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**Integration of Data Services Over Wireless
Indoor Voice Communication Systems**

by

Nazih Almalki, B. ENG.

A thesis submitted to the
Faculty of Graduate Studies and Research
in partial fulfillment of the requirements
for the degree of
Master of Engineering

Ottawa-Carleton Institute for Electrical Engineering
Faculty of Engineering
Department of Systems and Computer Engineering
Carleton University
Ottawa, Ontario, Canada

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Submitted by

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in partial fulfillment of the requirements
for the degree of Master of Engineering.

Thesis Co-Supervisor, S.A. Mahmoud

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Chair, Department of Systems
and Computer Engineering

Carleton University
September, 1996

Abstract

The main objective of this thesis is integration of data communication with an existing voice service over a premises wireless radio system. In order to maximize the information rate over data links, a set of measurements were conducted to gain data on the types of the signal impairments introduced by the wireless channel. Using the results of these experiments, a channel model was developed and verified. A Reed-Solomon error correction algorithm was incorporated and resulted in improvement of the transmission performance of the channel. Different multiple access methods were examined in order to allow many data users to share each speech channel and to increase the capacity of the system for specific input traffic patterns.

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To My Dear Parents

**Integration of Data Services Over Wireless
Indoor Voice Communication Systems**

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The PCI handset will be used in the synchronous¹ mode. The handset has all of the hardware and software required to achieve and maintain a link with a PCI voice network. The PCI network has three main components:

- 1) base stations that communicate with the handsets,
- 2) a controller to locate users as well as performing all the necessary actions required by the PCI standard to maintain the link with the users, and
- 3) a PBX (Private Branch eXchange) which allows calls from the PSTN (Public Switch Telephone Network) to be connected with users of the PCI network.

All of the links components are commercially available, except for the converter / multiplexer box. The system design of the converter box will be described later in this thesis.

1.2.1 System Issues

The following provides a summary of the issues that need to be investigated in order to implement the data transmission service. These issues are: bit error distribution, interface between the PDE and the voice network, base station effect on pure data transmission, signal path through the controller and the PBX, and the effect of diversity and mute on the data link. These issues are explained in detail below.

1.2.1.1 Bit Error Distribution in the Cordless Channel

The bit error distribution will give insight into how the wireless channel will distort the data. This will enable the selection of a suitable data protection scheme. It will also help in determining if packet / frame interleaving is required before Forward Error Correction (FEC) is applied. Finally, it will enable the determination of the achievable data transmission rate over the voice channel.

1. Putting the hand set in the synchronous mode will allow for data to be taken from the interface port of the hand set and be transmitted inside of the voice data: thus feeding data to system as voice data.

DTE	- Data Terminal Equipment
FCC	- Frequency Control Commission
FEC	- Forward Error Correction
FFEC	- Frame Forward Error Correction
FIFO	- First-In-First-Out
GF	- Galois Field
GFSK	- Gaussian Shaped Frequency Shift Keying
IR	- Infrared
IRDA	- Infrared Data Association
kbps	- kilo-bits per second
kHz	- kilo-hertz, 1000 cycles per second
LAPR	- Link Access Protocol For Radio
LPI	- Landline Physical Interface
NAck	- Negative Acknowledgment
PAD	- Packet Assemble/Disassembler
PBX	- Private Branch exchange
PCI	- Personal Communication Interface Standard
PCM	- Pulse Coded Modulation
PCS	- Personal Communication Services
PDE	- Personal Data Equipment
PSTN	- Public Switch Telephone Network
PRMA	- Packet Reservation Multiple Access
RFU	- Reserved for Future Use
Rx	- Receive / Receiver
R-S	- Reed-Solomon

RSSI	- Received Signal Strength Indicator
SNR	- Signal to Noise Ratio
TDD	- Time Division Duplexing
TDMA	- Time Division Multiple Access
Tx	- Transmit / Transmitter

1

Introduction

1.1 Objectives and Motivation

The demand for wireless digital data networks is on the increase, and the resulting economics of scale are likely to lead to reductions in cost of transmission per unit of data. The new ATM (Asynchronous Transfer Mode) wireless LAN (Local Area Networks) is one of the important advances in the wireless data networks field. However, the cost of such a system is still high for most users, especially when the total bandwidth required is relatively moderate. As well, the complexity of the wireless system cannot be justified in economic terms.

As a relatively low cost solution, it may be feasible for the latest generation of wireless voice networks, such as PCI (Personal Communication Interface)¹ [2], to permit the integration of the data communication services with voice. These unlicensed band etiquette systems have the main advantage of being cost efficient and reliable from the network point view. Furthermore, they already use digital data transmission methods for voice communications.

The advantage of using PCI wireless digital voice network is that it already provides end user to end user interface (voice calls), as well as multi-user to server access (voice

1. The PCI standard operates in the unlicensed band from 1920-1930 MHz, and is subject to the spectrum etiquette rules defined by FCC part 15d.

mail server access). The networks also provide high voice quality based on relatively low Bit Error Rate (BER) performance.

The existing wireless voice networks have some limitations. First, several networks are already in operation, and the introduction of data transmission must be done with minimum disruption. Second, the networks are designed for digital speech communication which can tolerate relatively high bit error rate (BER). Such high BER is unacceptable for data transmission. To combat high BER, an error control mechanism must be employed.

Improvement of BER requires full understanding of the channel behavior. Data protection or correction schemes can then be incorporated to gain acceptable data transmission performance.

1.2 Problem Description

The problem under investigation in this research work concerns the transmission of data messages over wireless networks that have been designed primarily for digital voice transmission. The transmission of data should not reduce the total network capacity, nor affect the quality of the voice calls. Also, the integration of data transmission services should be realized with minimum changes to the hardware and the software of the existing system.

Once the transmission of data can be achieved transparent to the existing voice system, wireless data connections will be introduced as new service. This will also allow for wireless access to file servers, databases, and personal computers, as shown in Figure 1-1.

Figure 1-1 provides an example in which a data link will enable a user with a PDE (Personal Data Equipment) with an IR (Infrared) port to connect to a multiplexer / converter box. The box will be attached to a PCI handset. Then the handset will be used as the transmitter of data over the PCI channel. Thus, a link between the handset and an PCI network via base stations will be achieved. At the receiving end of the link, another handset is used as a receiver. This handset is connected to a file server through a PCI protocol data

to a computer readable data converter.

Figure 1-1: A configuration for the proposed system for the integration of the data and voice service

The multiplexer box will perform two actions. First, the box will convert the IR data signal and protocol to the PCI protocol and vice versa. Second, it will multiplex many data user signals onto one speech channel.

The PCI handset will be used in the synchronous¹ mode. The handset has all of the hardware and software required to achieve and maintain a link with a PCI voice network. The PCI network has three main components:

- 1) base stations that communicate with the handsets,
- 2) a controller to locate users as well as performing all the necessary actions required by the PCI standard to maintain the link with the users, and
- 3) a PBX (Private Branch eXchange) which allows calls from the PSTN (Public Switch Telephone Network) to be connected with users of the PCI network.

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The bit error distribution will give insight into how the wireless channel will distort the data. This will enable the selection of a suitable data protection scheme. It will also help in determining if packet / frame interleaving is required before Forward Error Correction (FEC) is applied. Finally, it will enable the determination of the achievable data transmission rate over the voice channel.

1. Putting the hand set in the synchronous mode will allow for data to be taken from the interface port of the hand set and be transmitted inside of the voice data: thus feeding data to system as voice data.

To find this distribution, a set of measurements on a data link has been performed. Measurement of BER versus power level has been done by transmitting a sequence of data words through the handset and then collecting the received data at the base station over the uplink. For the downlink, the same set up was used, except that the base station was the transmitting source and the handset was the receiving end. The measurements results confirm the well-known results that the channel experiences Rayleigh fades.

1.2.1.2 Interface of IRDA Protocol and the Cordless Protocol

One way of interfacing between a PDE (Personal Data Equipment) and the voice system is by using PDE with an IR (Infrared) port that complies with the IRDA specifications, thereby allowing a convertor to be designed (or purchased) to meet the IRDA and the cordless specifications. This convertor box has to allow a signal to be converted for IRDA to cordless and vice versa. The problem with such a system is that the IRDA link can only operate over a maximum distance of one (1) meter, and in line-of-sight. This short distance and line of sight requirement can be a problem when the system is used, for example, in a class room location. For such situations, an alternative method has to be provided.

1.2.1.3 Base Station Effect on Pure Data Transmission

The voice data is compressed using ADPCM, and in the base station an ADPCM to PCM conversion is performed. This change is done so that the voice data is in the same format as the Public Switch. This change from ADPCM to PCM and back to ADPCM should have no effect on the data link. In the event that it does, a software change to disable the ADPCM-PCM translator in the base station will be required.

1.2.1.4 Signal Through the Controller and PBX

There exists two types of controllers for the voice network. One type allows for the digital data of the speech channel to go through the controller without having to go through a DAC (Digital to Analog Converter) and then through an ADC (Analog to Digital Converter). In the second type, the digital data will be converted to analog signal by using a DAC. This conversion will be done on the signal between the controller and the PBX. Then, the signal is transferred back to a digital data by using an ADC from PBX to the controller. Quantization noise associated with these transfers could cause distortion to the data, and this will mean that the speech channel cannot be used for data communication. To resolve this problem, the controller could be programmed via the handsets to by-pass the PBX if the call is in the same network (which is the case in most situations).

1.2.1.5 Diversity Effect on the Data Transmission

In many practical implementations, the voice network will probably employ switching diversity. This diversity can have a positive impact and / or a negative one. The effect will not be known until a data link is set up and tested. It will be shown in chapter 5 that switching diversity, when implemented, does have a negative impact on the data in some cases and a positive effect in other cases.

1.2.1.6 Mute Function Effect on the Data Transmission

The mute function which may be implemented on the voice channel will possibly have some effect on the BER. This will be determined by testing a working data link during a mute period. The experimental results will show in chapter 5 that the presence of a mute function increases the BER.

1.3 Methodology

Solution of the stated problem is based on the following methodology:

- 1) A background research on the latest methods of improving the channel as well as multiplexing many data users signals is surveyed.
- 2) An experiment is designed in which a data link is constructed and measurements are conducted to determine the impairment effects of the channel.
- 3) The results of testing the data link are then analyzed in order to arrive at suitable channel model and error control scheme.
- 4) Recommendations with regard to the entire integrated system architecture are then made.

1.4 Thesis Organization

Chapter 2 provides the background information on the wireless communication system. The channel multipath effect, the etiquette (PCI) protocol, Reed-Solomon forward error correction codes, and the different multiple access techniques are surveyed. Chapter 3 presents simulation results of the PRMA (Packet Reservation Multiple Access) scheme.

Chapter 4 describes the experimental set up and the equipment used. Chapter 5 describes the characteristics of the channel by analysing the results of the experiment.

Chapter 6 gives an analytical solution for predicting the BER of the channel and the use of Reed-Solomon codes to improve the reliability of the channel. It will also compare the analytical results to the data obtained through the experimental measurements.

Chapter 7 describes a recommendation for implementing a data link. Chapter 8 will present the complete results of the measurements and analysis of the hybrid access scheme, PRMA (Packet Reservation Multiple Access). It will also present recommendations for future research work.

Background and Literature Survey

2.1 Introduction

The purpose of the literature survey is to examine the state of the technology required for adding data communication to wireless voice networks, and allowing a speech channel to be shared among many data users.

2.2 Wireless Communication Systems

Personal communication services (PCS) are experiencing a very high and rapid growth. To try to keep up with this growth, systems that are based on microcellular indoor wireless communications are becoming one of the most important research areas of this and the last few decades, and the indications are that it will remain so in the future. Although the systems being developed are ambitious, they raise many questions. PCS will provide many types of services ranging from telephony to video conferencing in addition to providing high speed Wide Area Network (like the Internet) connections and Local Area Networks. New services such as paging and fax are being introduced for the indoor wireless users (example DECT). With such diverse services that differ in their requirements, a classification method is required. One can classify them into narrowband or low rate services, or wideband or high bit rate services. This thesis is interested in the narrowband data service.

2.3 Multipath Channel And Delay Spread

The multipath phenomenon [25] of the transmission channel is always of a great concern when designing wireless systems. This is due to the fact that this kind of channel will cause signal dispersion. This is made clear in the following:

If the transmitted signal is:

$$s(t) = \Re\{u(t)e^{j2\pi f_c t}\} \quad (1)$$

Then because of multipath the received signal is:

$$r(t) = \sum_n \alpha_n(t)s[t - \tau_n(t)] \quad (2)$$

where $\alpha_n(t)$ and $\tau_n(t)$ are the attenuation factor and the propagation delay for n^{th} path respectively. Thus from equation (1) and equation (2) the channel model is:

$$C(\tau, t) = \sum_n \alpha_n(t)\delta[\tau - \tau_n(t)] \quad (3)$$

In which $C(\tau, t)$ is called the power delay profile. And delay spread, τ_{rms} , is often used as a measure of this power delay profile, and could be obtained from the power delay profile, for a given time t , as

$$\tau_{\text{rms}} = \frac{\sum_n \tau_n^2 C(\tau_n)}{\sum_n C(\tau_n)} - \left(\frac{\sum_n \tau_n C(\tau_n)}{\sum_n C(\tau_n)} \right)^2 \quad (4)$$

The values of τ_{rms} are usually in the range of 15 to 100 nsec for indoor environments, with a typical values for a large room of 50 nsec and 30 nsec for a small one [3].

Another important parameter related to delay spread is: the coherence bandwidth, which is defined as the range of frequencies that experience statistically dependent distortion. This is the reciprocal of the multipath spread.

$$B_c = \frac{1}{2\pi\tau} \quad (5)$$

This bandwidth will be a limit on the transmission rate, $R = 1/T$, in which T is a bit or symbol period such that:

$$R \leq \pi B_c \quad (6)$$

The signal bandwidth W compared to the coherence bandwidth B_c will determine if the channel has a flat fading characterization ($W \ll B_c$) or a frequency selective ($W \gg B_c$).

Since α_n and τ_n are time dependent, the time variations of the channel, i.e $C(\tau, t)$ change for different values of t . By using a similar argument as for B_c , we can define the coherence time T_c and the doppler Spread, B_D , where the doppler spread is due to moving object(s) in the channel. B_D and T_c are related by

$$B_D = \frac{1}{2\pi T_c} \quad (7)$$

The most commonly used, and simplest channels, are channels where $T_c \gg 1/R$, those are known as slow fading channels. If T_c is much smaller than $1/R$, the channel is characterized as a fast fading one.

Systems with high data rates usually go through a channel that is slow, frequency selective, and fading. Techniques such as commutation signalling and multi-tone signalling, where the symbol alphabets have been increased are used to combat the effect of multipath. Diversity techniques also exploit the multipath effect to provide the receiver with statistically independent (or at best not very correlated) replicas of the same information signal. Combining these signals reduces the probability of the signal fading below some receiver threshold [3].

2.4 PCI Protocol

The PCI (Personal Communication Interface) protocol [2] is based on the CT2 standard with some modifications required to follow the FCC Part 15 subpart D "etiquette" for sharing the spectrum in the unlicensed band. Compatibility or interoperability is not maintained between equipment operating according to this specification and equipment following I-ETS 300 131 (CT2).

The air interface specification, for the PCI standard, is divided into the following sections:

- 1) 2.4.1 Physical radio interface and channel management
- 2) 2.4.2 Communication layer 1
- 3) 2.4.3 Communication layer 2
- 4) 2.4.4 Communication layer 3 and
- 5) 2.4.5 Speech coding and transmission

Some of the relevant material for this thesis will be presented here, if more information is required please refer to [2].

2.4.1 Physical Radio Interface And Channel Management

The operating frequencies is in the band 1920 to 1930 MHz, and the channel center frequencies are given by:

$$1920.0 + 0.1n \text{ MHz, where } n \text{ is an integer between } 1 \text{ and } 99 \quad (8)$$

With a frequency accuracy of ± 10 KHz maximum difference between nominal equation (8) and actual channel center frequencies.

The equipment can employ a dynamic channel allocation strategy when assigning a channel to a user. The dynamic channel allocation will use a channel to monitor, which

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requires that the CTA (Cordless Telephone Apparatus'), monitor the signal strength during at least their intended transmit frequency and time interval on the channel for at least 10msec before selecting that channel for use as a marker channel or as a traffic channel. The channel will be considered free if:

- a) the channel has been monitored below a maximum of -94 dBm (the lower threshold); or
- b) the channel has been monitored above the lower threshold and at least 40 of these have been monitored within the preceding 10 seconds.

Then the channel which has the lowest strength of all channels and does not exceed the upper threshold of -74 dBm can be selected. This may exclude any channel on which interference is known to exist. If no channel is monitored below the upper threshold then no free channel is available. For other rare cases please see [2]. After the marker channel has been selected by the above, if interference is encountered or anticipated, the marker channel may be moved to another free channel.

The minimum RF power to be transmitted is +12 dBm. An adaptive transmitter power strategy can be used to reduce the near-far effect. The receiver radio shall have a sensitivity such that the BER is 1 in 1000 or better at a received power level of -94 dBm. The transmitter employs a GFSK (Gaussian Shaped Frequency Shift Keying) modulation scheme.

2.4.2 Signaling Layer 1

Signalling layer 1 specifies methods for the following:

- 1) Time Division Duplexing (TDD);
- 2) The multiplexing of data onto the TDD structure;
- 3) Two-way digital link initiation;
- 4) Handshaking

Two different burst structures (MUX1 and MUX2) are used. These specify the use of

a 2 msec transmission burst at 72 KBit/s with a length during speech communication mode of 68 bits or 66 bits during signalling communication mode. This is repeated every 144 bits (2 milliseconds) with an interleaved receiving window (Figure 2-1) of up to 68 bits. Since the speech channel (B channel) occupies 64 of the bits in a burst, systems can signal during speech communication mode by using the remaining 2 bits in the frame. Eight or two bits of guard time remain per burst frame for the burst lengths of 68 and 66 respectively.

With the TDD format the transmit bit rate is 32 kBit/s, since the CTA transmits for one slot of 1 msec and receives in the next 1 msec time slot.

Multiplex 1 (MUX1), illustrated in Figure 2-1, is invoked from MUX 2 by means of layer 3 channel control message. MUX 1 is used bi-directional over an established link to carry the D (signalling channel) and B channel (i.e voice).

Figure 2-1: Multiplex 1

Multiplex 2 (MUX 2), illustrated in Figure 2-2, is used to carry the D and SYN channels for link establishment, signaling, and re-establishment. The B channel is non-existent in MUX 2. MUX 2 is used for call set up, hand over and other system functions, prior to a switch to MUX 1 by means of layer 3 message.

For Call detection, link set-up and establishment, and ID handshaking please see [2].

Figure 2-2: Multiplex 2

2.4.3 Signaling Layer 2

This is used for maintenance of the channel and has little effect on data links.

2.4.4 Signaling Layer 3

This is used for maintenance of the channel and has little effect on data links.

2.4.5 Speech Coding and Transmission

The voice data is coded using an ADPCM, (Adaptive Differential Pulse Code Modulation), voice codec algorithm. This codec is specified in ITU-T Recommendation G.726. The codec transformation of data is fully reversible if no changes to ADPCM or PCM codes take place. The rest of the specification for speech coding and transmission is there to maintain a good voice quality.

2.5 Reed-Solomon Codes

Reed-Solomon (R-S) codes are forward error control (FEC) codes. They are a special class of BCH codes which are a class of multiple-error-correcting codes with a cyclic property. Reed-Solomon codes and BCH codes are usually classified as nonlinear block codes.

A Reed-Solomon code is described as a $(N, K; p)$ code, where N is the number of m -bit

symbols in a code word, K is the number of information message symbols and p is the number of symbols introduced for error detection. For example, a R-S code of $(255, k, 3)$ with 8 bit symbols has m equal to 8 bits per symbol; thus 255 symbols per code word ($N=2^m-1=255$) and k symbols out of 255 are information symbols. The code could be shortened to $(n, k; p)$ code where $n=N-i$ and $k=K-i$ by setting the i most significant code words (i.e., $m \cdot i$ bits) to zero.

The minimum distance for Reed-Solomon decoder is equal to d ,

$$d = N - k - p \quad (9)$$

The Reed-Solomon code can correct up to t symbols.

$$t = \text{Int}\left(\frac{d}{2}\right) \quad (10)$$

For the case of shortened code, $(63, k, 3)$ $n = 63$ and $i = 192$ symbols (or 1536 bits) and the number of correctable symbols is $(60-k)/2$, if $k = 40$, then $d = 10$ and $t = 5$ symbols.

Reed-Solomon codes use polynomials in the Galois Field (for 8 bits a GF(256) is used), which are usually extensions fields of the GF(2), that are constructed from a primitive polynomial. Since a code word containing all zeroes is a valid code word, the parity symbols of the code word are usually inverted.

In order to find out if a received codeword is correct, Reed-Solomon codes are usually placed in a frame so that an ARQ (Automatic Retransmit reQuest) can be used in case the received code word is wrong and is required to be retransmitted. Thus, the framing and FEC check the parity symbol by computing the syndrome over the code word. If the syndrome is zero, the code word contains no detectable errors. If the parity symbol indicate that the code word is in error (i.e it has a non-zero syndrome), then the framing may either apply the error correction algorithm to the code word or leave it unaltered. To reduce the number of the decoder errors, if the syndrome indicates that more than t (t is max. number of correctable symbols) symbols are in error, the code word will be declared uncorrectable and no change will be made to the code word, and a request for retransmission will be placed.

2.5.1 Erasure Decoding

The BCH decoding algorithm [21] (including R-S decoder) does not generalize to utilize demodulator soft decisions in a direct fashion. The only approaches that have been used require multiple application of the hard decision decoding algorithm. However, in the special case of demodulator erasures, an algebraic decoding approach is possible. That is, if the code is defined over the field $GF(q)$, $q = 2^k$, k is number of bit in one symbols, then the demodulator is assumed to have the normal q outputs plus a $(q+1)$ th output corresponding to an erasure. The erasure output is assumed to contain absolutely no information about which of the q possible symbols is actually transmitted. This decoding algorithm does not provide significant additional gain on the Gaussian channel, but it is of benefit, on certain interference channels where the interference may be detected and treated as erasures.

Now, if a code has minimum distance d and a received word contains e erasures, then the e "erased" positions can be ignored in the decoding process. However, even ignoring these positions, all code words will differ in at least $d - e$ of the remaining position allowing the correction of up to

$$t_m = \left\lfloor \frac{d - e - 1}{2} \right\rfloor \quad (11)$$

channel errors in addition to the erasures. Thus, all combinations of v channel errors and e erasures are correctable providing that

$$2v + e < d \quad (12)$$

For the complete description of the erasure and error correction with BCH codes, please refer to [21].

2.6 Multiple Access Methods

When there is a requirement for channel to be shared between multiple users or portables, an access method is required. There are many different multiple access strategies. These access methods comply to a standard that has the following: first, fair

opportunities for users to access the channel; second, very low interference from one signal to another; third, a clear and defined way for the base station and portable to communicate, i.e a protocol. The multiple access techniques can be divided into three major areas as follows:

2.6.1 Fixed Assignment Techniques

In these methods a slice of channel is taken in time, frequency, or codes, and given to each active user: thus every user has a fixed portion of the shared channel. This makes the channel capacity fixed, and it is equal to the sum of all incoming traffic. Incoming traffic is the traffic that every active user is allowed to send or receive in the assigned fixed portion of the channel.

System capacity is the total allowable number of users in the system: both active users and inactive users. The maximum allowable active users in a system is the channel capacity. Therefore, the system capacity is usually bigger than the channel capacity. This is due to the fact that not every user is trying to access the channel at the same time. But it is still the channel capacity which limits the maximum number of users at a given time. These type of access methods are designed for real time application, where access time is somewhat known, and also where virtual connection is required.

2.6.1.1 Frequency Division Multiple Access - FDMA

In this method, the frequency axis is divided into N center frequencies. Each frequency is used by one user or portable. Usually after some distance the frequencies could be reused again. The duplexing could be done in two ways. One way is to have two sets of frequencies, one set for transmitting and one set for receiving. The other way of implementing full-duplex is to divide the time on the used frequency: one time slot for transmission, and other for reception. If the allocated bandwidth for the channel is much wider than the coherence bandwidth, a Multiple Frequency Diversity (MFD) [4] scheme could be used to compact fading.

2.6.1.2 Time Division Multiple Access - TDMA

In TDMA [3] several users time-share a common carrier frequency to communicate with the base station. Each user is allocated one or more time slots within a frame in the downstream and upstream directions. In these slots users transmit digitized speech or other digital data, with bit rate lower than the frame bit rate. In the downstream direction, the base station broadcasts to the active users in a TDM or FDM format. In the upstream direction, each active user's terminal transmits to the base station only in its own assigned slot. Inter-user interference is prevented by strict adherence to time slot schedules, and by guard times and time-alignment procedures between upstream time slots. This is done in order to prevent overlaps due to different propagation times.

2.6.1.3 Code Division Multiple Access - CDMA

The two basic problems which the mobile radio system designer is faced with are, multipath fading of the radio link, and interference from other users. Spread spectrum signals are effective in mitigating multipath because of their wide bandwidth. The results of these effects is better access methods.

Thus with direct sequence (DS) spread spectrum, every active user is given a spreading code: hence, code division multiple access. The DS waveforms can be used to either reject multipath returns that fall outside of the correlation interval of the spreading waveform (code), or enhance overall performance by diversity combining multipath returns in a RAKE receiver. Alternately, a frequency hopped (FH) CDMA could be used, where the user code is used to change or hop between a given set of frequencies. Therefore frequency diversity is obtained through coding the data and interleaving it over multiple-hops

2.6.2 Random Access Techniques

In these methods, it is assumed that all users will rarely be transmitting at their peak rates simultaneously. Thus, with a specified delay and blocking probability, one can design a random access channel with a capacity less than the sum of all peak traffic. Each user contends for access to the channel to transmit its information on the available channel. This

leads to collisions: the major disadvantage. The advantages of these protocols are expendability and suitability to irregular loads or traffic (e.g. e-mail).

2.6.2.1 ALOHA

The system uses two channels, one to transmit the data and the other to listen for the acknowledgments (ACK). A user transmits and then listens for ACK. If none is found, then the user assumes that a collision has happened and schedules a retransmission after some random time.

Pure ALOHA, is a scheme where a user can start transmission as soon as data is available. Thus there is virtually no coordination required with the base station. This leads to a great amount of time wasted in collisions as well as retransmissions. Walrand [5] has shown that the maximum throughput is about 18% and occurs when the transmissions are suitably randomized. This low efficiency is due to the low cost of the ALOHA protocol.

Slotted ALOHA, which improves the efficiency to 36%, is an improved version of pure ALOHA. The time axis is broken into slots. Every user may transmit only at the beginning of a slot. This improvement comes at a cost of synchronizing all users with the base stations. ALOHA is one access technique that does benefit from the near/far effect.

2.6.2.2 Carrier Sense Multiple Access-Collision Detected

A more improved version of ALOHA is the CSMA (Carrier Sense Multiple Access-Collision Detected). In CSMA-CD a node must wait for the channel to idle before it can transmit (carrier sense) and it must stop transmitting as soon as it detects a collision (collision detection). p-persistent CSMA, where p is a probability figure, is a scheme in which a node senses the channel when it wants to transmit. If the channel is free the node transmits. If the channel is busy the node waits until the channel is free and then with probability p it transmits; with probability $(1-p)$ it waits for a random time first and then senses again. In a nonpersistent CSMA, when the node finds the channel busy it waits for a random time period and then senses again.

2.6.3 Hybrid

These multiple access schemes try to combine both fixed assignment and random access techniques in one. Reserved ALOHA is one of them, in which the slots are from a TDMA system and the reservation of a slot is done via a slotted ALOHA protocol. PRMA is another hybrid system. PRMA will be explained in chapter 3.

2.7 Conclusion

One way to understand how the channel is working, is to take a set of measurements to discover the fade intervals and fade rates. The other way is to try to model the channel so that an analytical model can be obtained: a model that can predict the results of the measurements. Thus, as described in chapter 4, an experiment can be performed to find out the characteristics of the channel.

From the survey it was also found that there are a few ways that could improve the bit error rate of a fading channel. The one that is most suitable is the Reed-Solomon coder/decoder combined with erasure decoding.

Multiplexers are usually designed to fit one type of data service. Thus if a multiple access scheme is to be designed then a lot of information about the service has to be known. This will make the simulation valid only for that type of the traffic type. If one wishes to design a multiplexer for several types of services, one can use the simulation results of the PRMA multiple access scheme (chapter 3).

3

Multiple Access

3.1 Introduction

The cases for which multiple access schemes will be most suitable will depend on the application served by the system. For example, if the system is used for voice service, then a TDMA or CDMA will result in the most effective access techniques. On the other hand, if the service is a file transfer, or network applications, then a CSMA-CD multiplexer will be more suitable.

This argument above leads to the conclusion that if one is to simulate a multiple access scheme, knowledge about the application or service will be required. These details are required in order to find out some of the very basic parameters such as the rate of arrival, the average call length, the number of users, and also the nature of the link: whether it is bursty or constant.

In case of multiplexer design, the first step is to simulate the multiplexer, and most of the time the implementation of a multiplexer will follow in very easy manner from the simulation results and parameters.

Without these parameters' values, one can only use access techniques that are flexible yet efficient. One technique stands out. A hybrid scheme called Packet Reservation Multiple Access (PRMA). As explained below, the PRMA scheme allows for different applications to share the same channel.

3.2 Packet Reservation Multiple Access - PRMA

With a unified access method it is possible to serve a variety of applications, from personal organizer to traffic light controller, and the one phone number per person idea. PRMA is one of the relatively new access methods considered for such networks. Based on [8] & [11] a presentation of PRMA will be offered in this chapter. Simulation results are quoted and graphs are reproduced from [8] & [11].

TDMA and slotted ALHOA were combined together to give rise to the new method of PRMA. This combines the advantages of both fixed assignment multiple access and contention based strategies, which results in an inherently statistical multiplexing access protocol.

PRMA differs from TDMA in that it makes use of speech gaps, and uses speech activity detector, to improve the bandwidth efficiency of the TDMA. It also differs for R-ALOHA [8] in two ways. First PRMA drops voice packets that have been delayed, while R-ALOHA packets are never dropped and they keep accumulating. Second PRMA is for indoor use where rapid acknowledgments are possible.

3.2.1 The PRMA Protocol

The publications [8] and [11] examine a PRMA technique which is designed for a two-way communication with a star topology. Feedback information from the base station to terminal (down link) is broadcasted to inform the terminal of whose transmission was successful in the previous time slot, and whether a collision has occurred in the upstream (from terminal to base station). It makes use of both Ack and Nack.

The following simulation will focus on the upstream traffic with collision free downstream transmission. Thus, only the performance of the terminals to base station channel is evaluated here.

3.2.1.1 Traffic Types

Periodic and random traffic are distinguished during the packet assembly in each terminal. Speech packets and continuous data streams are considered as periodic traffic, while all other traffic is random traffic. An indicator is used in the control field of the packet header to tell what type of traffic is it.

PRMA allows a terminal to transmit for more than one slot in a frame, if the data rate is higher than that of speech. The frames that make up the upstream are divided into slots.

3.2.1.2 Channel Access

Once the channel has been assigned to the users, then the channel access, for periodic data, is done in two stages. The first stage is the reservation stage, and it starts when the first packet, of a periodic stream of data, arrives at the terminal. In this stage each of the active terminals start to compete for an available slot or frame. If the first trial fails, the terminal will try to retransmit on the next available slot with probability p . Where p is the permission probability. When a successful transmission happens, the base station sends an Ack, to tell the terminal which slot to use, which makes it like a TDMA system. Thus other terminals cannot use these reserved slots until they become free. The second stage is the data transmission stage. It starts in the next frame, and the terminal transmits its periodic data without contention; thus no collision occurs during this stage.

When all of the data packets are sent, the terminal releases the slot by sending the appropriate information to the base station. The base station responds with a negative acknowledgment in the empty slot, and adds it to the available slots list.

Access for random traffic is done differently, the terminal always has to compete for a slot and a success in transmitting the packet does not reserve the slot. Thus, this is similar to the slotted ALOHA.

3.2.1.3 Packet Loss (Dropping)

While a user is contending for unreserved time slots, the terminal holds the packets in a first-in-first-out (FIFO) buffer. If the packets are speech, then the buffer size is limited according to the delay constraint imposed by the network. i.e., when a new speech packet arrives at a full buffer, the oldest packet is dropped. The size of the buffer or the delay is a key factor in determining the sound quality of the delivered speech. On the other hand, for random traffic, the FIFO buffer can be quite long.

With this system all packet losses occur at the beginning of talkspurts, and it has been observed that this "front-end clipping" is less harmful to subjective speech quality than other types of packet loss.

3.2.1.4 Effects Of Network delays

When the network imposes a decrease in the maximum allowed delay, (D_{max}) the packet dropping probability (P_{drop}) increases, thereby increasing the front end clipping, which in turn leads to degradation in the speech quality. P_{drop} is usually used as a measure of the impairment of the front end clipping.

The simulation below uses the maximum number of speech terminals that can share the channel, without probability of packet loss exceeding P_{drop} as a key performance measure for PRMA.

Also, as mentioned before, PRMA uses speech activity detectors. Each speech terminal is equipped with a speech activity detector, to find the silence and talkspurts gaps. The simulation deploys a slow detector for recognizing the talking and principle silent gaps, and a fast detector of minigaps, which are caused by the short silent intervals that punctuate continuous speech.

3.3 System Parameters

The following table, Table 3-1, has a list of the PRMA system parameters with their initial values

Table 3-1: Simulation Parameters

Parameters		Starting Value
R_c	Channel Rate	720 000 bits/sec
R_s	Source Rate	32 000 bits/sec
T	Frame Duration	0.016 sec
H	Overhead	64 bits
D_{max}	Maximum Delay	0.032 sec
p	Permission Probability	0.3
K	Conversations	

The packet or slot size is formed from $R_s T$ bits of information plus H overhead bits, which gives the number of slots per frame as

$$N = \text{int} \left[\frac{R_c T}{R_s T + H} \right] \quad (13)$$

Therefore a packet duration, τ , is defined to be:

$$\tau = \frac{(R_s T + H)}{R_c} = \frac{T}{N} \quad (14)$$

Packets arriving at the terminal are buffered into a FIFO queue. The speech packet is kept at the head of the queue until it is successfully transmitted or it is dropped. A delay counter, C , is used to keep track of the total delay per packet. Therefore, packets will be dropped, when the delay reaches the D_{max} (imposed by the network), or when C is equal to:

$$C = \text{int} \left[\frac{ND_{\max}}{T} \right] \quad (15)$$

C is incremented by one every slot time (T seconds).

Using equation (13) through equation (15) and the initial values provided in Table 3-1, the initial simulation parameters for the PRMA system can be computed. Table 3-2.

Table 3-2: Initial Simulation Values

Parameter	Value
Total number of bits in a slot	= 567 bits/slot
Slot duration, T	= 0.8 msec
Number of frames	= 62.5 frames/sec
Number of slots	= 1250 slot/sec
Number of slots, N	= 20 slots/frame
Delay limit on speech packets	= 40 slots

3.3.1 Performance Parameters of Interest.

The main parameter of performance that the simulation is attempting to establish are the following:

- 1) Number of conversation per channel as a function of total number of channel for a given bandwidth.
- 2) Relationship between the number of simultaneous conversations per channel and the maximum speech delay.
- 3) Frame duration, and the effect of fading interval (section 3) on K .
- 4) System capacity, K , as a function of P_{drop} .
- 5) The effect of p , permission probability, on K .
- 6) P_{drop} versus K , with optimal p .

3.3.2 Simulation Results

With the above parameters in mind, a simulation program of PRMA system was performed to evaluate these performance measures. A file of speech packets was used throughout.

Figure 3-1, presents the results of changing the number of channels and its effect on the number of simultaneous conversations per channel. The number of channels, R_c/R_s , was varied from 2-channel to 30-channel, by changing R_c from 64 kbps to 960 kbps. The number of conversations per channel, S , is the normalized (to that of TDMA) capacity figure.

$K_{0.01}$ is the number of conversations in the system, with $P_{drop} \leq 0.01$. This number, S_{PRMA} , assumes H to be small, and it allows us to compare PRMA to other methods, such as TDMA, which has

$$S_{TDMA} = \frac{R_c}{R_s} = 22.5 \quad (16)$$

simultaneous conversations.

$$S = S_{PRMA} = \frac{K_{0.01}}{\frac{R_c}{R_s}} = \frac{K_{0.01}}{S_{TDMA}} \quad (17)$$

The curve shows that for $R_c/R_s \geq 20$, S is essentially constant. This is due to the fact that for a high number of channels, the statistics of the collisions are dominant over those of the speech, unlike the other regions where the number of channels is relatively small.

Figure 3-1: Conversations per channel (S) as a function of the number of equivalent TDMA channel.

The effect of D_{max} on the number of conversations carried out on each channel is given in Figure 3-2. This is the effect of maximum delay on the system capacity. From the results it can be seen that for a small D_{max} the packet dropping probability increases, causing the efficiency to decrease. While for D_{max} increasing beyond a certain limit (~ 20 msec for this fixed load), then the success of a packet is guaranteed. This means that increasing the buffer size for speech beyond the speech arrival rate does not improve the quality of speech or the performance of the network.

The frame duration (T) effect on the number of conversations on the system, K , can be seen in Figure 3-3. It gives the relation of the fade interval and frame duration, and the effect those two parameters have on the system capacity, or K . The speech data rate determines the frame duration. As depicted in the result, there is an optimal range for T : a range where it makes the system operate under maximum capacity. It also shows that the other ranges will degrade the system performance. That is, if T is very small, a small number of slots will be allowed, thus leading to higher number of collisions and P_{drop} . For T very large, there is very little chance for a speech packet to be successful transmitted before it gets dropped, as given in equation (15). The optimal interval for the graph, is

$$7 < T < 20 \text{ msec}^1.$$

Figure 3-2: Conversations per channel as a function of D_{max}

1. Slow speech activity detector, $T \geq 20 \text{ msec}$ [8]

The most important parameter in the simulation is P_{drop} . It is obtained as the ratio of dropped packets to the total number of packets used in the simulation. The performance measure adopted for this study is the number of simultaneous conversations, K , carried out with P_{drop} less than 0.01.

Figure 3-3: Capacity vs. frame duration (with fast detector.)

Figure 3-4, presents the results of P_{drop} as a function of a number of simultaneous conversations. The results show that the PRMA system can accommodate a maximum of 26 conversations when $P_{drop} \leq 0.01$. That is $K_{0.01} = 26$ and 16% more conversation per channel than TDMA ($S_{PRMA} = 26/22.5 = 1.16$).

Figure 3-4: P_{drop} , Packet dropping probability vs. K , number of simultaneous conversations,

The stability of PRMA occurs in two regions: one region for low congestion and the other when the congestion is high. $p = 0.3$ (as seen in Figure 3-5) corresponds to the optimal permission probability, and the system performance in terms of P_{drop} as a function of K is shown in Figure 3-4. The graph shows that when the threshold probability is increased from 0.005 to 0.01, the system capacity, or the number of simultaneous conversations goes up from 35 ($K_{0.005} = 35$ users) to 37 ($K_{0.01} = 37$ users): a little more than 5% increase in the capacity. This behaviour is similar to that of the ALOHA.

The result, given by Figure 3-5, shows the effect of the permission probability, p , (probability of retransmission in the next slot), on the capacity of the system, K . As seen from the graph, there is an optimal range for p ($0.25 \leq p \leq 0.35$) to be in to allow the system to operate at maximum capacity (efficiency), with the packet dropping probability less than 1%. In this range the system operates at a capacity of 37 ($K_{0.01} = 37$) simultaneous users,

and 64% ($S = 1.64$) more conversation per channel than TDMA. Using the above information for p , Figure 3-1 through Figure 3-4, were the results with $p = 0.3$ (optimal).

Figure 3-5: Maximum number of conversations as a function of permission probability, p , with $P_{drop} \leq 0.01$

3.4 Conclusion

The capacity of PRMA depends on the packet dropping probability. This in turn depends on the maximum delay allowed and the permission probability. Also the length of the frame is significant, for a given number of channels affects the capacity of the system. The above simulation showed that the system will operate at maximum capacity of 37 simultaneous users, with P_{drop} less than 1%, for $D_{max} > 20$ msec, $p = 0.3$ and a frame duration of 15 msec for 20 or more channels.

PRMA will be one of the most examined scheme (especially for wideband applications), and may be used as an access scheme for indoor wireless system. This is due to its ability to offer both real-time services such as voice and video, and non-real time services like file transfers and on-line help (where delays are allowed). But the success of PRMA will also depend on whether its performance will be affected by the presence of other types of traffic. The simulation was for voice traffic only (no video data, where the bit rate is higher).

4

Wireless Indoor Data link Experiment

4.1 Objective of The Experiment

The performed experiment determined the characteristics of the channel, and the impairments of the channel that cause data bits to be in error. Thus it give the relationship between the received power and the BER. From there one can tell if the data transmission is a valid option to add to the existing voice.

In deciding which channel is usable for data transmission, the BER distribution will govern the type of data protection and correction scheme required. It will also determine the available information data rate: in other words, how much of the transmission bit rate will be taken as overhead for the protection/correction scheme of the data links.

4.2 The Locations and Cases for the Measurements

The measurements were done for four (4) different cases as well as four different building types. The four cases were;

- a) transmit on the uplink (handset and receive at the base station) with the antenna diversity on;

- b) same as a) but with antenna diversity off;
- c) using the downlink (transmit from the base station and receive at the handset) with diversity on; and
- d) diversity off for the downlink.

The handset was put in the synchronous mode, so that data could be sent and received through the handset via the serial connection of the handset. At the base station the data was sent via an access to the radio data input and output. Then by using a converter box (Appendix B) the base station data format was changed to one that is compatible with that used in the computer. The data was then collected at the receiving end, and compared to the transmitted data.

The four different building locations where the measurements were taken, were:

- 1) Open Office Layout (OFL) location, with floor size of about 200 meters by 200 meters
- 2) Carleton University: Minto Case building, 4th floor (new research building)
- 3) Carleton University: Mackenzie Building, 4th floor (thick, brick walls)
- 4) Carleton University: Annex, 5th floor of the Architecture building ("false" walls layout)

These locations were chosen because most types of building, where such system will be used, are represented in the four locations.

4.3 The Equipment and Layout of the Experiment

The equipment that was used, were:

- 1) Compaq 286 P.C., Serial #: 4649AK7B0350
- 2) HP machine 286, Serial #: 2742A00622
- 3) 2 AST synchronous Cards
- 4) PCI handset
- 5) PCI base station

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- 6) Toshiba LapTop with base station control s/w to turn diversity OFF/ON
- 7) Power supply (+12V & - 12V)
- 8) Bits, synchronous card controller and file transmitter / receiver
- 9) Bit comparator and bit error counter software
- 10) Base station to synchronous card adapter (Appendix B)

4.4 Method of Performing the Experiment

The equipment in Figure 4-1 was used in the experiment. For measurements of the uplink, a voice link between the handset and the base station was setup. Then using the computer that is connected to the base station as the receiving end, the data from the base station was translated using a transformer box (10). This computer was set-up to start collecting data once a predefined synchronization sequence was received. After that, the data file was transmitted from the handset-connected-computer, via the voltage translator to the handset, to the base station. The power reading off the handset was recorded after a few seconds. After the complete file was transferred, the receiving computer was stopped, and the collected data was compared to the transmitted one.

The above procedure was for the case of the uplink with diversity on. In case of diversity off, the diversity in base station was turned off using the base station control software. Then the transfer of the file took place.

For the downlink, the same steps as for the uplink took place, but instead of sending the file from the handset-computer to the base station computer. It was sent from the base station-computer. Thus the handset-computer was the receiving computer.

Figure 4-1: The Experiment Setup

For every location, four sets of measurements were taken. For each measurement the 2 Mbits file was transmitted and received. The file was made up from random data. Also the measurements were done for different received power values.

To achieve the different power levels the handset was moved from one location to another. The handset position was fixed for the duration of that measurement. For every location the power reading of the handset was recorded, and this was used as the received power for the BER calculations. During each measurement, movement of people was observed to ensure that there were people moving between and around the base station and the handset.

Using an error counter software, the bit error rate was calculated, then the file was transferred to a spare station where it was analysed for error distribution, and where the different error control algorithms were applied. Also the collected data was used as a bench mark for the analytical analysis of the BER.

The error counter software compares the received data to the transmitted data, and counts the number of bit(s) that are not in agreement.

5

Results of The Experiment

5.1 Introduction

The following results were obtained by carrying out the experiment described in chapter 4. The effect of the channel on the data can be found by comparing the received data to the transmitted data. This comparison will also show the bit error distribution. Depending on this distribution of error, one can determine if the Reed-Solomon code, for data protection and correction, is a good scheme to use.

5.2 The Measurements Results

There are two links in the current voice channel; one link is the transmission of data from the handset to base station, and the other is reception from base station to handset. We consider a system in which the base station employs an antenna diversity scheme.

Four cases are then considered in the measurements:

1. Diversity OFF, Handset to base station, Up link
2. Diversity ON, Handset to base station, Up link
3. Diversity OFF, base station to Handset, Down link
4. Diversity ON, base station to Handset, Down link

5.2.1 The Channel (No Coding)

The Bit Error Rate (BER) for each measurement was calculated by combining every 5 (five) close measurements together; for example: measurements with received power equal to -78 dBm to -82 dBm were combined to give an average at -80 dBm. Figure 5-1 was plotted, the table of results can be found in Appendix A.

U/L OFF: Handset to base station with diversity OFF
U/L ON: Handset to base station with diversity ON
D/L ON: base station to handset with diversity ON
D/L OFF: base station to handset with diversity OFF

Figure 5-1: The Raw BER Results

The uses of antenna diversity will improve the channel performance. This improvement occurs because of higher received signal to noise ratio (SNR) and thus a better BER. The better SNR comes from choosing the antenna with the better RSSI (Received Signal Strength Indicator). To achieve higher RSSI the base station checks the currently used antenna to see if it is experiencing a fade. Then, if the current antenna is in a fade, the base station will check to see if the other antenna is not experiencing a fade, i.e. has a higher RSSI. In case the other antenna has a better RSSI then the base station will switch to the better antenna. This kind of antenna diversity is called selection diversity: sometimes it is called switching diversity.

From Figure 5-1 we see that the cost of switching diversity is very high for uplink, while there is no diversity cost for the downlink. Hence, diversity improves the downlink by few (2 to 5)dB.

The cost of diversity for the uplink is due to fact that the base station switches between the different antennas during the reception period from the handset (at beginning and/or at end of the reception interval), while it transmits only on one antenna (no switching during transmission on the downlink). This explains why the measurements differ for the uplink and downlink. The high cost of diversity for the uplink comes from these switches in the base station. When a switch of antennas occurs, the base station does not wait for co-phasing or synchronization between the two antennas before it switches. This leads to loss of bits (2 to 3) whenever there is a switch.

It is also clear that the down link is much better than the up link. This is due to the mute function in base station that was not disabled. While in the handset it was disabled. The mute function job is to mute the voice signal when the voice link experiences deep fades. This prevents the users from hearing noises, and interference. This function, when activated, will set all bits to zero for a complete received PCI frame whenever a deep fade occurs. This leads to extending the fade intervals to a constant multiple of 32 bits frames.

Now, since the channel is symmetric, i.e. the downlink and uplink are very similar, the results for the downlink can be applied for the uplink for cases where diversity is on and mute is off, and the cases where diversity is off and mute is off.

The diversity does improve the downlink where the data is not encoded. This is because there is no switching of antenna during the transmission period, since the coherence time for these type of channels is bigger than one TDD frame. Then the switch in the reception period will improve the channel during the transmission period by reducing some of the fade intervals.

Diversity also improves the uplink where the mute function is active, and this is because whenever the base station switches between antennas, the mute period gets stopped. This happens when it switches to achieve a better signal to noise ratio, which means it switches to the non-fading antenna. On the other hand, when mute is off, the diversity increases the bit error rate, and this can be seen in Figure 5-2, which shows the high cost of diversity.

Figure 5-2 shows that the PCI wireless channel experiences burst error. Bursts that have 2 to 3 bits per PCI frame in error, indicates that the base station is switching antenna due to lower RSSI which occurs even at high received power (-38 dBm) as well as at low power (-60 dBm). This type of distribution indicates that the channel is experiencing Rayleigh fades.

Figure 5-2: Cost of diversity,
a) Received Power = -60dBm b) Received Power = -38 dBm

5.2.1.1 Carleton's Layout Versus Open Office Layout Measurements

A comparison between locations was done. And since all of the Carleton locations were different from those of the open office layout, it is important to look at the differences and try to explain them.

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As shown in Figure 5-3, the measurements' results for the case of handset to base

Figure 5-3: Handset to base station with diversity off Carleton Vs. Open Office

station with diversity off, this shows that Carleton's layout and the open office's layout results were very close to each other. Allowing for the combining of the results from all locations one obtains a better confidence interval, as will be seen later. Also the results show that the BER is independent of the location, and only dependent on the received power level. This is expected since the receiver radios' hardware performance does not depend on its location as long as it received the required power.

It was noticed that for different locations the coverage area differs, and this is due to the path loss, which comes from the distance and the material that the signal has to go through; it is harder to go through brick walls than to go through fibre ones. In an open office layout one can expect to have a better coverage area than in a location similar to that of Carleton's.

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Note, as seen in Figure 5-3, there are variations between -78 dBm to -68 dBm region. These variation are due to the power control algorithm, which, when active, will cause a lot of bits to be in error (usually several PCI frames).

5.2.2 The Coded Channel

The transmitted signal (data) was assumed to be coded using Reed-Solomon code words. This was done to see if Reed-Solomon codes work for such a channel. Reed-Solomon codes were chosen due to the fact they are the most powerful codes to protect data in a fading environment, and the channel in this analysis does experience fades. The power of Reed-Solomon codes is that a Reed-Solomon code's decoder works on the basis that it corrects up to a certain number of symbols (not bits). In this case, where a (63,40,3) code was used, the Reed-Solomon decoder can correct up to 10 symbols out of the 60 symbols code word.

The code words were made up from 8 (eight) bits symbols, which leads to 480 bits code words, and 512 bits frames¹.

When looking at the data and especially at the number of bits in errors per symbol, it was noticed that most of symbols had less than 3 error/symbol, leading to an analysis of the data using 4 bits/ symbols (256 bits per frame). In most cases 4 bits code actually improved the BER, and in others it had no improvement or a very small degradation in BER. The 4 bits code will reduce the delay caused by retransmission, and also reduce the complexity of the decoder. This leads to less power usage for coding and decoding, giving rise to longer battery life, and a smaller size of decoder chip.

1. 1 symbol for synchronization and 3 (three) for error detection. This means, if a frame is required to be retransmitted using ARQ (Automatic Repeat reQuest), the resultant delay will be 32 ms (the data rate is 16 kbps for Rx, and 16 kbps for Tx, and since an error cannot be detected until the complete frame has been received, then a request to retransmit will be made after the 32 ms from the start of transmission

Figure 5-4: BER after Reed-Solomon decoder for a 8 & 6 bit /symbol codes

There is a problem with a 4 bits symbol frames: there are only 15 code symbols to be used, and thus a 60 symbol frame will be impossible. The latter problem led to an analysis of the measurements' values using a 6 bits symbols code. This will have a higher retransmission delay than the 4 bits, but still smaller retransmission delay than the 8 bits symbols.

Figure 5-4 presents the result after a 6 and 8bit / symbol, 60 symbol code word Reed-Solomon decoder. The downlink was better than the uplink for both codes. This is due to the switching diversity as well as the mute function which had no negative effect on the data in the downlink case. For the uplink, the results of 6 bit code were better than those of the 8 bit. This small difference was due to small number of bit error per symbol, and thus the 6 bit code was able to correct more symbols then the 8 bit code. The 6 bit code word also reduced the delay, complexity, required power, etc. For the downlink, all of the advantages of the 6 bit codes were found, and the BER was the same as that of the 8 bit code.

The errors that are due to panic diversity, for the case of (60,40) 8-bit R-S code, will reach a maximum of 16 symbols per frame (one R-S codeword will take 16 TDD frame). This means that a regular Reed-Solomon decoder cannot correct them all. An erasure decoder can correct all of these errors that are due to diversity. The erasure decoder will correct them by increasing the number of correctable symbols by a factor of two¹ (2), when the locations of the errors are known. Since the diversity errors' location is known, then, with an erasure decoder, the frames will be correctable. This will make diversity a very useful function to have, especially in case of low received power, i.e. lower than -84 dBm. Also, with an erasure Reed-Solomon decoder, diversity will reduce the BER by increasing the number of correctable symbols.

Figure 5-5 shows the improved results of the addition of an erasure decoder. Here, the result for the 8 bit/symbol are better than the 6 bit in only one case where the mute function is on. This is because the percentage of 6 bit frame effected by mute function is more than

1. Usually, there is a limit to the number of correctable erasures. For this case 16 erasures is the most, which will allow for two other symbols to be in error,

Figure 5-5: BER using R-S and Erasure decoder for 6bit / symbols & 8 bit codes

that of the 8 bit. Since the delay cost is still almost the same, a 6 bit /symbol code word will work well, and; especially from an implementation point of view, it is a lot less complex to use 6 bit-symbol than a shortened 8 bit-symbol decoder with erasure algorithm

There were few points that are not satisfied by any of the above explanations. These anomalies occurred during: any measurement taken at a power level close to -69 dBm to -74 dBm. Of the anomalies that occurred the one that effected the data the most was the power control algorithm. Also, at -74 dBm the etiquette PCI protocol starts looking for another channel to switch to. If it does find one and switches to it, it causes a loss of bits. Both of these losses are worse than those that are due to diversity. This means that the Reed-Solomon coding/decoding with erasure decoding cannot correct all of the hit frames, thus confirming the requirement of the use of an ARQ scheme.

With the above R-S coder/decoder the BER improves a lot compared to the uncoded data. But the BER is still low for data application, for the cases of received power of less -70 dBm. Again, an ARQ must be implemented to make sure the received data at the application level is correct.

The result, presented in Figure 5-4 and Figure 5-5, of the BER values are also the values of Frame Error Rate (FER). Therefore a FER of $1\text{E-}3$ will decrease the information rate by about 0.1%. This means that when using ARQ and a 6 bits/ symbol, 60 symbols code word per frame, 384 bits frames, with up to 16 (4 unknown location, 12 erasures) correctable symbols R-S code, one can get information data rate of 19.6 Kbps to 21 Kbps (depending on FER).

5.3 Confidence Interval

To establish how accurate the results were, the confidence interval on the collected data was performed. These confidence intervals are presented in Figure 5-6 and Figure 5-7

Figure 5-6 and Figure 5-7, show the 90% and 95% confidence intervals on the

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Figure 5-6: Confidence Interval for downlink
a) diversity off and mute is off b) diversity on and mute is off

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Figure 5-7: Confidence Interval for uplink
a) diversity OFF; b) diversity ON

measurements results that are presented in section 5.2 in Figure 5-1. The 95% confidence interval means that if measurements were to be taken again, then on average nineteen (19) out every twenty (20) measurements results would fall into this confidence interval.

The 95% interval for the downlink is mostly small. Where the interval was large, it was because most measurements' values for that power level were zeros (no error), and one (1) had a lot of error (big standard deviation). This can be explained by one of the following: human error in recording, or measuring, or the measurement was carried out in a place where the channel experienced, deep, fast fades. Since the power reading was an average over one second, it was not very accurate. For the uplink the confidence interval was narrow except for few points: again, this can be explained as above.

To find the confidence interval, consider an ensemble of samples $\{x_1, x_2, \dots, x_n\}$ of size n . The sample mean $\bar{x}$ and sample variance S^2 are defined by [18]

$$\bar{x} = \frac{1}{n} \sum_{i=1}^n x_i \quad (18)$$

and

$$S^2 = \frac{1}{n-1} \sum_{i=1}^n (x_i - \bar{x})^2 \quad (19)$$

respectively

Assume the true variance σ^2 and true mean ξ are known, and assume that all x_i s have a Normal or Gaussian distribution, which is given by

$$Pr(x < \theta) = \int_{-\infty}^{\theta} \frac{1}{\sigma\sqrt{2\pi}} \exp\left[-\frac{(x-\xi)^2}{2\sigma^2}\right] dx = \Omega\left(\frac{\theta-\xi}{\sigma}\right) \quad (20)$$

in which $\Omega(u)$ is the zeros mean, unity variance Normal distribution. Since the

sample mean is the sum of n Gaussian variables, its variance is σ^2/n and its distribution are given by

$$\Omega\left(\frac{\bar{x} - \xi}{\sigma/(\sqrt{n})}\right) \quad (21)$$

If a confidence interval of $1-\alpha$ is required, in which α is the percentage error, it can be calculated by determining the point $u_{\alpha/2}$ at which

$$\Omega(u_{\alpha/2}) = 1 - \frac{\alpha}{2} \quad (22)$$

From equation (21) and equation (22)

$$u_{\alpha/2} = \frac{\bar{x} - \xi}{\sigma/(\sqrt{n})} \quad (23)$$

then the confidence interval is

$$\bar{x} \pm u_{\alpha/2} \frac{\sigma}{\sqrt{n}} \quad (24)$$

That was the case for known variance, but in case of the preformed measurements, the variance is unknown. For such cases, the form of the calculation of the confidence interval in equation (24) changes to [19]

$$\bar{x} \pm t_{n-1, 1-\alpha/2} \frac{S}{\sqrt{n}} \quad (25)$$

in which $t_{n-1, 1-\alpha/2}$ is the value of the Student-t distribution evaluated for $n-1$ degrees of freedom at the point $1-\alpha/2$, and S is the sample standard deviation defined in equation (19). The derivation and form of the Student-t distribution can be found in [20]. The distribution is repeated below for reference

$$P_{T_v}(t) = \frac{\Gamma\{(v+1)/2\}}{(\sqrt{\pi v})\Gamma\left(\frac{v}{2}\right)\left(1 + \frac{t^2}{v}\right)^{(v+1)/2}} \quad (26)$$

in which

$$\Gamma(x) = \int_0^{\infty} y^{x-1} e^{-y} dy \quad (27)$$

Is the gamma function.

Standard tables are available for the evaluation of the Student-t distribution [20]. The above equations work for the samples $\{x_1, x_2, \dots, x_n\}$, with the values of x_i linear. Since the values of the BER of the measurements are of an exponential form, it is therefore required, before using the table of Student-t, to linearize the data. This can be achieved by taking the logarithm of the average BER found by equation (18). Equation (18) then becomes:

$$\bar{x}_{ln} = \ln\left(\frac{1}{n} \sum_1^n x_i\right) \quad (28)$$

and similarly

$$A(n) = \left[\frac{1}{n(n-1)} \sum_1^n (\ln(x_i) - \bar{x}_{ln})^2 \right]^{1/2} \quad (29)$$

With $A(n)$ and $\bar{x}_{ln}$ the confidence interval can be found to be

$$[\exp(\bar{x}_{ln} - t_{\epsilon} A(n)), \exp(\bar{x}_{ln} + t_{\epsilon} A(n))] \quad (30)$$

Where the values of t_{ϵ} are given the table of values for Student-t distribution, and $\epsilon = 1 - \alpha/2$.

5.4 Conclusion

From the above results, one can see that if mute and diversity functions are disabled then the BER will improve, and Reed-Solomon codec will do an excellent job in correcting the errors. These errors will be due to Rayleigh fades, which Reed-Solomon codec is known to combat, while if mute is on, the antenna diversity will result in better BER. This is because when a long fade happens (more than 2 ms) and diversity is off, mute will impair all the data frames. If diversity is on, however, then the mute function will be stopped once the switch between antennas takes place.

If diversity is on and the mute function is off, then with the use of an erasure Reed-Solomon decoder, the channel will be suitable for data transmission. The best information rate will be achieved in this case. This is because the cost that the diversity induces will be correctable by the erasure decoding, and what is left is a reduction of the fade intervals in the channel.

The use of 6 bit R-S code words will be more suitable than 8 bit codec. However, 8 bit per symbol code words only helps in one situation, which is when diversity is on, muting is on, and an erasure decoder is used. In all other cases, however, the 6 bit code will outperform the 8 bit one, or equal it. In all cases the 6 bit will result in smaller delays, less complex coder and decode, less power usage and furthermore, will be small in size.

The tightness of confidence intervals indicates that the result obtained from the measurement are dependable, and give a very good indication of how the channel operates. From these results, we can deduce that the ability to integrate data communication with the voice communications in an indoor wireless networks is possible. Also allowing multiple users to access one voice channel is an achievable prospect. Since the results of the distribution of errors is very close to that of wireless channel experiencing Rayleigh types of fades, the channel can be modeled and the result can be predicated. The modelling of channel is detailed in the next chapter.

6

Analytical Results for Channel Bit Error Rate

6.1 Introduction

The experimental results provided data on the channel impairments for a given frequency and indoor environment. In this chapter, analytical approaches are presented for predicting the bit error rates. Accuracy of the analytical approaches is essential to provide a reliable alternative to expensive and laborious experimental work.

6.2 Background Equations and Assumptions

From chapter 5 we know that the channel experiences Rayleigh fading, which is defined in equation (31). The multipath environment which produces Rayleigh fades results from the movement of people in the path of the transmitted signal. Using these results we can perform the following analysis:

6.2.1 Uncoded Channel Analysis

Modelling the channel as a Rayleigh channel will mean that the signal power will have an exponential distribution. On the basis of this distribution, the probability of the received signal dropping below a given power level can be calculated as follows:

$$p(r) = \begin{cases} \frac{r}{\sigma^2} \exp\left(-\frac{r^2}{2\sigma^2}\right) & r \geq 0 \\ 0 & r < 0 \end{cases} \quad (31)$$

$$p\left(\frac{r^2}{2} \leq z\right) = 1 - \exp\left(-\frac{z}{\sigma^2}\right) \quad (32)$$

Using [25] we can determine the number of times the received signal fades below a given threshold in one second. This number is called the Level Crossing Rate and is given by $N(R)$. $N(R)$ is defined as the number of times the signal drops below a specified signal level R , and given by:

$$N(R) = \int_0^{\infty} \dot{r} p(R, \dot{r}) d\dot{r} \quad (33)$$

where $\dot{r}$ is the time derivative of r , the received signal envelope $p(R, \dot{r})$ is the joint density function of r and $\dot{r}$ at $r = R$. The joint density function given by Rice in the four random variables, $r, \dot{r}, \theta, \dot{\theta}$ of a Gaussian process for the case $b_1 = 0$ is:

$$p(r, \dot{r}, \theta, \dot{\theta}) = \frac{r^2}{4\pi^2 b_0 b_2} \exp\left[-\frac{1}{2}\left(\frac{r^2}{b_0} + \frac{\dot{r}^2}{b_2} + \frac{r^2 \dot{\theta}^2}{b_2}\right)\right] \quad (34)$$

where $\tan \theta = -T_s/T_c$, T_c and T_s are the in-phase and quadrature components of the signal respectively, b_0 and b_2 are moments values given in [25] (equation 1.3-6 to 1.3-11). Integrating this expression over θ from 0 to 2π , and $\dot{\theta}$ from $-\infty$ to $+\infty$ we obtain:

$$p(r, \dot{r}) = \frac{r}{b_0} e^{-r^2/2b_0} \frac{1}{\sqrt{2\pi b_2}} e^{-\dot{r}^2/2b_2} \quad (35)$$

Since $p(r, \dot{r}) = p(r)p(\dot{r})$, r and $\dot{r}$ are independent and uncorrelated. Substituting equation (35) into equation (33) we get the level crossing rate:

$$N(R) = 2\pi f_m \rho \exp(-\rho^2) \quad (36)$$

where $\rho = \frac{R}{\sigma_{rms}} = \sqrt{\frac{p_0}{\sigma^2}}$, p_0 is the reference power level, σ^2 is the average received signal power, f_m is the doppler frequency, $f_m = \frac{v}{c} f_c$, v is the speed of the mobile, in this case the speed of people walking, f_c is the central frequency (1.92 GHz) and c is the speed of light (3E8 m/s). We can also determine the average fade interval, $\langle \tau \rangle$, for a given received average power level.

$$\langle \tau \rangle = \frac{P(r \geq R)}{N(R)} \quad (37)$$

$$\langle \tau \rangle = \frac{1 - \exp(-\rho^2)}{N(R)} \quad (38)$$

Using $\langle \tau \rangle$ and $N(R)$ the BER (Bit Error Rate) is given by:

$$BER = \langle \tau \rangle \times N(R) = P_M \quad (39)$$

which is equal to:

$$BER = 1 - \exp(-\rho^2) \quad (40)$$

Equation (40) can be rewritten in term of the experimental results to be equal to:

$$BER = 1 + \exp\left(-\frac{\text{Reference Power}}{\text{Received Power}}\right) \quad (41)$$

The reference power level for the radio is given in the PCI standard to be -94 dBm for BER of $1\text{E-}3$.

6.2.2 Analysis for The Coded Channel Case

The Reed-Solomon code, which is a kind of block code, was chosen as the error control scheme because of the fact that it is one of the most powerful data protection algorithm in a fading channel environment. Reed-Solomon code as presented in Section 2.5 is described by the following:

$$\begin{aligned}
 N &= q - 1 = 2^k - 1 \\
 K &= 1, 2, 3, \dots, N - 1 \\
 D_{\min} &= N - K + 1 \\
 R_c &= \frac{K}{N}
 \end{aligned} \tag{42}$$

Table 6-1: Parameter description of equation 42

Parameter	Descriptions
N	Total number of symbols transmitted in a codeword.
K	Number of information symbols encoded in the N symbol code word
$D_{\min}$	The minimum code distance
R_c	The code rate
k	The number of the bits in one symbol, and q is the number of symbol available $q = 2^k$.

Such a code is guaranteed to correct up to t symbol errors, where t is given by:

$$\begin{aligned} t &= \left\lfloor \frac{D_{min} - 1}{2} \right\rfloor \\ &= \left\lfloor \frac{N - K}{2} \right\rfloor \end{aligned} \quad (43)$$

Using [12], the upper bound on the code word error probability is given by:

$$P_e \leq \sum_{i=t+1}^N \binom{N}{i} P_M^i (1 - P_M)^{N-i} \quad (44)$$

where P_M is the bit error probability given by equation (40). Therefore when a code word error is made, the symbol error probability is given by:

$$P_{es} = \frac{1}{N} \sum_{i=t+1}^N \binom{N}{i} P_M^i (1 - P_M)^{N-i} \quad (45)$$

If the symbols are converted to binary digits, the bit error probability corresponding to (45) when an erasure decoder is added will be defined as:

$$P_{eb} = \frac{2^{k-1}}{2^k - 1} P_{es} \quad (46)$$

Figure 6-1: BER results for the channel model

The number of erasure e can be determined by the maximum number of diversity switching that occurs in a frame, and a new value for t (the correctable symbol errors) can be derived. The guaranteed correctable symbols in this case is defined by t_m and is given by:

$$t_m = \left\lfloor \frac{d - e - 1}{2} \right\rfloor \quad (47)$$

channel symbol errors in addition to the erasures can be corrected. Thus, all combinations of v channel errors and e erasures are correctable provided that

$$2v + e < d \quad (48)$$

Using equation (40) and (46), the bit error rate after decoding can be calculated. Different parameters for different coder and decoder are used and these parameters are given in Table 6-2 and the results using these parameters are plotted in Figure 6-1

Table 6-2: Calculation Parameters

R-S coder/decoder		R-S and Erasure decoder	
k = 6 bit	k = 8 bit	k = 6 bit	k = 8 bit
N = 60	N = 60	N = 60	N = 60
K = 40	K = 40	K = 40	K = 40
t = 10	t = 10	e = 12, t = 16	e = 16, t = 18

6.3 Analytical Results

In this section we examine the analytical results using the PCI standard specification parameter. This will enable us to perform a comparison between the analytical model result given in the previous section, and the results obtained from the measurements presented in chapter 5.

6.3.1 Raw Bit Error Rate

Computing the BER using equation (40) and for different received power levels, the resultant BER is plotted in Figure 6-1. The PCI standard [2] requires a maximum bit error rate of 1 out 1000 ($1E-3$) at -94 dBm power level, thus, $p_o = -94$ dBm and the BER at -94 dBm is $1E-3$. p_o is the reference power as defined in equation (41)

6.3.2 Bit Error Rate after Coding

When using Reed-Solomon code the errors due to channel fading will be correctable if the number of fades in one frame is less than t , and maximum fade duration is less than one symbol time. Where the maximum fade duration exceeds one symbol time, the number of fades in the frame has to be less than t . For complete analysis of all cases, equations (44), (45) and (46) represent the analytical BER. Figure 6-1 is plotted.

Based on these equations Figure 6-1 shows that the Reed-Solomon coding improves the performance of the transmission by 45 dB from -35 dBm to -80 dBm for BER of $1\text{E-}9$, which is the design goal of data communications. There is about 2 dBm improvement when diversity is off and an erasure decoder is used. The latter will require an access to the soft data values. There is no noticeable difference between 6 bit-codes and 8 bit-codes. The reason that the 6 bit codes and 8 bit codes give the same results is that both employ the same code word length ($N = 60$ and $k = 40$), which results from the use of a shortened 8 bit code word length.

Figure 6-1 presents the estimated BER for the channel where no diversity or a mute function is used. It is shown in section 5.4 that the use of diversity improves the BER by a few (2 to 5) dB. The cost of diversity will be offset by using an erasure decoder.

6.4 Experimental Versus Analytical Results

Results for the analytical approach are now compared to the BER obtained from the measurements.

6.4.1 The Uncoded Channel Comparison

Figure 6-2 shows the comparison between the estimated BER using the above model, and the BER obtained from the channel measurement presented in chapter 5. It can be seen (from Figure 6-2) that the model gives a good approximation (for the downlink) of experiment results.

- 1) U/L OFF: Handset to base station with diversity OFF
- 2) U/L ON: Handset to base station with diversity ON
- 3) D/L ON: base station to handset with diversity ON
- 4) D/L OFF: base station to handset with diversity OFF

**Figure 6-2: Uncoded Channel Measurements' results
versus Model's results**

The analytical results do agree with the measured ones for the case of the downlink. This is to be expected since the model does not take into account the cost of diversity or the mute function. Since the channel is very close to being symmetric, the analytical results will also give a good indication of the BER of the channel for the uplink direction. For the uplink case where diversity is off, the analytical results are equal to the best fit line of the experimental results.

The difference between the experimental and the analytical results can be explained as follows. One the radio design aims at obtaining BER of $1\text{E-}3$ at -94 dBm, while the receiver could in practice provide a better performance. and two if the 95% confidence interval is to be added to the experimental results and the fact the radio could have a better performance, then the model's results will represent an upper bound for the confidence interval of Figure 6-2 and Figure 5-6 and Figure 5-7 of chapter 5.

6.4.2 The Coded Channel Comparison

Plotting the results of the analytical model and the experimental results in Figure 6-3, it is observed that the analytical results are more pessimistic than the results obtained through the measurements. This is emphasized if the system designer takes into account the confidence interval for no coding. Thus $1\text{E-}3$ BER will occur at a received signal level lower than -94 dBm, which means that the radios are designed to provide a BER of $1\text{E-}4$ at -94 dBm.

The signal strength margin corresponding to a change in BER from $1\text{E-}4$ to $1\text{E-}3$ is approximately 5 dB. This explains the difference between the analytical results and the results obtained experimentally¹.

1. Note: the analytical results do not take into account errors due to power control. Thus the difference in the results are noted in the region -74 dBm to -70 dBm; also, when the experimental data is analysed, where no error is found or left after the decoding, then a BER of $1\text{E-}8$ is assumed

Figure 6-3: Coded Channel Measurements versus Model

6.5 Conclusion

The analytical Rayleigh model of the channel gives a good approximation of the BER. This can be used to determine the coverage area as well as the deployment parameters for the system: parameters such as the minimum required received power for a given application and the type of information rate required. For example, for a given information rate and required BER, the minimum received power allowed can be determined by the analytical mode. Range prediction models, which require signal strength as input, can then be used to determine coverage zones for each base station.

In situations where information rate efficiency is not a critical factor, different error control codes can be used. Examples of such codes include check sum (CRC - Cyclic Redundancy Check), linear block codes, etc. These codes are simpler to implement than Reed-Solomon code. However, a simpler implementation of Reed-Solomon codes are expected to be available in the near future. For example, 6 bit-symbol Reed-Solomon coder is already available as a single chip.

7

Recommendations

7.1 Recommendations

Data communication with multiple users access to a voice channel is possible, and it does not requires any major changes to the existing voice networks. The recommended protocol would be implemented at the end users, and will be transparent to the voice network. However, there are a few recommended changes for the voice network so that a better voice quality as well as high data rates are achieved for data links.

This chapter provides recommendation for the voice network, protocol for integrating the data with voice, deployment, and multiple access technique.

7.2 Recommendation for Voice Network

There is only one major recommended change for the voice network. This change is to improve the switching diversity by co-phasing the two antennas together, so that no bit will be lost. This will eliminate the ticking noise that occurs every time there is a switch of antennas. The other minor change is to enable a software switch to turn the mute function off when the channel is used for data communication.

7.3 Recommended Data Transmission Protocol

In order to achieve a data link a protocol will be required to regulate the traffic, and also to ensure that data gets transferred between any two points correctly.

In the following, a protocol is described, which not only allows for data transfers, but also for an easier implementation. It also includes the addition of the Reed-Solomon code as well as ARQ. This protocol is based on current technologies and uses some of specification used in the European CT2 protocol [24].

This protocol could use both full-duplex asynchronous and synchronous data services. With the direct synchronous service the ARQ could not be implemented, thus the integrity of data is not guaranteed. This protocol can support the asynchronous services rates of 300, 1200, 2400, 4800, 9600, 14400 and 19200 bits per second. For the synchronous, the protocol can support the same rates as the above, with the addition of the 32000 bits per second rate without FEC. The latter can be useful for short messaging service, where the information can be transmitted few times. The system components for protocol are presented in Figure 7-1,

7.3.1 The Radio Link

The radio link will provide a full-duplex unrestricted binary channel. All user data are carried in the B channel portion of the radio link bursts. Users wishing to transmit data at 32 kbps without error control may use this channel directly.

7.3.2 Framing and Forward Error Control

This will group the data into a Reed-Solomon frame. In the transmit direction, parity check symbols made up from 18 bits CRC codes are added, which the receiver will check and remove these symbols.

The FEC frame will have 6-bit synchronization pattern (011110, the left most bit will be transmitted first), followed by 378-bit Reed-Solomon codeword. The codewords are made up from 63 six-bit symbols, k of which carry control information and user data, and $63-k$ will be parity symbols. The resulting 384-bit frame shall fully occupy the B channel portion of twelve contiguous transmit bursts

Figure 7-1: Data Transport Components

The FEC frames will be synchronous, with the DCP (Data Cordless Portable Part, handset) synchronized to DCF (Data Cordless Fixed Part). During a data connection using the FFEC, the DCF will transmit an FEC frame every 12 ms. The DCP will find the frame boundary by searching for the synchronization pattern of bursts. Once found, the DCP will verify alignment by checking the parity symbols. A DCP will be in alignment

when it has received three consecutive frames containing a synchronization symbol, in the first symbol of the frame. When the DCPD has achieved frame alignment, it will transmit an FEC frame every 12 ms. The DCPD will start a frame on the ninth burst following the start of a DCFD frame. This will give an Automatic Repeat Request (ARQ) protocol 7.5 ms (in the DCPD) and 6 ms (in the DCFD) to check the Reed-Solomon parity before the first opportunity to send an acknowledgment or rejection.

7.3.3 Link Access Protocol for Radio (LAPR)

LAPR is an ARQ data link protocol providing strong error control in addition to forward error correction provided by FFEC. LAPR is based on X.25 LAPB [22]. The modification to LAPB are only with respect to section 2.2 of [22] (frame format). LAPR frames will be carried in the FFEC message symbols, one LAPR frame per FFEC codeword. The LAPR frame consists of several fields. Starting at the beginning of the FFEC codeword, these will be:

- RFU: Reserved for future use. 4 bits, initially set to 0.
- Length: 6 bits, indicating the number of valid octets following the user Defined Field in the LAPR frame.
- User Defined: 6 bits. This is available to higher level protocols to carry status and control information out of band.
- Address: 8 bits. Usage is as per LAPB
- Control: 8 bits. Usage and format are as per LAPB, basic (modulo-8) mode.
- Information: An integral number of octets, in this case 48.

Figure 7-2: LAPR Frame Format

7.3.4 Synchronous Rate Adaptor

The radio link may be used to carry synchronous channels between a DCPD and a DCFP. The DCPD (and the DCFP) may connect to DTE (Data Terminal Equipment) or DCE (Data Communication Equipment) using an interface such as CCITT V.24 [23]. When using data rate lower than 19.2 kbps, then a different Reed-Solomon code rate can be used.

7.3.5 Asynchronous Packet Assemble/Disassembler (PAD)

In the DCFP, the PAD lies between the LPI (Landline Physical Interface) and LAPR and the asynchronous device such as a terminal or host computer. The PAD receives asynchronous characters from the land-line or DTE. Asynchronous characters typically consist of a start bit that indicates the start of a character, one or more stop bits that indicate the end of the character, and a certain number of bits in between that define that specific character. The character may also contain a parity bit.

The PAD will strip the start and stop bits from an asynchronous character. If the number of data bits per character, plus parity (if any) is less than or equal to eight, the bits will be placed in the least significant bits of the octet. The parity bit (if any) will go in the least significant bit position not occupied by a data bit, and the remaining bits will contain 0 (zero).

7.4 Deployment Recommendations

It is recommended that this system be deployed in such way that no data user will transmitting or receiving data from a location that will results in received power level below -75 dBm, and if possible below -70 dBm so that the power control algorithm will not be activated. This will ensure a better data channel.

7.5 Multiple Access

In order to implement the multiple access method, a new "base station" will be required. This base station will be used by the data users to link to the PCI network. The data base station duties will be as follows:

- 1) Multiplex the users on a single voice channel,
- 2) Convert the user data (from IR data or others) to data that could be sent through the PCI network,
- 3) Implement the above protocol, and
- 4) Act as a handset so that it will be transparent to the voice network, and at the same time making sure that a speech channel is allocated to it during data calls.

8

Summary, Conclusions and Recommendations for Future Research

8.1 Summary

The literature survey gives few ways to improve the bit error rate (BER) of fading channel. The most suitable scheme is Reed-Solomon coding with erasure decoding.

The survey also looked at different methods of allowing multiple users to share the channel. From this it was noticed that to achieve the most efficient scheme a number of details about the application were needed. The requirement of these parameters led to the use of a flexible scheme which allows different applications to work at the same time. This scheme is the PRMA (Packet Reservation Multiple Access). PRMA is very flexible, yet very efficient.

Simulation results were taken from several publications and presented in chapter 3 to show how PRMA works and how it compares to the TDMA (Time Division Multiple Access).

To see the effect of the wireless channel on the data service an experiment was designed. This experiment had an objective of finding the BER of the channel for different locations, different received power levels and different system set up parameters (diversity, mute, uplink and downlink).

The experimental results, presented in chapter 5, showed that the channel experiences Rayleigh fades. The effect of the mute function was clearly noticed. This mute function, when active, caused a very high number of errors to occur. The results also showed that switching diversity has a negative effect on the BER. It increased the BER when every switch of antennas occurred. Reed-Solomon coding the data protection scheme, chosen in chapter 2, was very effective in removing most of these errors.

This Reed-Solomon coding scheme, together with erasure decoding and ARQ (Automatic Repeat request), will allow for an information through rate of about 19.6 Kbps. This rate is a very good information rate. With this rate, and for applications where users do not require a big bandwidth, then users can share the channel very easily.

These results mean that the integration of data communication with voice is possible, and to achieve this integration a protocol that includes an error control will be required to be added to the current system.

Chapter 6 showed that the experimental results can be predicted with a very good accuracy using an analytical method. It was also verified that a 6 bit per symbol Reed-Solomon code will perform better than 8 bit per symbol shortened Reed-Solomon code. The improvement in performance comes from a smaller delay for the ARQ and less complex implementation. At the same time, 6 bit per symbol Reed-Solomon code performed just as well as 8 bit code in correcting errors due multipath and the switching diversity.

A recommendation for a protocol was given in chapter 7. This protocol does not require any modification to the existing voice network. In fact it could be implemented at

the data users' equipment. This will mean that the data communication feature will be transparent to the voice network. A few recommended changes to the voice network were also presented in chapter 7. Recommendations such as a better antenna diversity scheme will improve the performance of the channel for both voice and data links.

8.2 Suggestions for Future Research

The work done in this thesis represents the first set of steps required for adding data communications to wireless voice networks. The next step is to implement the recommended protocol developed for this purpose and to determine if other system problems exist. Testing with multiple users in the network and measuring the effective information rate will be required to determine the impact of any added interference.

After a prototype system of wireless data and voice has been implemented, it will be possible to examine ways of improving the system performance. This can be done by investigating alternative data protection schemes for enhancing the information data rate. Protection methods such as interleaving the symbols of many frames, can be examined. However, the delay caused by interleaving and the resultant complexity in system implementation must be assessed.

In addition, implementation of a multiple access technique would be an extension to the work presented here. It will give a true indication of how many users can share a single speech channel and of the other problems that should be solved in order to provide this type of service, as well as achieve a higher system capacity.

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A

BER results

A.1 Tabulated Results Values

The results that are presented in chapter 5, were plotted using the values in Table A-1

Pr = Received Power, LOG STD = log standard deviation

Table A-1: Results Values.

Pr	BER	LOG STD	90% Confidence Interval		95% Confidence Interval	
			lower limit	upper limit	lower limit	upper limit
Uplink with diversity off						
-70	4.30e-05	0.72	9.36e-06	1.98e-04	5.90e-06	3.14e-04
-75	1.45e-05	0.25	7.06e-06	2.97e-05	5.02e-06	4.18e-05
-80	2.55e-05	0.18	1.51e-05	4.29e-05	1.18e-05	5.49e-05
-85	4.82e-05	0.39	2.18e-05	1.06e-04	1.75e-05	1.32e-04
-90	1.26e-04	0.59	3.17e-05	4.99e-04	1.94e-05	8.11e-04
Uplink with diversity on						
-70	1.75e-05	0.33	6.77e-06	4.5243e-05	4.31e-06	7.09e-05
-75	1.60e-05	0.10	8.51e-06	3.0084e-05	4.49e-06	5.70e-05
-80	1.44e-05	0.09	1.20e-05	1.7242e-05	1.13e-05	1.82e-05
-85	3.20e-05	0.60	5.68e-06	1.8020e-04	2.50e-06	4.09e-04
-90	8.00e-05	0.34	3.59e-05	1.7821e-04	2.70e-05	2.36e-04

2 OF 2

PM-1 3 1/2" x 4" PHOTOGRAPHIC MICROCOPY TARGET
NBS 1010a ANSI/ISO #2 EQUIVALENT

PRECISIONSM RESOLUTION TARGETS

Table A-1: Results Values.

Pr	BER	LOG STD	90% Confidence Interval		95% Confidence Interval	
			lower limit	upper limit	lower limit	upper limit
downlink with diversity on						
-35	5.07e-07	1.67	9.95e-09	2.57e-05	2.49e-09	1.03e-04
-40	5.50e-03	2.22	7.32e-05	4.14e-01	2.38e-05	1.27e+00
-45	1.02e-02	0.49	4.07e-03	2.57e-02	3.17e-03	3.30e-02
-50	1.49e-02	0.29	8.09e-03	2.74e-02	6.72e-03	3.29e-02
-55	1.19e-02	0.61	3.26e-03	4.31e-02	2.20e-03	6.36e-02
-60	1.01e-02	1.46	3.27e-04	3.10e-01	9.78e-05	1.03e+00
-65	1.60e-02	0.10	8.51e-03	3.01e-02	4.49e-03	5.70e-02
-70	1.38e-02	0.61	4.47e-03	4.26e-02	3.28e-03	5.78e-02
-75	1.23e-02	0.52	4.64e-03	3.28e-02	3.64e-03	4.18e-02
-80	6.98e-03	0.44	2.90e-03	1.69e-02	2.27e-03	2.15e-02
-85	1.86e-02	0.37	9.13e-03	3.78e-02	7.59e-03	4.55e-02
-90	2.95e-02	0.64	7.50e-03	1.16e-01	4.95e-03	1.76e-01
downlink with diversity off						
-35	1.00e-08	0.10	7.90e-09	1.27e-08	7.27e-09	1.37e-08
-40	4.30e-05	0.10	5.32e-09	1.88e-08	2.80e-09	3.56e-08
-45	1.00e-08	0.64	4.08e-08	4.47e-07	2.94e-08	6.18e-07
-50	1.35e-07	0.10	8.08e-09	1.24e-08	7.57e-09	1.32e-08
-55	1.00e-08	1.22	7.84e-08	1.10e-05	3.96e-08	2.17e-05

Table A-1: Results Values.

Pr	BER	LOG STD	90% Confidence Interval		95% Confidence Interval	
			lower limit	upper limit	lower limit	upper limit
-60	9.28e-07	1.95	1.24e-07	5.17e-04	3.51e-08	1.82e-03
-65	8.01e-06	1.90	5.56e-05	5.83e-02	2.46e-05	1.32e-01
-70	1.80e-03	1.75	2.82e-04	1.38e-01	1.42e-04	2.74e-01
-75	6.26e-03	2.13	6.19e-05	2.46e-01	2.11e-05	7.22e-01
-80	3.91e-03	0.21	1.14e-02	2.40e-02	1.05e-02	2.60e-02
-85	1.66e-02	0.25	1.25e-02	3.12e-02	1.10e-02	3.54e-02
-90	1.98e-02	0.06	1.94e-02	2.50e-02	1.86e-02	2.59e-02

B

Circuit Diagram

Designed by: Nazim Almaliki, Date: July 11, 1995, format cavoring circuit for the data link

END

1 8-0 6-9 7

FIN