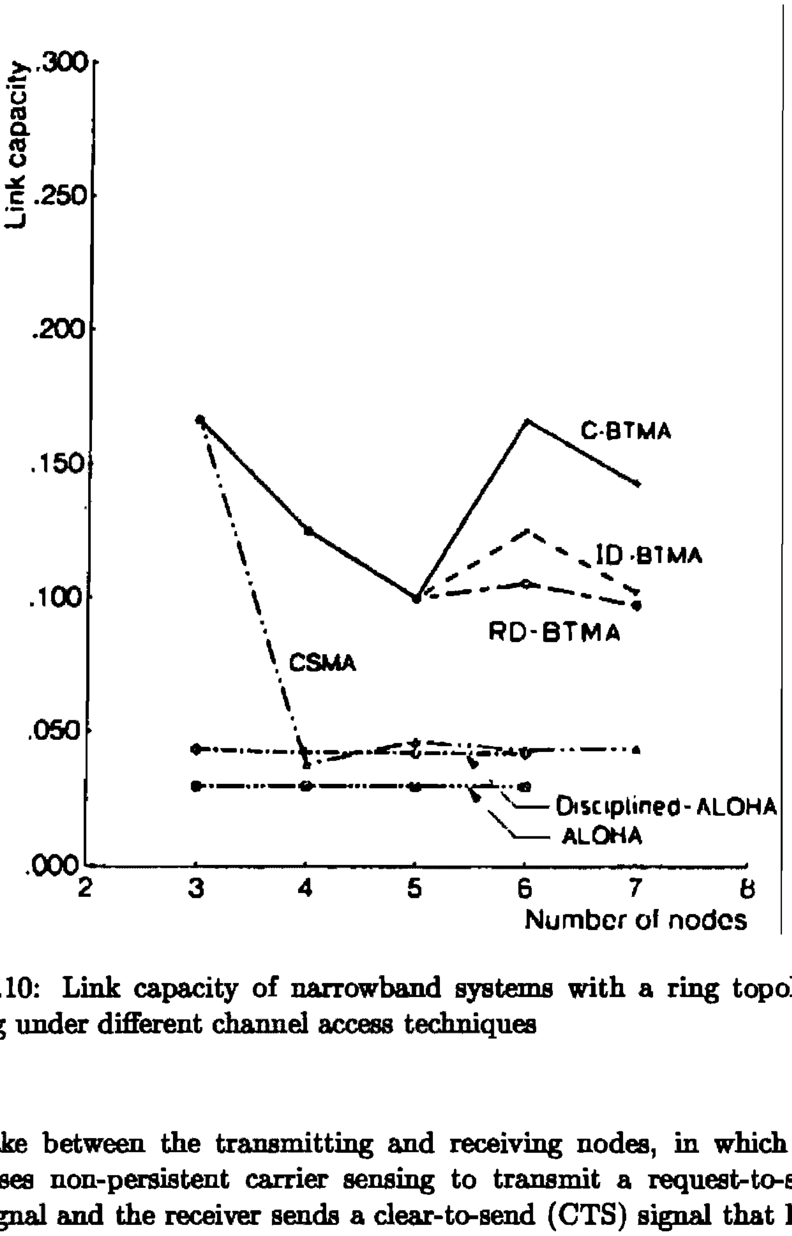


Figure 5.10: Link capacity of narrowband systems with a ring topology operating under different channel access techniques

hand-shake between the transmitting and receiving nodes, in which the sender uses non-persistent carrier sensing to transmit a request-to-send (RTS) signal and the receiver sends a clear-to-send (CTS) signal that lasts much longer than the RTS and serves as a *busy tone* that forces all hidden nodes to *back off* long enough to allow a collision free data packet to arrive at the receiver from the sender.

FAMA-NCS requires a node who wants to transmit one or more packets to acquire the floor before transmitting the packet train. The floor is acquired using an RTS-CTS exchange. To acquire the floor a node transmits an RTS using *carrier sensing* or *packet sensing*. Packet sensing corresponds using the ALOHA protocol for transmitting the RTS as a packet containing control information requesting the floor. CSMA is employed for sending the RTS using carrier sensing. A CTS is only returned once the requested node has received an error free RTS that is address to itself. The requesting node (sender) knows it has acquired the floor when it receives an error free CTS addressed to itself.

After acquisition the floor holder (sender) is able to transmit packets without collisions occuring on the channel. Any reliable link mechanism (discussed in section 5.3) can be implemented on top of FAMA-NCS between the floor holder and the nodes with whom it wants to communicate. Nodes that do not have the floor are forced to wait a minimum predefined period of time (a minimum of twice the maximum propagation delay) before being able to contend for the floor. When a node, ready to transmit data, detects that the floor has already been acquired then it reschedules its bid for the floor.

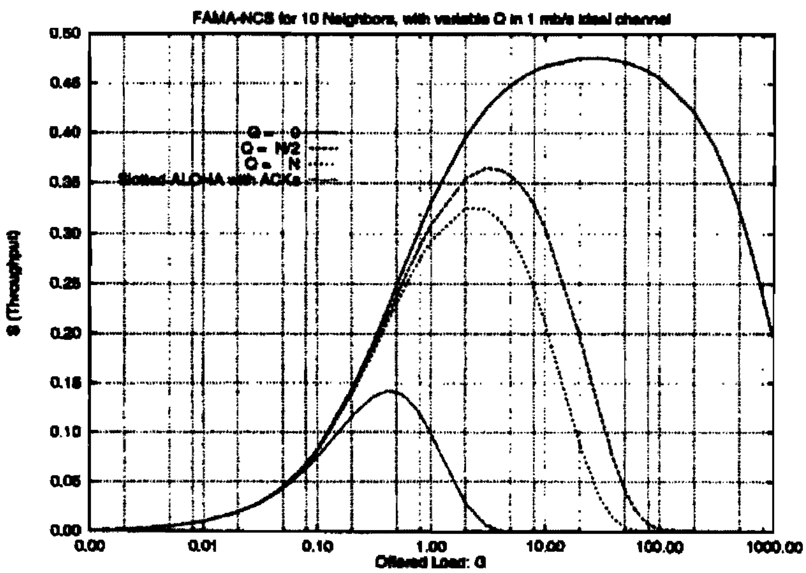


Figure 5.11: Comparison of results for slotted ALOHA and FAMA-NCS with various Q values

Figure 5.11 (from [29]) shows the comparison of results for slotted ALOHA and FAMA-NCS with various Q values (explained in this section). The results for slotted ALOHA throughput are lower in this plot compared to figure 5.8 because positive acknowledgements (ACK) have been taken into account.

[29] assumed a network where each node had 10 neighbours for varying

Q values. $Q = 0$ means the network is fully connected, $Q = N/2$ means that half the neighbours of any node are hidden from a neighbours' neighbour. $Q = N - 1$ means that all the nodes are hidden from their neighbours' neighbourhoods. [29] assumed that a network with a propagation delay of $6\mu s$ (one mile), 500-byte data packets, 25-byte RTS, and a 50-byte CTS, and presented results for a 1-Mb/s channel with zero preamble and processing overhead, and a 298-kb/s channel with preamble and processing overhead based on the Utilicom 2020 radio transceiver. Using the Utilicom model 2020 transceiver, results in the transmitter using $11ms$ to send an RTS and be ready to receive a CTS. If transmission overhead is included, then a 500-byte data packet becomes $25ms$ long.

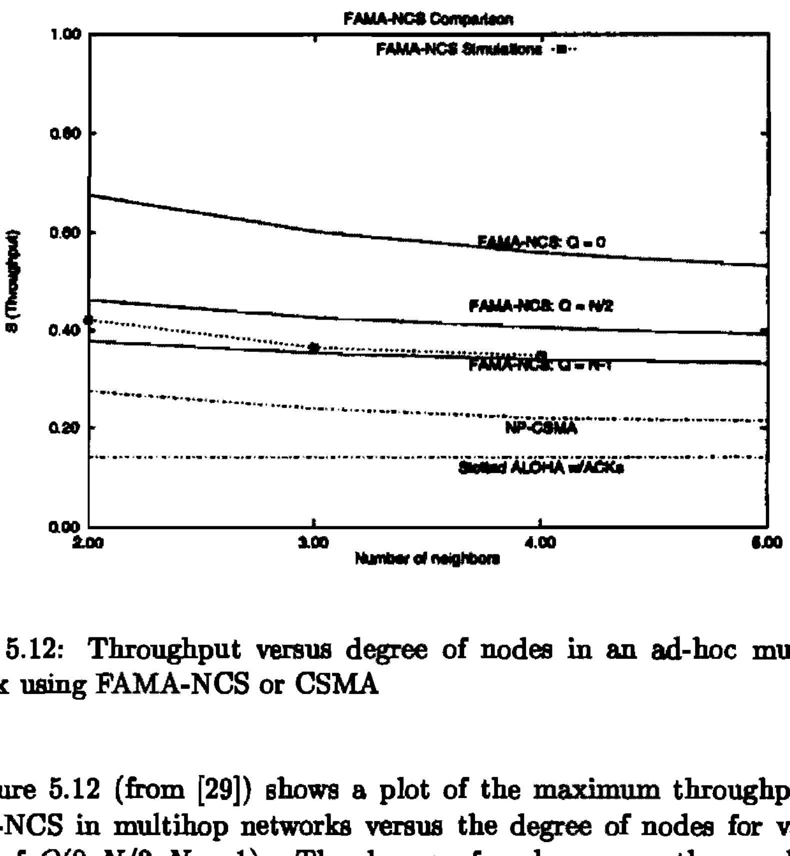


Figure 5.12: Throughput versus degree of nodes in an ad-hoc multihop network using FAMA-NCS or CSMA

Figure 5.12 (from [29]) shows a plot of the maximum throughput for FAMA-NCS in multihop networks versus the degree of nodes for various values of $Q(0, N/2, N - 1)$. The degree of nodes means the number of nodes in the vicinity of another node, or the number of neighbours per node. The analysis assumes a 1-Mb/s wireless network with about a $6\mu s$ propagation delay. A 500-byte data packet and a 25-byte RTS were used. The throughput for slotted ALOHA with acknowledgements is also shown in the plot. The performance of non-persistent CSMA in the plot is a value of the upper bound throughput of CSMA [23].

The results presented by [29] show that FAMA-NCS is a better choice than CSMA for an ad-hoc multihop packet radio network. The results obtained from the use of the Utilicom model 2020 transceiver indicate that radio parameters significantly affect the performance of FAMA-NCS in ad-hoc multihop packet radio networks. With good carrier sensing and short turn-around times (time to ramp up the transmitter or the time to ramp down the receiver), throughput can be substantially increased. [29] also claims that FAMA-NCS performs better than all prior proposals based on collision avoidance protocols, such as MACA [34], MACAW [48], and IEEE 802.11 DFWMAC [19] [10], in the presence of hidden terminals.

5.2.6 Spread Spectrum Techniques Applied to Packet Radio

Most of the above mentioned channel access schemes can be applied to wideband systems using spread spectrum techniques. In narrowband signalling, data is modulated directly onto the carrier. A collision results if two or more packets with similar power levels overlap in time at a receiver. If the first of the overlapping signals arriving at the receiver has a distinctly stronger power level than the others, then the *power capture* effect may result in the received packet being error free. Spread spectrum signaling schemes are based on modulating the message data with a spreading code, thus using a much wider bandwidth than a narrowband system.

Direct sequence pseudo-noise (PN) spread spectrum and *frequency hopped* spread spectrum are two commonly used wideband modulation techniques used. Frequency hopped spread spectrum has the added advantage of reducing the near-far problem at receivers [42]. Chapter 3 discusses spread spectrum modulation. Here we simplify the discussion by making no distiction between direct sequence PN modulation and frequency hopping.

The advantage of spread spectrum signalling over narrowband signalling is the reduction in interference between users in a multiuser environment and its improved performance in the presence of multipath fading and narrowband interference. In addition, spread spectrum eases network deployment and frequency planning, by allowing codes to be allocated to each node to uniquely identify each node. Codes can be used in a spacial sense. In other words, if there are more nodes than there are codes, then nodes allocated with the same code must not be in each others transmission range. This requires that the nodes have intelligent code assignment software to avoid erroneous code distribution.

Code-division and *time-capture* are the two main aspects that make the use of spread spectrum attractive in ad-hoc multihop wireless networks.

Code-division means that the transmissions are separated in code space, i.e. are orthogonal to each other, allowing different coded signals to overlap in time with little or no effect (ideally) on other transmissions. The time-capture [40] ability of a spread spectrum receiver with a specific identification code enables the receiver to receive a packet with the same code despite other time-overlapping transmissions with the same or different codes, i.e. the reciever can *lock on* to a signal.

Each spread spectrum packet radio transmission must be preceded with a coded preamble transmission. The purpose of the preamble is to synchronise the transmitting and receiving nodes. In *space-homogeneous* preamble code assignment the preamble code is known by all network nodes. Idle nodes constantly listen for this preamble code. The preamble code must have strong autocorrelation properties and be orthogonal to the codes used for encoding transmitted packets. A failure may occur when preambles collide thereby disrupting node synchronisation, or when the interference of ongoing transmissions, which act as background noise, reduce the effective signal-to-noise ratio at the receiving node.

Errors occuring due to preamble collision can be reduced by using *receiver-directed* preamble codes. Each receiver is assigned a distinct preamble code. Any transmitter that wants to send a packet to a node must use the receiving nodes preamble code as the preamble. Therefore the receiving node will only process packets which it identifies with its preamble code. All other transmissions appear as background noise. Once the preamble of the incoming packet is successfully processed, the receiving node is locked on to the packet until transmission ends. Once the receiving node is locked on to a packet it is important that the transmission is not disrupted by overlapping packets. Interference may occur depending on the encoding method used on the data packet part of the transmission. In addition, multipath effects and Rician fading as well as insufficient signal power level may result in erroneous reception.

Several methods of selecting and using the PN sequence can affect the success of transmissions. Consider *space-and-bit-homogeneous code assignment* where a fixed PN sequence identical for each transmitter throughout the network is used [17]. The only time overlapping transmissions will collide is when the autocorrellation peaks of the transmission coincide. In other words when the codes used in the transmission match at the same time. This means that there is a vulnerable period of a few chip periods for each bit where a new transmission can cause data disruption. The cumulative vulnerable period for a packet is small compared to the entire transmission period [47].

An important difference to narrowband signalling that arises in spread spectrum packet radio transmissions is the possibility that two packets collide while at the same time a third packet is successfully received because the third packet is using a different PN sequence and thus does not collide. Multipath affects spread spectrum transmissions by smearing the autocorrelation peaks at the receiver, spreading them over time, causing intersymbol interference (ISI). Multipath effects can be reduced by the use of a multipath-combining receiver such as a RAKE receiver [41] [42].

Receiver-directed bit-homogeneous code assignment is used to reduce the effect of overlapping packets. In this code assignment method distinct orthogonal code waveforms are assigned to different receivers. A transmitter has to use the matching receivers code for data encoding to perform a successful packet transmission. This mode of tranmission ideally limits traffic interference to those transmissions directed to the particular receiver.

The elimination of interference from overlapping packet transmissions can be approached by using *bit-nonhomogeneous code assignment*. In this method the coding pattern varies on a bit-by-bit basis. The receiver requires a programmable matched filter which can follow the bit to bit varying code pattern inorder to decode the signal. If the code pattern is such that it does not repeat before the end of the longest packet, then any overlapping packet would not interfere with a packet locked onto earlier by the receiver because it would not autocorrelate during the entire transmission with the locked onto packet.

Another way to reduce collisions in spread spectrum packet radio networks is to use *transmitter-directed code assignment*. In this assignment method packets transmitted by different radios have different spreading codes, no matter when the transmission began, transmissions beginning at the same time do not collide. However, random errors can occur due to mutual interference between transmissions. This code assignment method has better capture properties than receiver-directed code assignment [40]. However, the receiver does not know which spreading code will be used on an incoming transmission. Therefore it must search the set of all the spreading codes of the neighbouring nodes within transmission range. This requires the receiver to know the spreading codes of the neighbouring nodes and the ability to search for the codes, which requires sophisticated high speed equipment. Such protocols have poor acquisition performance [40] compared to the other code assignment methods described directly above.

Several multiaccess protocols have been proposed and commercial systems have been deployed [13] that use CDMA. Some code assignment schemes have been discussed above with emphasis on the interference and collision properties. Spread spectrum easies network deployment and frequency planning, by allowing codes allocated to each node to uniquely identify each node. If there are more nodes than there are spread spectrum transmission codes, then nodes allocated with the same code must not be in close proximity of each other in order to prevent collisions and to reduce interference as much as possible.

An important design consideration in a multihop packet radio network using spread spectrum modulation techniques is the method of assigning transmission codes to network nodes. The hidden terminal problem can also arise in spread spectrum packet radio, if a receiving node has two neighbouring nodes with the same transmission code, and they transmit at the same time. This problem arises in transmitter-directed and receiver-directed code assignment methods [33]. In order to prevent this code assignment problem require that no set of stations which are two hops apart may have the same transmission code.

5.3 Link Reliability

The data link layer of a network performs the functions that enable a reliable link between adjacent nodes. Usually an acknowledgement mechanism is used to support such a link, for example automatic request queing (ARQ). ARQ uses redundancy for the purpose of error detection, and is generally used with channels which are commonly called *compound-error channels* [26]. If data received by a node is corrupt and an error is detected then a negative acknowledge (NACK) is sent in reply (feedback) and acts as a request for the data to be retransmitted. If the data is error free then a positive acknowledge (ACK) is usually sent informing the transmitting node that data was received correctly. Figure 5.13 shows a block diagram representation of an ARQ system for data link control.

In packet radio the link performance is usually variable due to interference from nearby nodes and radio propagation characteristics such as multipath and Rician fading. In the case of poor link performance the data link has to be made more robust. Forward error correction (FEC) coding such as *block coding* or *convolutional coding* [26] can be used in addition to the ARQ mechanism for this purpose. Bose-Chaudhuri-Hocquenqhem (BCH) codes, Reed-Solomon (RS) codes and cyclic redundancy check (CRC) codes are some common codes types used for forward error correction.

If the error rate is high then ARQ procedures would result in many

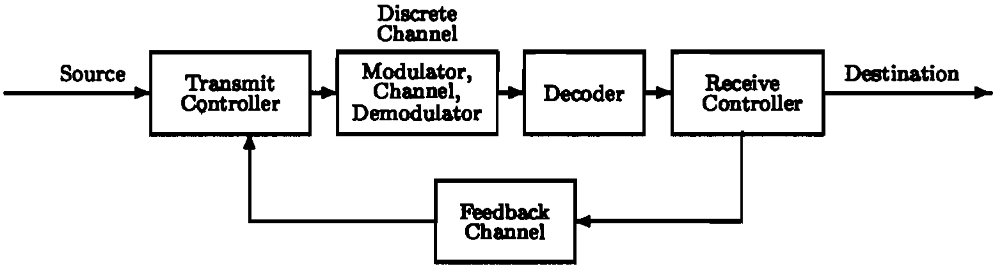


Figure 5.13: Block diagram of an ARQ system used for data link control

retransmission requests, thereby using valuable bandwidth resources. The result will be very low throughput of error free data. Although corrupted data is still received on a poorly performing channel, the FEC coding can be used to correct errors in the corrupted signal, thereby increasing the chance of correct reception. This decreases the number of retransmission requests allowing better channel utilisation than if no FEC was used.

The rate of coding for error correction should not be slow as this will reduce the effectiveness of the link by making the link slow. FEC usually requires adding in extra bits of data into the transmission thus reducing the data throughput. If excessive FEC coding is used in the transmission, then the actual data rate can drop significantly. There is a balance between ARQ and FEC that must be reached in order to make ARQ effective without degrading the rate of correct reception by using FEC coding.

A hop-by-hop (HBH) acknowledge can be achieved by various methods. One method entails the receiving node to reply to the sending node with an explicit short message indicating an acknowledgement. The message can be the header of the received packet, since the header uniquely identifies the packet. Another method is to exploit the broadcast property of the radio channel by using a *passive acknowledgement*. To illustrate this scheme: suppose node A transmits a packet to node B. Node B needs to forward the data to node C and to acknowledge node A that it received the packet correctly. Because node A and node C are both in node B's radio coverage area, they will both hear the broadcast. Node C will store and process the broadcast while node A will recognize it as a positive acknowledgement. In this scheme the last hop is explicitly acknowledged as the last hop.

Some problems arise with passive acknowledgement schemes when nodes implement a first-in-first-out (FIFO) transmit queue. Assume in the previous example that the nodes have FIFO transmit queues with node A's queue containing one outstanding packet waiting for an acknowledgement from node B. Node A transmits the packet at the head of its queue to node B. Node B receives the packet and places it at the end of its transmit queue. In the mean time node A is waiting for node B's passive acknowledgement but node B still has to service all the packets in its transmit queue that come before nodes A's packet which is at the end of the queue. This can severly degrade network utilization. In such a case it would be better to use an active acknowledgement scheme where the acknowledgement signal is much shorter than the packet length.

Another problem that arises is due to passive acknowledgements being as long as the original packet. They are more likely to be exposed to interference than a shorter acknowledgement. In this case the original transmitting node would have to wait a timeout period and then would have to retransmit the packet because it did not receive the passive acknowledgement. This will further degrade network performance.

Passive acknowledgement cannot necessarily be used in spread spectrum systems which use receiver-dependent codes (but can used in transmitter-directed acknowledgements [40]). Referring back to the original passive acknowledgement example: suppose the network is implemented with receiver-directed codes (i.e. each packet is coded with a code which can only be used by the intended receiving node for reception and decoding purposes). Node A codes and transmits the packet intended for node B with node B's code (node A can only use node B's code when transmitting). If node B wants to send the packet to node C it has to code it with node C's code, which node A will not recognise. So, node A will not recognise B's transmission to C because it will not detect it due to the coding scheme.

Using an explicit acknowledgement results in more channel time being used because it accompanies the transmssion of the packet that it acknowledges as being correctly received. However, it is much shorter and therefore less prone to interference, thus resulting in increased reliability of explicit acknowledgements compared to passive acknowledgements.

We have discussed methods that improve the reliability of a link between two adjacent nodes. Source and destination nodes may be many radio hops apart. It is a network management task to ensure that the multihop link is reliable. Another link-level type acknowledgement scheme is end-to-end (ETE) acknowledgement. With end-to-end acknowledgements the final destination node of a packet in the multihop link will send a packet back to the source node acknowledging the packet it received. Therefore for every packet received a packet must be sent, and any node failure will increase delay because ETE acknowledgements travel multiple hops. In some computer networks where data integrity is very important it may be necessary to use ETE acknowledgements to ensure error free data reception. Studies have shown [40] that in a 5 hop route with a probability of 0.7 for packet transmission success per hop, the total average delay of a packet is 12.5 packet transmission times for a system using a HBH acknowledgement mechanism, compared to 53 packet transmission times for a system using an ETE acknowledgement mechanism [25].

In a voice packet switch multihop network the constraints are not as stringent because a conversation can still be continued with loss of a small amount of packets. The main constraint for voice communications is delivering speech packets at a guaranteed low delay. Voice quality degrades significantly when 1 % to 2 % of the voice packets are dropped [43] or corrupted. With the advent of modern compression techniques and vocoders it is possible to significantly reduce the amount of data required in a packet to reproduce acceptable speech quality. It is not neccessary to use HBH acknowledgement methods in a packetised voice communication system if no more than a small amount of the packets are dropped or corrupted at the final destination node or lost before reaching the final destination node.

A decentralized channel management scheme for multihop spread spectrum packet radio networks proposed by Shepard [45] ensures that packets are only sent at times when they will not be dropped due to collision. Such a method would not require an acknowledgement scheme. No practical information could be found for this channel management scheme so it is mentioned here for completeness.

In addition to data link control, there must also be a relationship between the level of the network that controls routing and the control of the link parameters [17]. For example, if servere interference occurs during a broadcast and link connectivity is lost then the network must decide whether to try to re-establish the lost link or to re-route the packet by connecting to a different node. Re-establishing a link may be achieved by increasing the transmit power, thereby effectively increasing the signal-to-noise ratio of the signal or by increasing the coding gain using FEC (assuming an AWGN channel with little or no multipath effects). If increasing the transmit power does not interfere significantly with other neighbouring transmissions then it is preferred to increasing FEC coding gain, because increasing FEC coding gain will reduce the effective data bit rate, while increasing transmit power will not.

5.4 Routing and Packet Forwarding

The job of network management algorithms is to ensure packets are routed through the network in an efficient and reliable process. Inorder to achieve this, the management algorithms must establish the routes (paths) for source-destination pairs and enable the forwarding of data packets along the routes. The choice of routing algorithm and the process used for disseminating routing information will significantly effect the capacity of the network because routing affects the traffic flow of the network. Thus ineffective routing may result in long paths when shorter paths are possible thereby reducing packet delivery speed, or may result in congestion because the network is not coordinated effectively.

Routing protocols for computer networks and internetworks are based on shortest-path algorithms. In general, common shortest-path algorithms can be classified into distance-vector algorithms (DVA) and link-state algorithms (LSA).

In a distance-vector algorithm, a router (node) knows the length of the shortest path from each of its neighbouring routers to every network destination. A DVA uses this information to compute the shortest path and the next router in the path to each destination. A router will transmit updated routing information only to its neighbouring nodes. Each updated message contains a vector of one or more entries which specifies the minimum distance to a given destination. Most DVAs are based on a distributed implementation of the Bellman-Ford algorithm to compute shortest paths [20].

In a link-state algorithm, each node broadcasts messages containing the state of each of the routers' adjacent links to every other router in the network (flooding information distribution). Each router uses this information to compute the shortest paths to all network destinations. Therefore each node contains a replica of information of the complete network topology.

Incremental routing or *source* routing can be used for routing data packets. In *incremental* routing (also called hop-by-hop, point-to-point, or destination based routing) each router in the path to the destination node makes a decision of where to route the packet next. In other words, each router *decides* which node it will send the packet to.

In *source* routing the packet source router specifies the entire path, from source to destination, in the header of each data packet, alternately a connection is established with only the connection-establishment packet containing the specific path, while the rest of the packets contain a connection identifier (effectively a virtual circuit/connection).

If the wireless network has a fixed (static) topology then source routing may be preferrable because the the packet overhead is small, thereby leading to better utilization on the channel. Incremental routing can be used to route packets along multiple loop-free paths. This can be more desirable than source routing because in incremental routing the routers forwarding a packet can react more quickly to changing local network conditions, as in mobile networks, whereas the source router in a source routing network needs to modify the entire routing path to cope with changing network conditions.

Source routing also limits the number of different paths that can be used to those known by the source. This means that only a small number of possible paths can be used (depending on harware and software constraints). This may not be effective in congestion conditions, where it may be that all the paths known to a source router are congested, and thus the packet delivery speed will be reduced. Furthermore, the network throughput will be further reduced because the offered traffic to congested routes is increased. In contrast, multiple loop-free paths obtained using incremental routing allows packets to flow through paths unknown to the source router. If the packets arrive in a different order than they were sent, then the destination router can rearrange the packets using appropriate harware and software. Other methods exist to ensure that packets arrive at the destination in order [32].

A third class of routing strategies called *link-vector algorithms* (LVA) [30] was recently (1997) introduced by Garcia-Luna-Aceves and Behrens to address the scaling problem associated with traditional DVAs and LSAs. In LVA, a router communicates with its neighbours the cost of those links that belong to its preferred paths to each destination. It has been shown that LVAs are more scalable than LSAs and DVAs for flat addressing structures (i.e. non-hierarchical networks). However, a flat addressing structure is not sufficient for networks with many nodes and destinations. Any source based routing algorithm would result in large information storage requirements for nodes to keep a replica of the network topology information. Furthermore, computation requirements and communication overhead would be increased in large flat structured networks and may become to costly.

A method of address aggregation is used to reduce the information storage requirements of a node. The purpose of an address aggregation scheme is to reduce the size of the topology information database or routing tables stored in routers (in the nodes), thus reducing the amount of data that has to be processed, and included as header routing information. Reduced header routing information (overhead) allows for better network utilisation by reducing the channel occupation time for packet transmission. The router database stores two sets of information: one set is of the links it is close to, and the other set describes nodes that are nearby. Therefore the routers store local link and topology information. Hierarchies of addresses are formed by grouping together the addresses of clustered nodes (i.e. nodes which are close to each other).

Two protocols have been found to determine well defined areas (cluster of nodes) of a network or internetwork, namely OSPF [37] and ISO IS-IS [3]. Areas are defined statistically, and are interconnected via a backbone connection. OSPF requires all intra-area traffic to be routed via the backbones. Many routing schemes have been proposed based on the concept of aggregated address areas. The first such proposal was McQuillan's [36].

An area-based hierarchical routing scheme using a link-vector algorithm has been proposed by Behrens and Garcia-Luna-Aceves [31]. The scheme is called an area-based link-vector algorithm (ALVA). It supports a multiple level hierarchical structure and does not rely on a backbone to route traffic between areas. Behrens and Garcia-Luna-Aceves ALVA allows for more flexible topologies and shows improved performance by removing the bottleneck backbone [31]. Simulations by Behrens and Garcia-Luna-Aceves show that even with a one level hierarchy ALVA outperforms QSPF in terms of storage and communication requirements. Furthermore, with the use of hierachical addressing ALVA is more scalable than OSPF, and therefore constitutes the basis for a more efficient internetwork routing protocol based on link-state information.

Using OSPF is useful if one designs a network to use a hierachical structore using backbones to connect various subnets. If a flat strucutre is required then it makes more sense to use ALVA based hierachical addressing.

Chapter 6

Fixed Topology Multihop Packet Radio for Developing Rural and Urban Township Communities

Fundamental issues relating to data packet transmission and routing have been presented in Chapter 5. We now draw on this knowledge to describe a hypothetical proposition for a multihop packet radio network. The networks application is for fast rollout of telephonic and internet service in developing communities. The packet radio network described below is more suited to low density and low population communities. This proposition serves as a rough sketch of a possible network design for developing communities based on multihop packet radio networks. Advanced simulations of this network would need to be carried out to prove its usefulness for its proposed purpose of providing voice and Internet ability to developing communities. All the material that has been presented thus far is used to serve as a platform for the following suggestion of a possibly capable and realisable network. The network proposed is not unrealist because similar networks have been deployed for purely wireless Internet function [2] [18].

The prerequisite to fast rollout is ease of deployment, and therefore wireless technology is the first choice for the network. It has been shown that various wireless LOS systems are available that can be used for the purpose of providing wireless communications. The network proposed here differs from these single hop "last mile" type networks by employing a multihop strategy, thereby being more suited to an internet-like environment.

6.1 User Terminal and Network Access

The network uses nodes with store-and-forward ability to transport information via paths consisting of several nodes between the source and destination nodes. A node consists of:

- an omni-directional antenna
- radio
- microprocessor with memory

A node by itself acts as a repeater and router allowing connectivity to other nodes through its store-and-forward operation. This network is required to provide users with Internet and telephonic communications, therefore the node is combined with the following:

- interface for telephone, fax, personal computer to the radio/microprocessor system
- a telephone handset and dialing keypad

This makes up the user terminal (figure 6.1). It is possible for the user to connect a personal computer via its serial port to the user terminal in order to acquire Internet access. It is assumed that the telephone will not automatically digitize the voice signal so interfacing hardware must provide this function. The interface must provide sufficient connection ports to allow fax, personal computer and other communication devices to be connected at the same time.

6.2 Network Deployment

The network must be easily deployable, therefore multihop wireless networks were chosen. They can be configured easily to various ad-hoc topologies and terrains where line of sight (LOS) transmissions are not always practical or possible between source and destination nodes. Multihop packet radio is commonly used in ad-hoc networks [18] [45] [28] [32]. In fact, multihop packet radio networks employing wireless internet gateway technology, such as those produced in the WINGs project [2], are useful as an "instant communication infrastructure in disaster areas resulting from floods, earthquake, hurricane or fire". This means that wireless self-organising networks allow for quick deployment.

Multihop packet radio networks can be made self-organising by using centralised or decentralised control. The PRNET multihop network,

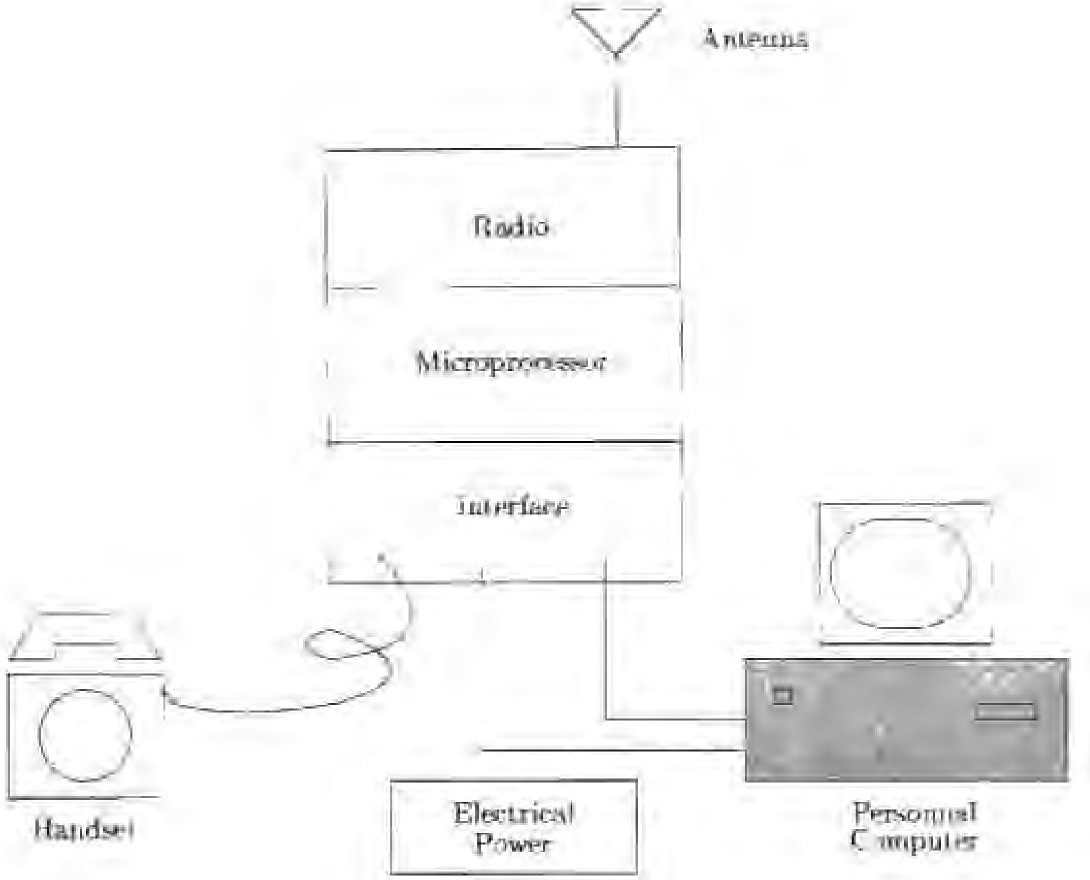


Figure 6.1: A Typical Network Node/Terminal

investigated by DARPA, originally employed centralised control, but it evolved into a decentralised network. The primary reason for this is that decentralised networks are easier to deploy and can function with partial loss of network nodes due to the inherent connectivity redundancy provided by nodes. If the control node of a centralised network is disfunctional, then the network will not operate correctly.

The approach here is to use a decentralised network. However, in order to provide connection to existing network infrastructure, such as a public switch telephone network (PSTN), a network interface must be provided. A specific type of node is necessary to provide this interface. We call this node an *interface node* (IN). The IN acts as a gateway to the packet radio network. Communication among other *local* network nodes continues without the use of the external network interface node. If a user wants to call someone outside of the network (e.g. in another city or country) then the call is routed through the interface node. The IN is therefore also used to connect the nodes to the global Internet.

If no IN is used then the packet radio network functions as a local internet/telephone system, by providing a distributed computer network to every person with access to a network terminal (node). If the service can deliver 33.6 kbps data traffic then it can be expected that the quality of the voice traffic will be equivalent to Internet quality voice transmission. According to the Rooftop Community Network company they claim the Rooftop multihop packet radio network, based on wireless Internet gateways, can deliver Internet quality voice transmission.

The IN can also be used to interface with other multihop networks of its own type. This enables network expansion beyond any limitations of the local multihop network. In effect, a collection of these local multihop networks connected via INs will constitute a larger network with a hierarchy. This is the approach implemented by the Rooftop Network [18]. INs can be connected via high-speed fibre or coaxial cable or via high capacity radio links. INs could also use a packet radio structure with sectored antennae.

When the network is deployed, each user is provided with a node. When a node is activated, initialisation protocols are used to determine the context of the node with respect to its neighbours and its link connectivity to at least two neighbouring nodes. If the network requires access to an external network, then an IN can be supplied. It is preferable to have the IN centralised so that the longest path to the IN is as short as possible and uses a minimal number of nodes so that delay is minimised.

Figure 6.2 may be a typical local network, and figure 6.3 is an example of a hypothetical hierarchical network of local networks connected via the interface nodes (INs). Routing between INs is not in the scope of this thesis. We are interested in a local multihop packet radio network that can provide Internet access and last mile telephone service.

A similar concept to INs is used in the Rooftop Networks [18] where they have a device called an *AirHead*, which stands for Air-to-Internet-Router, to serve as a connection to the global Internet or as a connection to other local multihop packet radio Rooftop networks. The advantage of using an IN (or Airhead) type approach is that with the local network amortizing the cost of the single fast link to the global Internet, these Internet-access costs can be shared, thus providing high speed Internet access at low cost per person to the local network community. It thus saves the community money and makes Internet access more affordable to community members.

Futhermore, if the community only requires a local community type internet access (i.e. not connected to the global Internet) then they only have to pay for their network terminal (node) and any maintenance costs. This is useful because institutions such as schools, clinics, libraries, police department, etc. can all be connected to the network via a node, and thus all residents are also connected to these institutions. Thus a local internet is formed.

Once the user has installed the node equipment and antenna, their radio node can be connected to the user's computer or to a local computer network in one of three ways— 1) Over a serial connection to a single computer, 2) over a link to a local area network such as an Ethernet, 3) over a short-range wireless link to a desktop or mobile laptop.

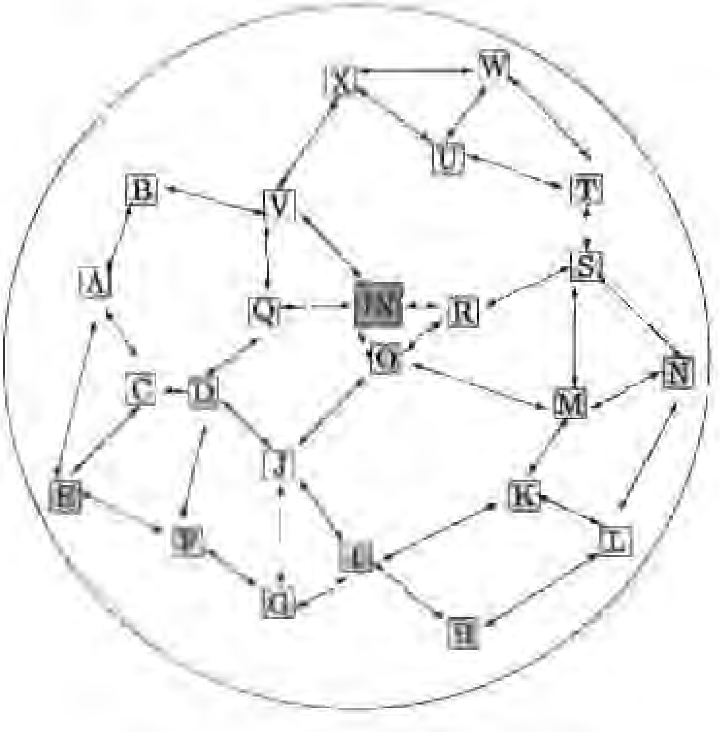


Figure 6.2: A Typical Local Network

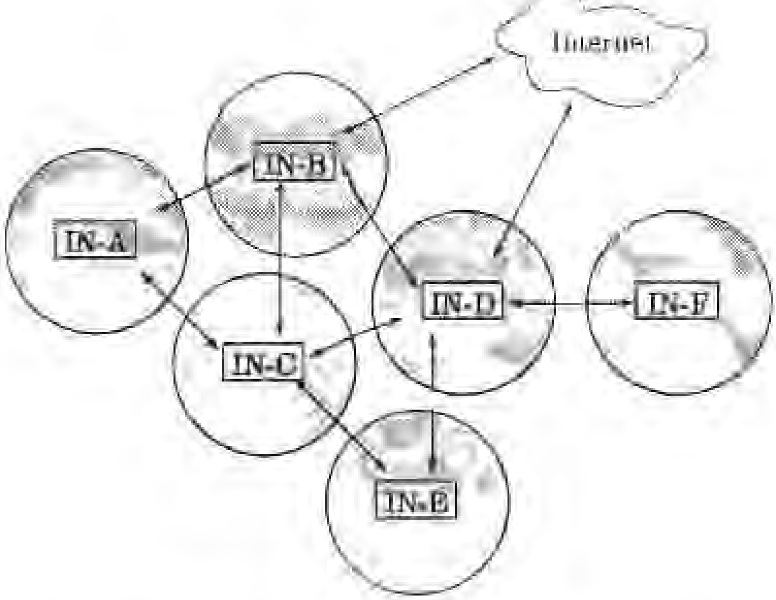


Figure 6.3: A hierarchical network of local node networks connected via the interface nodes (INs)

6.3 Local Network Size

The maximum size of the network depends on the management protocols and node hardware to enable the network to operate to specification (i.e. such as an acceptable voice quality specification). The management protocols affect the routing strategy which affects the hardware requirements of the nodes. For example, source routing requires each node to contain a replica of the network topology [31]. In large networks this may require expensive hardware to enable high-speed operation. Nodes are required to do high speed calculations because of the large amount of topology data that needs to be processed to determine routes.

A fundamental limitation on multihop network size is the cumulative delay that results from a long multihop path via a large number of nodes. The cumulative delay is mainly due to the combination of propagation and data processing delay. The size of the network is limited by the maximum allowable delay for voice communication. Therefore the store-and-forward processing time of a node must be minimised. This requires high-speed processing capability of node hardware.

If the network is only going to be used for data and internet computing and communications, then it can be extended in size further than the voice limitation.

6.4 Packet Delay

It has previously been mentioned that voice traffic is sensitive to delay, and that voice packets are dropped if they arrive late at the final destination node. Each node has an associated processing time due to store-and-forward operation. A maximum delay of between 0.01 and 0.1 seconds is reported [42] as acceptable to deliver recognisable speech, and a maximum delay of 0.001 and 0.01 is acceptable for video and therefore multimedia applications in personal communications networks. Any voice packets delayed more than the maximum allowable delay are considered late and are dropped.

The propagation delay between nodes is determined by the distance a signal travels between the nodes. We have a cumulative propagation delay when a packet travels multiple hops. A cumulative processing delay results from a packet traversing multiple nodes. The cumulative packet delay is the sum of the cumulative propagation and processing delays. We see that the cumulative delay will also determine the maximum allowable nodes in a source-destination route. In the Rooftop Community Network the maximum allowable hops is six.

The type of routing also affects processing delay. In general, incremental routing requires less network overhead in the packet header than source routing. Therefore incrementally routed packets can be processed faster. However, this does not mean that they will have an optimal route, especially if they encounter congested or blocked nodes and have to be rerouted along longer paths than anticipated by decentralised routing strategies.

6.5 Routing

We have already discussed incremental and source routing in Chapter 5. In general, incremental routing requires less control overhead in packets thereby providing better network utilization than source routing. However, it is possible to use a source routing strategy to reserve a path, and thereby reduce overall packet overhead.

We propose a hybrid routing strategy. Voice traffic is given priority over computer data and internet traffic. Voice traffic is routed using source routing and each voice packet is given a priority flag that indicates that it should be processed before data packets waiting in the nodes transmit buffer. Data packets are routed using the incremental ALVA algorithm. Data packets and voice packets are identified by an indication bit in the packet header.

The disadvantage with this proposed scheme is that it requires two separate memory stores: one for the partial routing topology information required by ALVA, and the complete routing topology information required for source routing. One way to get around this is to write software which allows both routing strategies to access the same information, but in the case of the incremental strategy only a subset of the information is accessible.

6.5.1 Routing: Voice Packets

In this particular source routing strategy an initial route is setup by a specialised packet sent from the source node to the destination node. The route is calculated from the complete network topology information database that is stored in the source node. All other packets from the source node following this packet use an identifier which indicates to intermediate nodes that it is using the predefine reserved path specified in the first packet. The nodes in the reserved path will recognise the packet identifier and forward the packet to the next node along the reserved route.

This identifier requires minimal information and therefore reduces network overhead contained in the packet header. Also the packets are guaranteed a path to the destination node once the path link has been established. The source routing strategy is useful for voice because it guarantees a route to the destination and minimises the voice packet header size, which in turn reduces the packet length. Smaller packets have a reduced probability of being corrupted or of collision during link transmissions because they require less transmission time (air time) than longer packets. Source routing should be implemented for voice communications.

If a node forms part of a reserved route using the above mentioned source routing strategy, it does not mean that it cannot be accessed by other data packets while it is not processing voice packets. The node can still be used for routing other data and voice packets. Data packet transmissions are routed along routes other than the voice routes. However, this may not always be possible.

Various recovery methods can be applied in the event that a link along a source routed path (reserved route) fails (e.g. a node fails or a link experiences excessive interference). One approach may be to terminated the connection, or a feedback mechanism like ETE acknowledgements can be used to instruct the source node that a new route must be established in order to continue transmission. In the node/network initialising stage of node/network deployment the link connectivity between nodes is established. This can be achieved using a measure of signal strength and/or bit error rate from transmissions received from neighbouring nodes. Only good link connectivity is considered for transmission. The proposed network is for fixed telephone, not mobile, therefore the event of a node or link failing is reduced, compared to mobile network nodes. Therefore ETE acknowledgements should not be used, only HBH acknowledgements.

Two-way Voice Traffic

In order to have a voice conversation via telephone, the users involved must be able to hear each other and talk to each other at the same time. They require full-duplex handsets. This can be achieved by having two voice packet routes between the users. For example, packets traveling across route 1 go from user A to user B, and packets travelling route 2 go from user B to user A.

The approach used here is to use a FDMA technique at the node. The node must have the ability to receive and transmit at the same time. The easiest way to achieve this is to have the transmit frequency different to the receive frequency. Such a node allows route 1 and route 2 to be assigned to the same nodes between users A and B. Thus the routes are the same but there packet transport direction is opposite. We call this combination of route 1 and route 2 a multihop packet radio voice channel.

6.5.2 Routing: Data Packets

On the other hand, we want the network to deliver computer data and internet services. The *local* network is not hierarchical. For this purpose an incremental routing strategy, such as ALVA, is favoured for local networks. Incremental routing is favoured for data transmissions because data is not delay critical. Of course delay should be minimised which is the reason that the ALVA routing protocol is used. [31] showed that the ALVA protocol outperforms the popular OSPF routing protocol for finding shortest paths and minimising memory storage and hardware processing requirements.

To allow the local network to connect to the rest of the Internet without forcing each node to maintain large routing tables to hold the many address known for the rest of the Internet, the INs can mask the majority of the routing information from the rest of the local network. This serves to reduce the amount of address content required in the network, and also reduces the network address overhead contained in data packets required for routing.

6.6 Handling Malfunctioning and Malicious Nodes

If a node forwards information to another node and does not receive an ACK or NACK after a certain time-out period, then it will retry access the node once more. If no response is received after a certain time out period then it assumes that the node is malfunctioning. The node which is trying to forward the information will then broadcast a special packet indicating which node is not operational. This will effectively erase the malfunctioning nodes' entry in all routing tables.

This packet will also be forwarded to the network administration (network telecommunication authority) center to inform the center of the malfunctioning node. The center can then investigate the problem. In many cases the user of the phone will realise that their phone is malfunctioning and will try to inform the network administrator of the problem.

The node which determined the problem with the node it was trying to forward a packet too will then redirect the packet to another neighbouring node in order to allow the data to continue to its final destination.

Any node that disrupts network operation from the networks normal operation is considered a malicious node. For example, a node which transmits when it knows that the channel has already been reserved by another node thereby disrupting other transmissions by causing packets to collision. In such a case an RTS will not be transmitted or a neighbouring receiver will receive non-coherent information which occupies the channel assigned to it. In this case the receiving node will inform the network administrator about a problematic node in its neighbourhood. The administrator will then have to organise a maintenance service of that neigbourhood to identify the malicious node and try to correct the problem.

6.7 Numbering Scheme

A numbering scheme is important in order that each node can be identified, without which routing will not be possible because source and destination nodes will not be identifiable. Two numbering schemes may be used for assigning telephone numbers to the networks nodes in the proposed. The first is a *self assignment numbering scheme* and the second is a *pre-sales assignment numbering scheme*. The latter scheme has been chosen for this network.

A Self Assignment Numbering Scheme for the Local Network

This method requires that each node be allowed to randomly choose an identification number from a large number range. The size of the number range would have to be bigger than the *local* network node population size. It is assumed that the number of network nodes in a local network will be no larger than a four digit number. This means that the number range is from 0000 to 9999 (ten thousand numbers). Each number is chosen using a uniform random number generator in order to minimise the same number being selected by more than one node.

Once a node has selected a number it will identify itself by broadcasting its identification number to its neighbours. This information will be used in routing tables. It is quite possible that more than one node will choose the same number. In this case each node will randomly choose another number. Each network node will regularly check its routing tables to see if two or more node entries have the same number. If a node identifies that another node has a number identical to its own number then the node will change its own number and broadcast the number to its neighbours. If a separate node identifies that multiple nodes have identical numbers then it will broadcast the error in a special packet identifying the nodes in question.

When a node in question receives this packet it will randomly choose another number and broadcast the new number to the network. It is possible that the new number will coincide with another nodes' number. In this case the process of detecting and correcting multiple identical node numbers will be repeated until the network is free of any numbering errors. This numbering scheme is highly dynamic and it is quite possible that this process may take a long time to settle. A reasonable assumption would be that if the network is small (of the order of tens) and if initial node activation of all nodes did not take place simultaneously but, say, over a period of three hours or longer then the process of numbering nodes using this numbering scheme will be completed relatively quickly compared to a large local network with a population size of the order of thousands.

Once the node has determined that it has secured a unique identification number in the local network it will indicate this number to the user via a display device (e.g. a liquid crystal display). This number will be used as the last part of the telephone number.

Pre-sales Assignment Numbering Scheme

The other method that can be implemented is to assign each node a unique identification number when it is manufactured or which can be programmed into the node before consumer sale transaction. This is the method used for ethernet board devices. In such a case a national database would have to be established to control the distribution of node identification numbers throughout regional areas. For example all node numbers sold in Cape Town will be extracted from a range of numbers assigned to the Cape Town region, any nodes sold on Johannessburg will similarly be assigned unique numbers. The result of assigning numbers regionally reduces the number range.

A local network cluster is described in figure 6.3. In such a network topology each local network is connected via a central node (IN) which serves as an interface between the local networks, it also enables connecting the local network to a PSTN and the global Internet.

Each IN within a network cluster is given a pre-assigned cluster network number in a similar fashion (regionally) as local network nodes are assigned numbers. The number is two digits long thus allowing each network cluster to have up to one hundred INs.

A telephone number (which is used in packet headers for routing purposes) consists of two parts: the local network number and the local node number. The local node number is a four digit number (i.e. XXXX) and the local network number (which is an IN number) is a two digit number (i.e. YY). They are combined to form a nodes' (telephone) number (YY XXXX).

Local node numbers can be reused in *different* local networks throughout a network cluster.

Any telephone number that does not conform to this numbering scheme is assumed to be connected via a PSTN.

The Use of Telephone Numbers

If a user wants to make voice contact with another user in the local network they will enter (dial) the respective telephone number of that user. This number is placed in the transmitted packets' header and is used by subsequent nodes to route the packet to the destination node described by the number dialed by the user. Any packet used for voice data transfer indicates that it is a voice packet by asserting a flag in the voice packet structure.

When a receiving node receives a packet it first identifies if it is designated to its local network. If so it continues to check which node it is designated to. If it is designated to itself then it process the rest of the packet, else it will forward the packet to the next suitable node based on the information in its routing table. If the address information in the packets' header indicates that it is designated to an IN other than the local IN then it forwards the packet along a route to the local IN, which will then forward the packet to the designated IN.

Any non-voice data transfers does not assert the flag that indicates voice transfer but still uses local node (telephone) numbers for addressing nodes in the network. This is necessary to allow packets with global Internet addressing information to pass through the network. Any global Internet numbers specified in the packet header will automatically cause the packet to be routed to the local IN which handles global Internet traffic transfers. Any local network or cluster network data transfers will not assert the voice packet indication flag and will use the telephone number for data routing.

To allow the local network to connect to the global Internet without each node maintaining a large routing table to hold the many address known for the global Internet, the IN can mask the most of the routing information from the local network. The IN will decide an aggregate address range for its local network, or in the case of a cluster of local networks, the INs will collaborate to decide who should advertise a route to an aggregate of address ranges in the global Internet. Each IN then distributes routing information messages to a few large address ranges in the global Internet, rather than a large number of information messages corresponding to very small address range. This is the same approach used in the Rooftop Community Network which is similar to the proposed network in this thesis.

6.8 Channel Access Technique

Various packet radio multiple access techniques have been described in Chapter refmulti. We have seen that the typical pure ALOHA and slotted ALOHA techniques are easily outperformed by CSMA, BTMA, and FAMA-NCS.

CSMA gives an 85 % network utilization when used in a fully connected network [16]. It may be that many rural communities may only need a small fully connected network, where the maximum distance between nodes is approximately one kilometer, thus requiring low power transmitters. However, if the rural community expands, a CSMA fully connected network will not be scalable and at the same time able to provide an 85 % network utilization because of the hidden node problem [16]. It would be preferable to give the community the option for network expansion and therefore a scalable network is necessary.

BTMA and FAMA-NCS are collision avoidance channel access techniques. FAMA-NCS has been shown to outperform all the multihop packet radio channel access techniques considered in Chapter 5. FAMA-NCS has also been used in the WINGs wireless internet gateways with success [2]. It has proven itself, and is therefore chosen as the channel access technique for this network.

6.9 Spread Spectrum CDMA

CDMA has been shown to use codes to modulate a signal onto a wide frequency band. If the codes are chosen to be orthogonal, as is the case in CDMA, then bandwidth efficiency is improved in a multiuser environment [42]. Codes can be automatically assigned to nodes in the multihop packet radio network at network/node initialization: they may also be reused in large networks. This practically removes the need for radio planning; which usually takes a relatively long time for narrowband systems unless dynamic frequency allocation is implemented, such as in DECT systems (Chapter 4) [24].

The other benefits of using CDMA is that it has inherent coding gain [12] which results in CDMA systems having excellent robustness in the presence of multipath and Rician channel fading effects. The code aspect also allows increase network capacity [42] because adjacent nodes will not cause a collision if they transmit using different codes [40]. In other words, a collision of the type described in figure 5.4 will not occur. Figure 6.4 shows this collision free situation. Transmission from node C to node B results from the omni-directional broadcast required for the transmission from node C to node D.

A receiver-directed bit-homogeneous code assignment scheme [10] [17] is used for the preamble of the data packet. The data packet following the preamble is coded with a transmitter-directed bit homogeneous code. The preamble contains a code identifier that instruct the receiver which code to use to despread the received packet transmission. In this scheme the preamble is more prone to collison than the packet. Once the preamble has been received error free then the probability of the packet colliding is negligible because no other transmitter will be assigned the same code in the two hop radius surrounding the transmitting node.

In the voice routing section of this chapter (section 6.5) we mentioned that the nodes use an FDMA technique to enable two-way voice communications. Therefore two spread spectrum channels are used, one for voice in

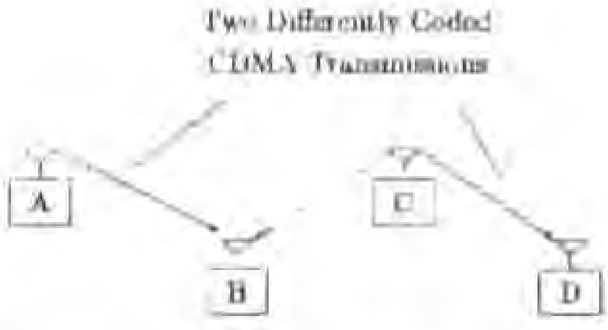


Figure 6.4: Transmission C to D does not collide with transmission A to B: transmission C to D succeeds, transmission A to B succeeds, transmission C to B has no effect

the directional from user A to user B and the other in the opposite direction. Computer data packets (such as Internet data carrying packets) use either channel when it is free.

6.10 Link Reliability and Acknowledgement Scheme

We have discussed end-to-end (ETE) and hop-by-hop (HBH) acknowledgements in Chapter 5. The disadvantage with ETE acknowledgements is that the total average delay is increased, relative to HBH acknowledgements, between source and destination node pairs [16] [25]. For this reason ETE acknowledgements are not used. An automatic request for transmission (ARQ) type HBH acknowledgement is preferred. Any corrupted packets cause the receiving node to transmit a negative acknowledgement (NACK) to the transmitting node. If the packet is unrecognisable by the receiver then the transmitting node will retransmit the packet after a specified timeout period if no ACK or NACK is received.

Forward error correction is also necessary in wireless links [42]. The use of FEC will reduce the number of ARQ retransmission requests, thereby increasing link utilization and improving packet transport speeds. A careful balance between FEC and ARQ must be reached so that link data rate and reliability are optimized.

6.11 Vocoder

A vocoder is a speech coding device. Its purpose is to reduce the amount of information required at the receiver to allow for recognizable voice reproduction. The device analyses the voice signal, extracts voice characteristics and then transmits them. These characteristics are used to synthesize the voice at the receiver. In general, vocoders are complex devices and achieve very high economy in transmission bit rate [42].

The IS-95 CDMA standard [42] specifies the QCELP13 vocoder from Qualcomm [11]. QCELP13 is a code-excited linear predictive coder. Its effective information data rates are 14400, 7200, 3600, and 1800 bps. When the channel is idle (about half the time) then the vocoder transmits at 1800 bps. In the CDMA channel this helps reduce channel activity and therefore reduces interference with other active channels. Furthermore, it serves to reduce random errors in packets and increases channel capacity.

A QCELP13 vocoder is used to minimise voice data from any telephone signal, and because of its good voice reproduction quality.

6.12 Electrical Power

Electrical power can be administered to the node in the same way Telkom connect their equipment to electricity. However in some rural communities there is no electrical power at all. A solution similar to Telkom's solar power system can be used.

6.13 Urban Township and Rural Application

6.13.1 Rural Application

The packet radio network can be applied to small rural communities that are very distant to major telephone or Internet switching centers. In this scenario the multihop network can be used as a local internet and supply an internet quality telephone service. It is expected that data traffic in these communities will be small until the community develops into a large community at which time they can upgrade the network to different technologies is possible and/or required.

The fact that the community is not near a major switch center means that a link must be made between the local multihop network and a switching center. The quickest way of doing this would be to use LEO or MEO satellites which will to become available to South Africa and most other countries on Earth in 1998. The other option would be to install a microwave link. However, this may be more expensive than the satellite solution. This can only be determined when satellite service cost are known.

6.13.2 Urban Township Application

In urban township communities like Khayelitsha, the high population density will create high demand for services. Effective voice communication may not be possible, however the will have to be determined through further investigation and research. The multihop network will still be able to provide Internet access. This is important especially in high density areas where networks are not easy to deploy. A self-organising packet radio network can be easily and quickly deployed.

An urban community is usually situated near a switching center and can be easily connected with a short distance microwave link or via a short underground G.703 trunking cable connection such as the one used by Telkom for their DECT system.

6.14 Billing and Revenue

The purpose of the proposed network is to enable rapid rollout of telephone services, not necessarily to achieve large revenues. Of course the network provider must be able to make a profit from the network and receive revenue to maintain and operate the network. However, it may be the case that the network belongs to the community using the network, in which case the community itself will be the service provider.

6.14.1 Co-operative Community Network

One of the original ideas behind this proposed network was to enable communities to achieve relatively cheap telephone and global Internet services. This network can achieve this if local network (or clusters of local networks) calls are not billed, only calls to a PSTN or external Internet service provider are billed. This results in the community having cheap and easy access to the local telephone neighbourhood. The maintenance costs of the network can be raised by a monthly surcharge to each user. In this way the network is run as a co-op, specifically serving the community in a co-operative manner.

By taking advantage of the typically bursty nature of Internet access needs of individual users, which are short bursts of high bandwidth followed by relatively long idle periods, the network may effectively share the cost of a single data link to the Internet among a number of local network members. An ADSL line to an Internet Service Provider can be install. ADSL lines have a 6 Mbps downstream (to the local network) data rate and a 640 kbps upstream (from the local network) data rate. An ADSL link may cost up to R3060 (Rand - South African currency) for installation and modem equipment with additional ADSL line provider and ISP service charges of up to R765 per month. For example, a local network community consists of 30 nodes (30 members) with each contributing on average R35 per month for the Internet link. It is a fair assumption [21] that not all members will be using the Internet link simultaneously. Each member will typically benefit from a downstream bandwidth of 1 Mbps or more during each data burst, and an upstream bandwidth of 100 kbps or more [21].

Unlike a dial-up connection of the typical wire based telephone network the proposed network only uses network resources when transmitting or receiving user data. It is therefore possible to have continuous connectivity for all users without bogging down the network. In other words, a lot more traffic would be required to slow the network (the network does not experience blocking, it is only slowed down) than would be required to block a typical wire based telephone network. Continuous connectivity allows instant delivery of e-mail rather than requiring the user to solicit it from an ISP. It also enables the user to become an information provider by hosting a Web site on a home computer without requiring a dedicated telephone line.

It would be in the interest of the community to keep telephone surcharges as low as possible to enable a low cost service, thus making telephone (node) ownership attractive to potential users. This will further develop and expand the network providing large coverage. In other words, the community should not miss use the network to derive large profits, but should use the network to extract sufficient funds to operate the network and encourage its expansion.

The network is designed to require minimal maintenance and to operate automatically. The only interaction that is required by the network provider is to handle malfunctioning network equipment and to manage the scalability of the network (i.e. manage the distribution of INs which operate automatically).

The local network and cluster of local networks is a web of peer-to-peer radio nodes that all automatically participate in forwarding network traffic. Each user is part of the local network and can send data to any other user in the network for free. This is the approach used by the Rooftop Community Network [18].

However, if a user wants to access an external network, say, the PSTN for telephonic communication, then the user would be charged a fee by the PSTN provider. Each IN records the nodes making external calls and sends this information to the network provider for billing purposes.

6.14.2 External Network Provider

If the PSTN provider and local network provider is the same organisation, say Telkom, then they can charge the community using various billing schemes.

A billing scheme can be designed to charge for each packet hop or a general connection charge and then for air time following the connection. The other billing scheme will only charge for calls made to external networks (e.g. a PSTN).

A per hop charge will generate relatively large amounts of traffic because each hop will require the identity of the source and destination node of the hop and then this information with information of the call initiator will be sent to a billing center. This method will generate a lot of overhead network traffic.

A better billing scheme would be to charge a flat connection rate and call duration rate as is similarly done for modern telephone services. The only information that the network provider would require for billing purposes is the call initiator address and the duration of the call. This information can be sent at the end of the call.

It may be that the provider will allow a similar billing system as described for the co-operative community network. Each IN records the nodes making external calls and sends this information to the network provider for billing purposes.

The network provider can make additional revenue from the network by treating the node and terminal equipment as a consumer product. If the consumer wants a telephone service then they must buy a node for their local network. If the node is to expensive for the consumer then it can be rented or payed off in monthly or weekly installments. This raises an important issue which is discussed in subsection 6.15.1, the cost of a node for the proposed network.

6.15 Reducing Terminal and Network Cost

The first factor which contributes significant to the cost of a wireless network is the ability of a wireless network to support mobile users. We see in the GSM network (Chapter 4) that special location registers are required to enable handoff and roaming of mobile terminals. This increases the networks backbone infrastructure and complexity because high-speed and high capacity data links are required to enable the network to move this data quickly through the network. In the proposed multihop packet radio network no mobile facilities will be included because it is primarily focused at providing fixed wireless infrastructure.

Packet radio transmissions would only be required to travel a few hundred meters in a township community like Khayelitsha, therefore the signal power used by a node in the packet radio network to transmit a packet would be much less than the signal power requirement of a GSM network terminal. Lower power signals require lower power electronics and therefore lower power transmitter equipment. This results in cheaper equipment because low power electronic equipment is generally cheaper than higher power electronic equipment.

The other concern that a multihop packet radio network addresses is the antenna size. We saw in Chapter 4 that often large antennae have to be disguised to look like a tree, and that this adds to the time of network deployment. In the multihop network all the antennae are small because LOS is only required over a few meters (or few hundred meters) between nodes. For example, the distance a signal would need to travel between (nodes at) houses in Khayelitsha would not be more that 200 meters because of the housing density in Khayelitsha. This would overcome the need for a 35 meter antenna (such as the DECT antenna in Khayelitsha), and would reduce maintenance costs of the associated mast/antenna station. The maintanance costs of large antennae are also eliminated.

We cannot solely compare the cost of the premises equipment of this network to the cost of premises equipment of, say, a wire based telephone network or even a GSM or DECT wireless network, but must compare the overall network cost. Current GSM and DECT networks require base stations which cost over a million Rand. This excludes installation and maintenance costs. If each house in a community of 500 house receives a node (terminal) then the total network cost will be approximately $5000 \times US\$1100$. (We have estimated terminal premise equipment to cost US$ 1100 in subsection 6.15.1). This amounts to US$ 5500000. This is approximately equivalent to R28 million. If we take into account that this is the cost of the entire network which provides Internet access and telephone services then we see that it is relatively low cost compared to the DECT network currently being implemented in Khayelitsha. This DECT network has a premises equipment cost of approximately US$ 2000 excluding maintenance costs. This amounts to approximately R10200 per house. If it is used in a population of 5000 thousand then just the premises equipment cost of the network will be approximately R51 million. The DECT base station and trunking link to a near by switching exchange will add additional costs to the network infrastructure.

The proposed network is wireless not wire based and therefore eliminates the costs of wiring the network. Wiring a network constitutes a substantial cost of a wire based network. In order to connect a wire based network trenches need to be dug. Special protective tubing must be used on the wire which is then placed into the trenches, which are then covered. In addition to the material costs of the tubing and wiring, additional costs are incurred for the labour used for laying the wire underground. Similarly for pole mounted wiring. An addition cost must be taken into account for the technician who is required for connecting the wires. In the case of a wireless network the technician would be required to install the equipment at the customers premises only, thus the cost of paying the technician for connecting wires has been eliminated.

When wire connection is damage and needs to be repair then the wire must be extracted from the ground or from the pole. This requires labour and may also require additional wire thus contributing to network maintenance costs and also slowing down network repair. Theft of wire has also been eliminated because there is no wire to be stolen. Furthermore, wirebased networks require considerable infrastructure such as buildings for housing large switching exchanges, roads to these exchanges and other network equipment. In general these factors make the wireless network less costly from an installation and maintenance point of view.

6.15.1 Terminal cost

It is important to remember that both Internet and voice services are being provided by the network. Thus the cost of the equipment cannot be compared only on an equipment cost basis to existing telephone services, but should also take into account the Internet service capability of the equipment.

To estimate the cost of terminal equipment we can look at two sources. The one source is to look at similar equipment and use this equipments pricing to estimate a cost of the network nodes proposed for this network. At best this estimate is an approximate upper bound on the equipment cost.

The manufacturer of Rooftop Community Network [18] equipment commercially sell a device called Spirit^{TM}. The Spirit^{TM} is an Internet radio and ranges in price from US$ 2290 for a short distance radio (similar to a node in the proposed network) to US$ 5790 for a long distance Internet radio gateway which can also serve as a AirHead (Air-to-Internet router) which is similar to the IN for the proposed network. These prices include a 10BaseT LAN interface, a SLIP serial host interface, and screw-on antenna. They also sell a 2.4 GHz lightening arrestor for US$ 140 and various combinations of cabelling and connector sets that range from US$ 15 to US$ 100. Excluding installation and power supply costs and the terminal device costs between US$ 2290 and US$ 6030. A PC computer can be used as the terminal and would add on another US$ 1000 bring the cost to an upper bound of approximately US$ 7000.

The developers of the Spirit^{TM} device say that current technology would permit the development of an integrated Internet Radio with a high-volume end-user price of US$ 500 to US$ 750. If this is possible then the upper bound price on a combination of their Spirit^{TM} device and a PC computer would come to no more than US$ 1750. These prices include a profit markup. It is a fair assumption to say that that the product is cheaper to manufacture than the prices of their equipment.

The reason we mention that a PC computer is used as a terminal for the Spirit^{TM} device is that the Spirit^{TM} device is designed for data computing and Internet usage only. If such a device could be adapted to voice usage then a PC computer would not be necessary as the terminal. A normal telephone handset which costs much less than a PC computer could be used as a terminal (at minimum). In this case the combination of the handset and Spirit^{TM} device should cost no more than US$ 1100. This analysis is very coarse and only serves as an approximate estimate for the cost of the device described for the proposed network.

It is assumed that in the case of a network provider who sells node equipment they can offer various packages. For example, they can market a node and telephone handset package which will only offers voice communication, but will still have Internet facilities which can be used by the node users if they buy a PC computer and connect it to the node. The network provider can also market a PC computer and node package only for Internet usage. The node owner could later install a telephone handset for voice usage. The network provider can also sell a package which includes a handset, PC computer and node. This will be the most expensive package and will provide voice and Internet capabilities.

This flexibility of the node could be used to develop profits and also provide cheaper packages for lower income customers thereby allowing network expansion to lower income communities.

6.16 Voice and Internet Traffic per customer

It has been stated in the introduction of this thesis that it is beyond the scope of this thesis to actually simulate the network. This is a difficult problem which requires long simulations and extensive programming of suitable simulation software.

The voice and Internet traffic per customer cannot be determined exactly as each customer will have different demands for network usage. Therefore a general probabilistic approach may be used to determine the traffic per customer.

It has been mentioned in subsection 6.5.1 that voice packets are given a higher priority than data packets. This means that a voice packet will always be forwarded before a data packet is forwarded. The following example using the results in subsection 5.2.5 will explain the implication that this priority structure has on voice and data traffic.

In figure 5.11 we see that a similar network to the proposed network can attain a throughput capacity of approximately 0.30 Erlang when the offered load is 1 Erlang. This means that the network has an efficiency of 30% when the offered load is 1 Erlang. If we assume that 30% of the offered traffic is purely voice traffic then all voice traffic will pass through the network unaffected, and the data packet transfer rate will be decreased.

Each node has a vocoder with a output bit rate of 14400 bps. Each node is designed to have a peer-to-peer data rate of 1 Mbps. If on average each node has four neighbours and three are directing their own voice traffic through the neighboured node to the fourth neighbour node, then the traffic to the fourth node has a bit rate of 3×14400 bps plus 14400 bps for the neighboured nodes own voice traffic. This amounts to 57.6 kbps which does not load the 1 Mbps link significantly (figure 5.11). However, each neighbouring node may be routing voice traffic from other nodes through the fourth neighbouring node, as a worse case scenario. The estimation of the offered traffic may exceed the data rate capacity of a node and will force the other nodes to reroute their traffic in order to prevent significant blocking. Such a scenario requires detailed and complex simulations to estimate traffic behaviour and traffic throughput capacity and is beyond the scope of this thesis. However, for a given load which is dependent on the type of community and its Internet requirements, figure 5.11 and figure 5.12 provide an estimate of the throughput capacity for networks similar to the proposed network.

To be able to predict the traffic generated by a customer depends on many factors such as whether the customer is a private user or commercial company, if the user is situated in a rural or urban community, etc. It is a highly complex task to determine the nature of the traffic generated by a customer and to determine how this traffic will affect the network traffic in general,and is beyond the scope of this thesis. This fact has been stated in many papers in the field of packet radio networks [29] [47].

Chapter 7

Conclusion

The focus of this thesis has been to look at wireless technology that is being used to provide telephonic access to developing communities with little or no telephone services, and to propose a multihop packet radio network that will enable the provision of both Internet and voice communication. The two technologies being used in South Africa at present are GSM and DECT wireless systems.

The purpose of Chapter 3 was to introduced the fundamental concepts used to design wireless communication systems. These concepts where used throughout the remaining text. Time division and frequency division duplexing techniques have been presented. Frequency division, time division and code division multiple access techniques have been described in detail. The most effective multiple access techniques in wireless networks are those that provide high bandwidth utilization, link reliability and immunity to noise. Spread spectrum multiple access can be combined with TDMA to achieve system flexibility and link reliability. This is the approach used in the DECT system which uses FHMA combined with TDMA.

Chapter 4 gave an explanation of two types of wireless networks commonly used in South Africa. It presented a description of the architecture of these systems. From this description we see that these systems, especially the GSM cellular system, are complex and not easy to setup in rural areas because of the long distances that have to be bridged to a switching center, or densely populated areas where it is difficult to find a suitable location for a base station. The use of the DECT system implemented by Telkom and the GSM system used by Vodacom have been explained.

Vodacom have a universal access approach to providing telephone services to the Khayelitsha township and other township and rural communities throughout the country. This approach uses the Vodacom GSM network infrastructure to connect fixed site phone shops to the network. Vodacom franchise the phone shops to local operators who then run them like a business, selling cards to the public that enable the use of the phones in the phone shop. The cards are used with a prepaid billing system (developed by Vodacom) where the user pays in advance for a certain amount of talk time. The results of Vodacoms' ventures in townships all over the country have been successful. The phone shops also offer fax service and are good candidates to be developed into more advanced telecenters. The phone shope service is limited to area which have access to Vodacom's GSM network.

Telkom have a universal service approach to providing telephone service. They have installed a DECT system in the Khayelitsha township. The system enables anyone, who can afford a phone service, to have a phone on their premises. Telkom charge the same standard rates on their wireless DECT phones as they do on their wireline phones. Telkom's approach requires a terminal station to be installed on an appropriate mount such as a concrete wall. In many informal premises suitable mounting places are not available so a special pole mount has to be installed. This reduces the rollout time of the network, and also increases the cost of the network and deployment. According to Alcatel Altec in Johannesburg, the average cost of a system after installation is about US$ 2000. The basic radio terminal costs about US$ 242. According to Telkom in Cape Town, their system does not have sufficient capacity to provide voice and Internet access at the same time so only a voice service is offered.

Both the GSM and DECT systems do achieve their objectives of providing telephone access to Khayelitsha. However, they do not enable rapid rollout and are expensive to deploy due to system complexity and installation costs.

Low power satellite voice communications technology will be launched during 1998. This technology provides voice communication access independent of geographical position. Developing rural communities are a potentially large market for satellites because they have do not posses adequate.

Packet Radio has been suggested to overcome some of the problems associated with the terrestrial networks. The network proposed is a self-organizing multihop packet radio network. We have introduced the fundamental packet radio multiple access schemes: ALOHA, slotted ALOHA, CSMA, and BTMA. FAMA-NCS, a multihop packet radio collision avoidance multiple access technique, was shown to outperform these multiple access techniques, and was thus chosen for the proposed network.

Spread spectrum was shown to be useful for allowing the network to be easily deployable. This can be achieved through spread spectrum because of its use of orthogonal spreading codes. Once a network is installed it is activated and the nodes begin to acquire information about there neighbours. The nodes use this information to determine routing paths, and the signal spreading codes of there neighbours. The codes can be assigned to nodes in such a way that packet collision can be minimised and thus network throughput is increased.

Issues relating to link reliability have been discussed. Hop-by-hop acknowledgements are used in conjunction with a forward error correcting code to provide reliable link connectivity.

Incremental and source routing have been discussed. It was decided to opt for a hybrid routing approach where voice packets are routed using source routing and data packets are routed using the ALVA incremental routing algorithm.

The primary limitation on voice is processing delay at the node (per-hop delay). In [42] a delay of more between 0.01 and 0.1 seconds is suggested to result in low voice quality. This means that if a 20 ms delay exists per-hop, then no more than five hops (0.1 seconds) can be made before voice quality is unrecognisable.

Frequency spectrum allocation of this network should be determined by the user data rate requirement. The WINGs devices work between 905 MHz and 925 MHz, and have a data rate of 289 kbps.

The Rooftop Community Network [18] is a multihop packet radio network that operates using Internet protocols, not the usual X.25 packet radio protocols, and has been shown to be successful with data rates as high a 1 Mbps. It provides an internet service and has been deployed and used successfully. The Rooftop Company claim it can deliver internet quality voice. This indicates that similar networks, such as the network proposed in Chapter 6 can do the same.

Recommendations

With improving radio and microprocessor technologies distributed computing networks such as the one proposed may be able to deliver acceptable voice quality and Internet service to small rural communities, and deliver acceptable internet service to larger township type communities.

A possible area of interest that may be beneficial to study is the proposed hybrid routing scheme where voice is routed using source routing and data is routed using incremental routing. The software required to distribute spreading codes also needs to be developed to determine if codes can be assigned to nodes in a way that minimises collisions. The most important software that needs to be developed is an easy to use package that can simulation multihop packet radio network traffic, because further research will be difficult to quantify without simulated results of network performance for analysis.

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Appendix A

GSM and DECT Radio Specifications

Table A.1: GSM Air Interface Specifications Summary

Parameters	**Specifications**
Reverse Channel Frequency	890 - 915 MHZ
Forward Channel Frequency	935 - 960 MHZ
ARFCN Number	0 to 124 and 975 to 1023
Tx/Rx Frequency Spacing Tx/Rx Time Slot Spacing	45 MHz 3 Time Slots
Modulation Data Rate	270.833333 kbps
Frame Period	4.615 ms
Time slit Period	576.9 μs
Bit Period	3.692 μs
Modulation	0.3 GMSK
ARFCN Channel Spacing	200 kHz
Interleaving (max. delay)	40 ms
Voice Coder Bit Rate	13.4 kbps

Table A.2: DECT Radio Specifications Summary

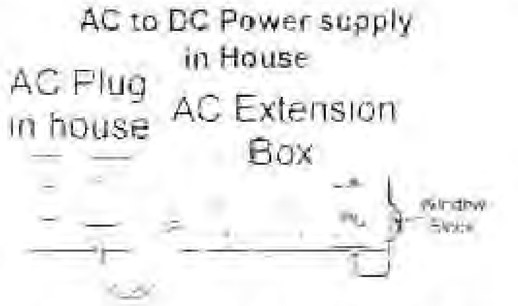
Parameters	**Specifications**
Frequency Band	1880 - 1900 MHz
Number of Carriers	10
RF Channel Bandwidth	1.728 MHz
Multiplexing	FHMA/TDMA 24 slots per frame
Duplex	TDD
Spectral Efficiency	500 Erlangs/km^2/MHz
Speech Coder	32 kbps ADPCM
Average Transmit Power	10 mW
Frame Length	10 ms
Channel Bit Rate	1152 kbps
Data Rate	32 kbps Traffic Channel 6.2 kbps Control Channel
Channel Coding	CRC 16
Dynamic Channel Allocation	Yes
Modulation	GFSK (BT = 0.3)
Speech Channel/RF Channel	12

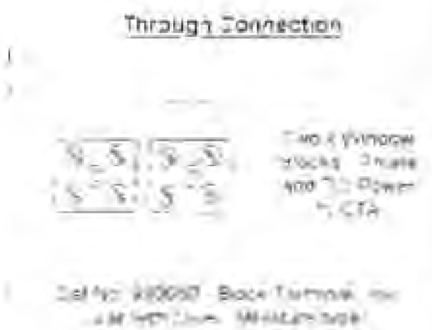
Appendix B

Telkom Radio Installation Diagrams

The following pages have been photostated out of a Telkom installation guide [7] that details the procedure for mounting equipment used in their DECT and CT2 wireless networks.

DECT/CT2 - CTA Installation



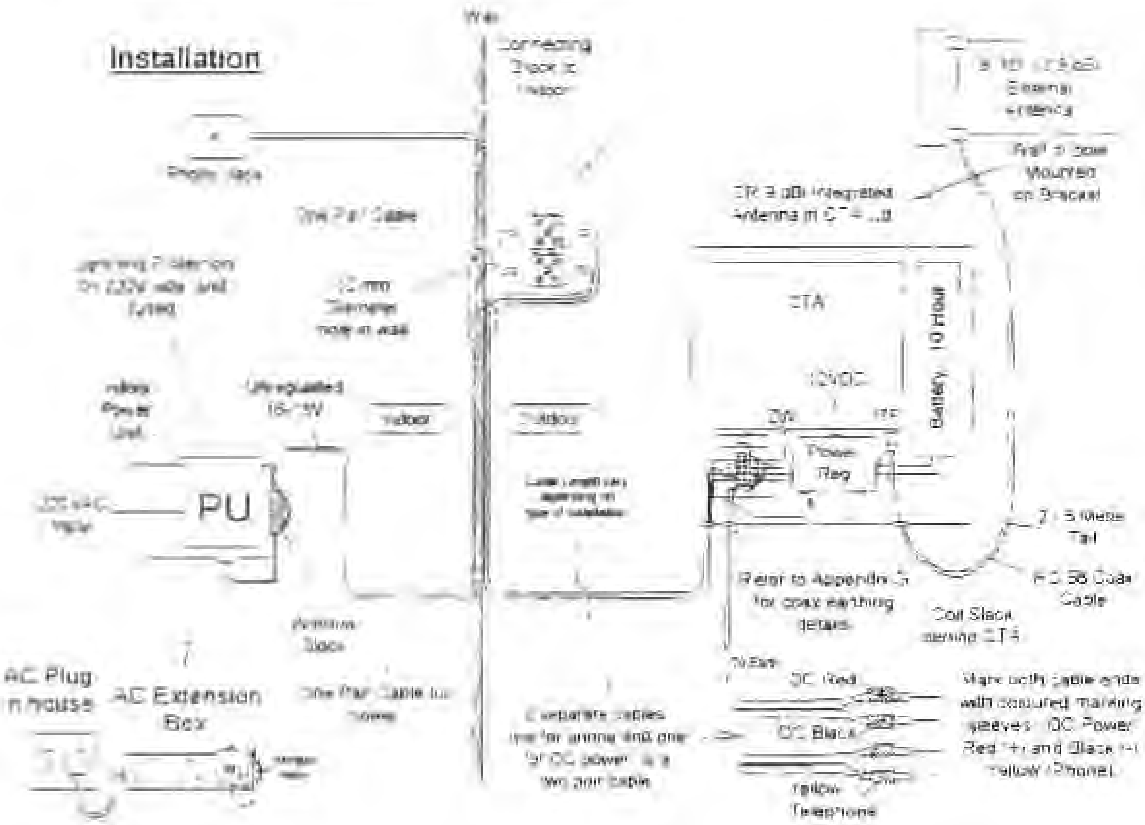


DC Power Cable to Outside CTA

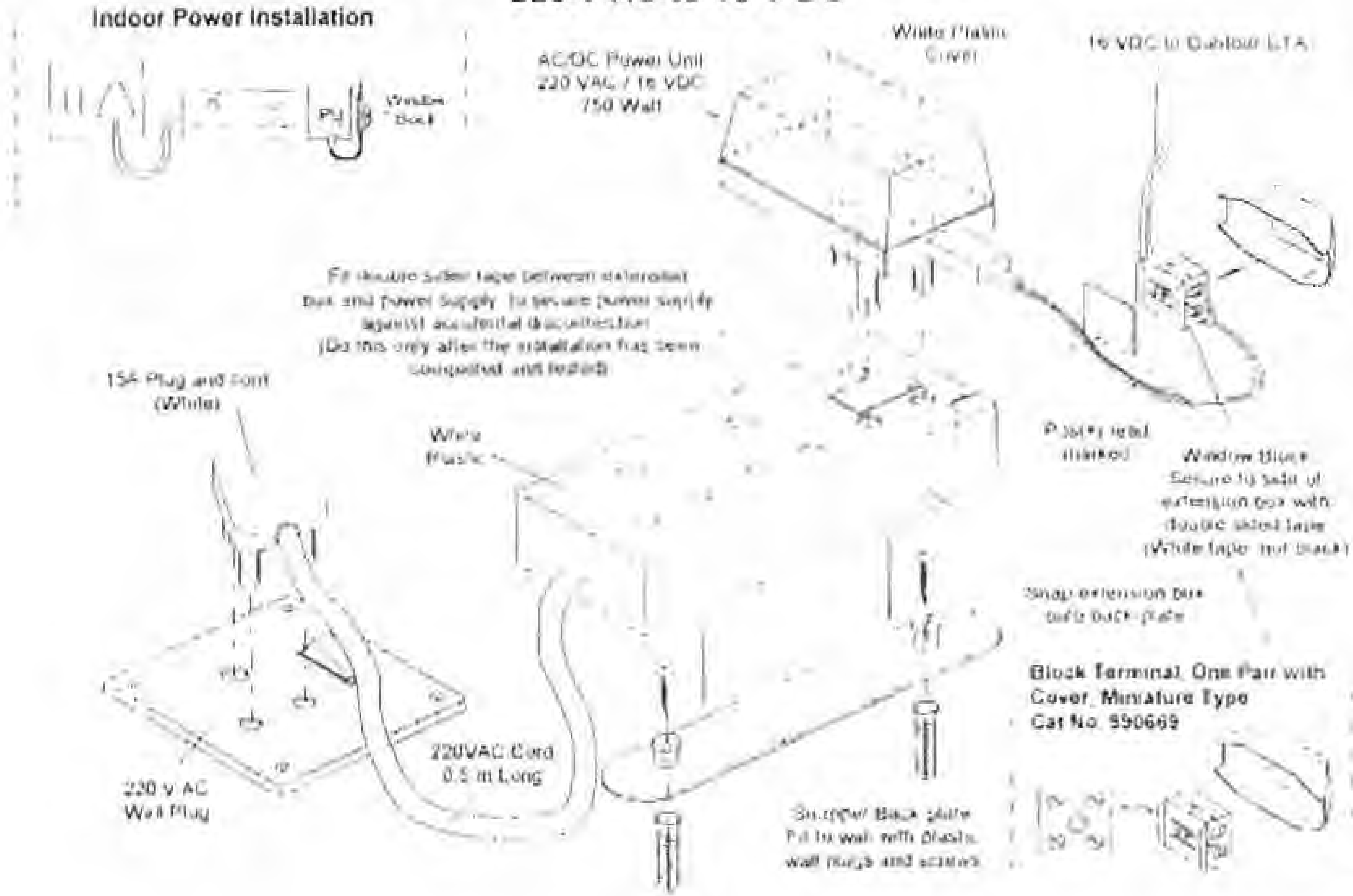
Cable Conductor Size	Distance to CTA
0,19 mm²	18 meters
0,31 mm²	30 meters
0,63 mm²	60 meters

Maximum distance between house and CTA = 60 meters (Then Solar Power)

Installation



DECT/CT2 CTA Installation - AC to DC Power
220 V AC to 16 V DC



DECT/CT2 - CTA Installation
Category - Light / Medium

**Where Possible
1st Option**