Practical Multiuser Detection Algorithms for CDMA

A thesis for the fulfilment of the

the Degree of Master of Engineering

By Kevin Anderson

An investigation carried out within
School of Communications and Informatics
Faculty of Engineering and Science
Victoria University of Technology
Melbourne, Australia

October 1999

Dedication

This thesis and work in Multiuser Detection is dedicated to my Lord and Saviour, our Lord Jesus Christ.

"Fear of the Lord is the beginning of knowledge. Only fools despise wisdom and discipline" *Prov. 1:7*

"Choose Instruction rather than silver, and knowledge over pure gold. For wisdom is far more valuable than rubies. Nothing you desire can be compared with it" *Prov 8:10-11*

"And now all glory to God, who is able to keep you from stumbling, and who will bring you into his glorious presence innocent of sin and with great joy. All glory to him, who alone is God our Saviour, through Jesus Christ our Lord. Yes, glory, majesty, power, and authority belong to him, in the beginning, now, and forevermore. Amen" *Jude 24-25*

Soli Deo gloria

Declaration

My Master studies were conducted under the guidance of Dr. Fu-Chun Zheng and Associate Professor Mike Faulkner. Some of the research results reported in this paper have been published as an academic paper and presented as a conference paper.

1. K. Anderson, F.C. Zheng, M. Faulkner, "A Study on the performance of the Partial PIC CDMA detector in the presence of time offset errors" *IEEE International Symposium on Signal Processing and its Applications, ISSPA 99*,pp. 709-712 Brisbane Australia.

I hereby declare that the contents of this thesis are the results of original research except where appropriately referenced, and have not been submitted for a degree at any other university or educational institution.

[redacted]

Kevin Anderson
School of Communications and Informatics
Faculty of Engineering & Science
Victoria University of Technology
Melbourne, Australia

Acknowledgements

Research for this thesis has been carried out at the School of Communications and Informatics at Victoria University, Melbourne Australia. The work towards this thesis was initialised in February of 1998.

I wish to express my gratitude to my supervisors Dr Fu-Chung Zheng and Dr Mike Faulkner for their comments and guidance throughout my research work and without them none of this work would have been possible. I am very grateful for the long hours that Dr Zheng has given me for reviewing my thesis. The financial support, which was provide by the School of Informatics and Communications and enabled this work, is also gratefully acknowledged.

I am very thankful to my parents for the help, support and the love they have provided me throughout my life. The positive attitude towards education in our home has been an important driving force for my latter studies.

Abstract

The aim of this thesis is to investigate practical multiuser demodulation algorithms for mobile communications systems that are based on code division multiple access (CDMA) technologies. These include the adaptive receiver and the interference canceller. The overall complexity of implementing them is examined. The effects of the proposed algorithms on reducing of multiple access interference (MAI) and their resistance to the Near Far Effect (NFE) will be explored.

Significant work was performed in the investigation of the effect of time offset errors on the partial parallel interference canceller (PIC). The performance of it is compared against that of the standard PIC. The bit error rate (BER) performances of the standard and partial interference cancellers are simulated in a near far environment with varying time offset errors. These simulations indicate that whilst timing errors significantly affect the performance of both these schemes, they do not diminish the gains that are realised by the partial PIC over that of the standard PIC.

Table of Contents

List of Abbreviations

A/D	Analogue to Digital Converter
AWGN	Additive White Gaussian Noise
B-ISDN	Broadband Integrated Services Digital Network
BER	Bit Error Rate
BPSK	Binary Phase Shift Keying
CDMA	Code Division Multiple Access
DS-CDMA	Direct Sequence Code Division Multiple Access
DS-SS	Direct Sequence Spread Spectrum
FDMA	Frequency Division Multiple Access
FIR	Finite Impulse Response
GFC	Gradient - Search Fast Converging
HD	Hard Decision
IC	Interference Canceller
ISI	Intersymbol Interference
LMMSE	Linear Minimum Mean Squared Error
LMS	Least Mean Square
MAI	Multiple Access Interference
MATLAB	Matrix Laboratory (Mathematical Programming Tool)
MF	Matched Filter
MLSD	Maximum Likelihood Sequence Detector
MUD	Multiuser Detector
NBI	Narrowband Interference
NFE	Near Far Effect
PIC	Parallel Interference Canceller
PN	Pseudo Noise
SD	Soft Decision
SIC	Serial Interference Canceller

SINR	Signal Interference Noise Ratio
SNR	Signal Noise Ratio
TDMA	Time Division Multiple Access
W-CDMA	Wideband Code Division Multiple Access
ZF-BLE	Zero Forcing Block Linear Equaliser

List of Figures

List of Tables

1 Introduction

Transmission of information has become a key feature of the modern way of life. The possibilities offered by telecommunications are changing the way people work, shop and spend their leisure time etc. The advancing communication and information processing technologies create more markets for new communication services and products. In particular, the demand for wireless communication services has increased rapidly and this trend is expected to continue. As these technologies improve, services will demand a higher throughput of data. Wireless video and Internet services will soon be available to the consumer's mobile phone. To cope with these changes, current wireless technologies will have to change in order to cope with both the higher transfer rate of data and as well as not causing a degradation in the overall system capacity. Current multiple access technologies include Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA).

With advancements in technology, wireless communications is forever changing. We are seeing new types of spectrally efficient modulation schemes, increasing emphasis on digital systems and increasing concern with spectral efficiency and user capacity.

The growth in wireless communications necessitates more efficient utilisation of the available spectrum. Increased capacity and sharing of the spectrum translates into a higher likelihood of users interfering with one another as well as interference from multi-path fading. Interference rejection techniques allows a higher capacity of users within the available spectrum.

Internationally the current trend in wireless communications is towards a system that can handle this increase in demand for higher capacity as well as suppressing any interference from multipath fading and other users. Wideband code division multiple access (W-CDMA) is such a system that the international communications community is considering for any future developments in wireless communications. It is deemed as one of the key growth areas in wireless communications.

Over the past several years, CDMA has shown to be a viable alternative to both frequency division multiple access (FDMA) and time division multiple access (TDMA), and the use of spread spectrum techniques (upon which CDMA is based) in wireless communications applications has become a very active area of research and development. While there does not appear to be a single multiple access technique that is superior to others in all situations, there are some characteristics of spread spectrum systems that give CDMA many advantages.

The two basic problems which can cause interference in a radio link are multipath fading and interference from other users or systems that are operating in close proximity. Spread Spectrum signals are effective in mitigating multipath interference and interfering signals due to the wide bandwidth and spreading gain, which introduces frequency diversity. This results in a system that has a higher capacity compared to that of a non spread spectrum system.

One of the key factors that limits the capacity and performance of CDMA systems is Multiple Access Interference (MAI). MAI is responsible for the Near-Far Effect (NFE), in which strong unwanted signals completely swamp out the weak wanted signal. The NFE is only a limitation of the conventional receiver not CDMA itself. If perfect power control is used then the contribution to MAI by any one user is usually small, but as the number of users increases, the problem with MAI also becomes significantly important.

In a communications system that is based on conventional detection, the immunity against MAI depends heavily on the selection of the spreading signatures. If the chosen spreading codes or signatures are orthogonal to each other then the impact of the MAI on the receiver will be minimal. In most practical systems, the signatures are asynchronous, i.e. the relative delays of signals coming from different users are arbitrary; this results in non-zero crosscorrelations between the received signatures. To achieve a low level of interference in an asynchronous system, the assigned signatures need to have low crosscorrelation for all relative time delays.

To reduce the impact of MAI and the Near Far Effect, a receiver could be designed to take into consideration this interference. This type of receiver would track and demodulate all of the user waveforms simultaneously; which is termed multiuser demodulation. The multiuser receiver makes a decision based on the observation of the whole received waveforms for all users.

This thesis comprises eight chapters. Chapter 1 is an introduction to spread spectrum communications from a systems point of view and presents why multiuser receivers are necessary for a CDMA system. Chapter 2 investigates mathematical models for a direct sequence code division multiple access system (DS-CDMA), it details how the simulations were setup and presents the results for the conventional matched filter receiver. An overview of multiuser detection is presented in Chapter 3. An indepth analysis of multiuser detection algorithms is detailed in Chapter 4. The adaptive linear receiver is discussed in Chapter 5 and results for the convergence of the adaptive algorithm and its performance in a near - far environment are presented. The operation of the parallel and serial interference cancellation techniques are discussed in Chapter 6. The thesis is concluded in Chapter 7 where a discussion of the future directions for research in the area of multiuser detection is also given.

2 Direct Sequence Spread Spectrum Systems

This chapter starts off by looking at a how a direct sequence spread spectrum (DS-SS) signal is formed. It then describes the advantages of spreading gain and processing gain and investigates mathematical models for a direct sequence code division multiple access (DS-CDMA) system. The chapter is concluded with the results for the conventional matched filter receiver.

2.1 Direct Sequence Spread Spectrum

The usage of a spread spectrum system implies that the signals spectrum has been expanded and its signal energy has been distributed over a much larger bandwidth. If the signal suffers from frequency selective fading in the channel, only a small portion of the original signal will suffer some degradation. This is due to that it is very unlikely that all frequencies within the widened signal bandwidth will be faded.

In the transmitter, the spreading sequence or signature multiplies the original narrowband signal to cause a spectral spreading of the original narrowband signal. This signature comprises of a pusedo-noise like (PN) sequence that operates at a much higher chip rate (the elements of the PN sequence are referred to as chips) than that of the users' symbol rate. A period of the PN sequence is exactly as long as one bit or symbol of the data sequence.

In the receiver the (complex-conjugated) spreading sequence again multiplies the received signal to collapse the spectrum. If the reference signature of the receiver is synchronised to the data modulated PN sequence in the received signal, the original signal can be recovered. A brief overview of the operation of Direct Sequence Spread Spectrum is detailed in Figure 1.

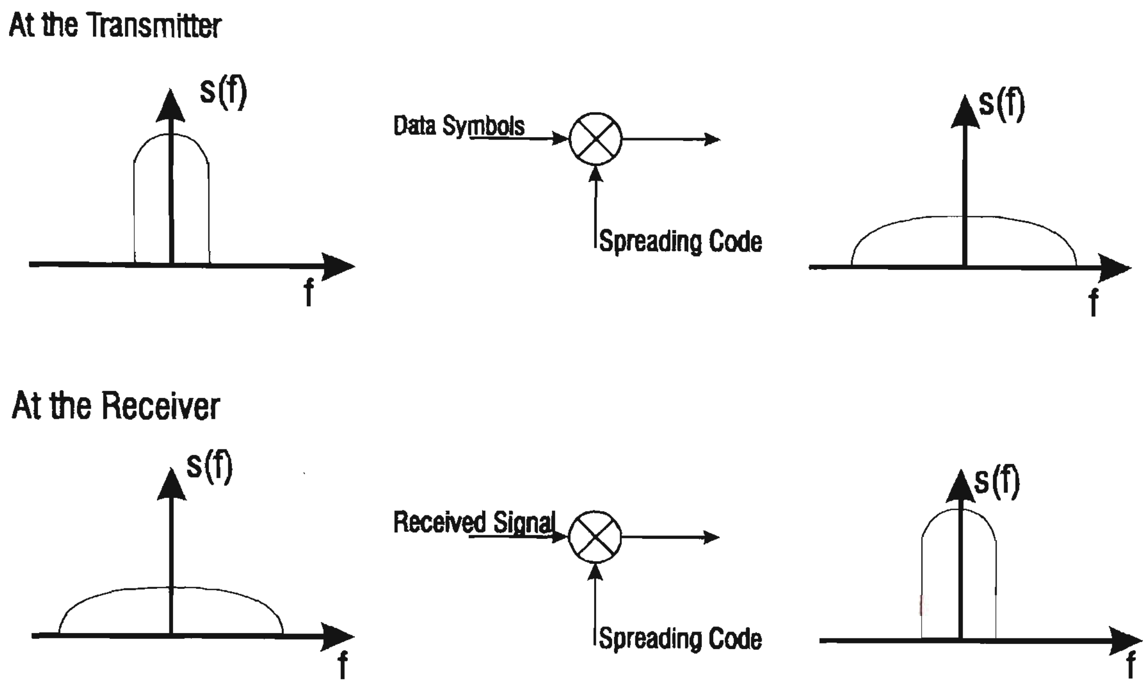


Figure 1. Formation of a Spread Spectrum Signal.

2.2 Spreading Gain

One of the big advantages of spread spectrum lies in its ability to reject interfering signals, this is accomplished by its processing gain. The processing gain of a spread spectrum system is the signal to noise ratio improvement of a spread spectrum system due to the spreading and despreading of the desired signal.

If we let ε be the spreading operation, then the wideband spread spectrum signal s_w becomes [1]

$$s_w = \varepsilon(s_n) \qquad (1)$$

At the receiver if the wanted signal, s_w, is received in the presence of a strong jamming signal, I_n, the despreading process becomes:

$$\begin{aligned} \varepsilon^{-1}(s_w + I_n) &= \varepsilon^{-1}(\varepsilon(s_n) + I_n) \\ &= s_n + \varepsilon^{-1}(I_n) \qquad (2) \\ &= s_n + I_w \end{aligned}$$

From (2) we can see that the despreading process has converted the received signal into a wideband interfering signal and a wanted signal containing the desired users information. After narrowband bandpass filtering, only a small portion of the interfering signal will remain as residual wideband interference.

Figure 2 shows that with spreading gain, an interfering signal can co-exist in the same spectrum as the desired spread spectrum signal.

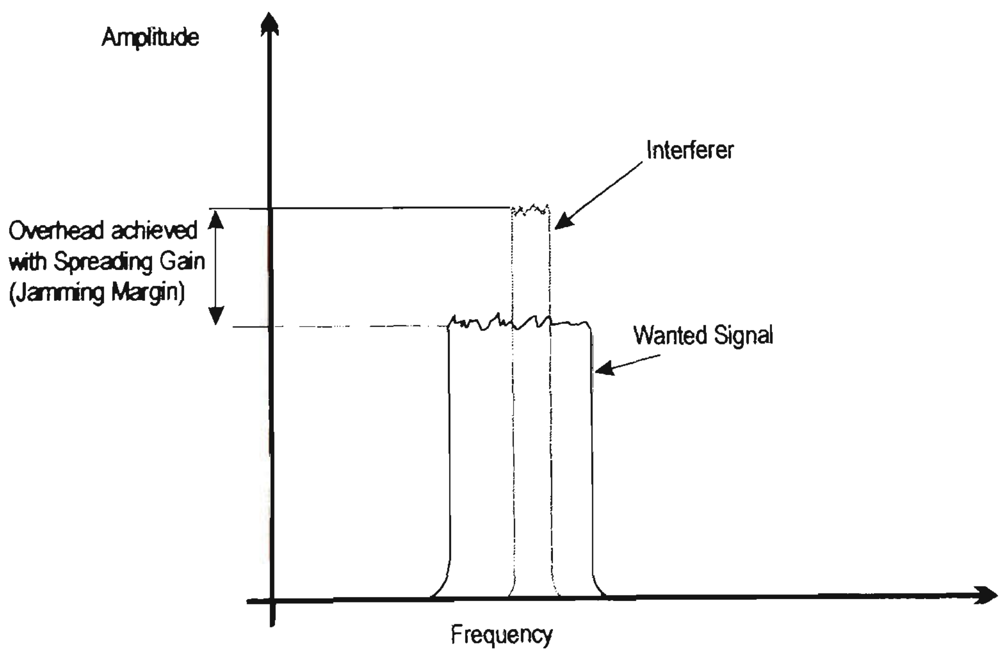


Figure 2 The effect of processing gain on an interfering signal.

The processing gain of the system can be expressed as the ratio of the bandwidth of the transmitted spread spectrum signal to the data rate of the baseband signal [2].

$$G_p = \frac{BW_{RF}}{R_{info}} \qquad (3)$$

Where BW_{RF} is the bandwidth of the spread spectrum signal or alternatively the chip rate of the signature and R_{info} is the data rate of the baseband signal.

To specify how well the spread spectrum system will perform in the presence of hostile and interfering signals, we need to take into account the signal to noise ratio at the information output as well as any overall system losses. The term that is used to specify exactly how well the system will perform in the presence of hostile and interfering signals is jamming margin.

Jamming Margin is defined as the amount of interference a system is able to withstand while producing the required output signal to noise ratio (SNR) or bit error rate BER [2].

$$J_m = G_p - \left[L_{sys} + SNR\right] \qquad (4)$$

Where J_m is the Jamming Margin, G_p is the Processing Gain, L_{sys} is the System Loss and SNR is the Signal to Noise Ratio.

2.3 Characteristics of a Code Division Multiple Access System

In a Code division multiple access (CDMA) system, which is based on Direct Sequence (DS) spread spectrum, each user transmits on the same frequency and uses orthogonal codes to identify one user from another. In this context orthogonality refers to that all of the codes employed in a network must have low mutual cross correlation so that they do not interfere with one another when all of the signals, present in the system, are applied to a receiver in the network. Using DS waveforms, the effects of multipath propagations can be rejected and the overall performance can be enhanced further by combining the multipath returns in a rake receiver. A rake receiver takes advantage of the multiple paths to provide diversity by demodulating and despreading the multipath components. It consists of a bank of correlators, each of them is used to detect separately one of the strongest multipath components.

The system deployment can be improved through the reuse factor of 1:1. This means that in a CDMA system each user can operate on the same carrier frequency throughout the entire network. In mobile systems, users transmit independently from each other and their signals arrive asynchronously at the base station. Since their relative time delays are randomly distributed, the cross correlation between the received signals coming from different users is non-zero. To achieve a low level of interference, the assigned signatures need to have low cross correlations for all relative time delays. Low cross correlation between signatures is obtained by designing a set of orthogonal sequences. However, there is no known set of code sequences that are orthogonal when they are used in an asynchronous system. The non orthogonal components of signals of other users will appear in the demodulated signal as interference.

One major disadvantage with CDMA is the near far effect. This is caused when a weak wanted signal is received at the base station from a distant mobile, that is *far,* in the presence of a strong unwanted signal from a *nearby* interferer. An interfering signal with a power n times stronger than that of the desired signal will have the same effect on system capacity as n interferes of the same power as the desired signal. To combat this near far effect (NFE) power control is used to adjust each users signal so that they arrive at the base station with the same signal level.

Multiuser detectors are NFE resistant and provide a way of relaxing the power control specifications of the CDMA system.

Interference rejection capabilities of the CDMA signals means that it can co-exist with both existing analogue and digital systems, any sources of interference are transformed into wideband noise during the despreading operation.

2.4 Model of Asynchronous CDMA System

A model of an asynchronous CDMA system consists of the transmitter, a channel through which the signal is passed through and a receiver. A general diagram of a CDMA system is illustrated in Figure 3. In this model K users share the same communications media and the signals transmitted by the users pass through separate and independent channels. The transmitted data is demodulated in a centralised multiuser receiver, which makes a joint decision of the data of all users.

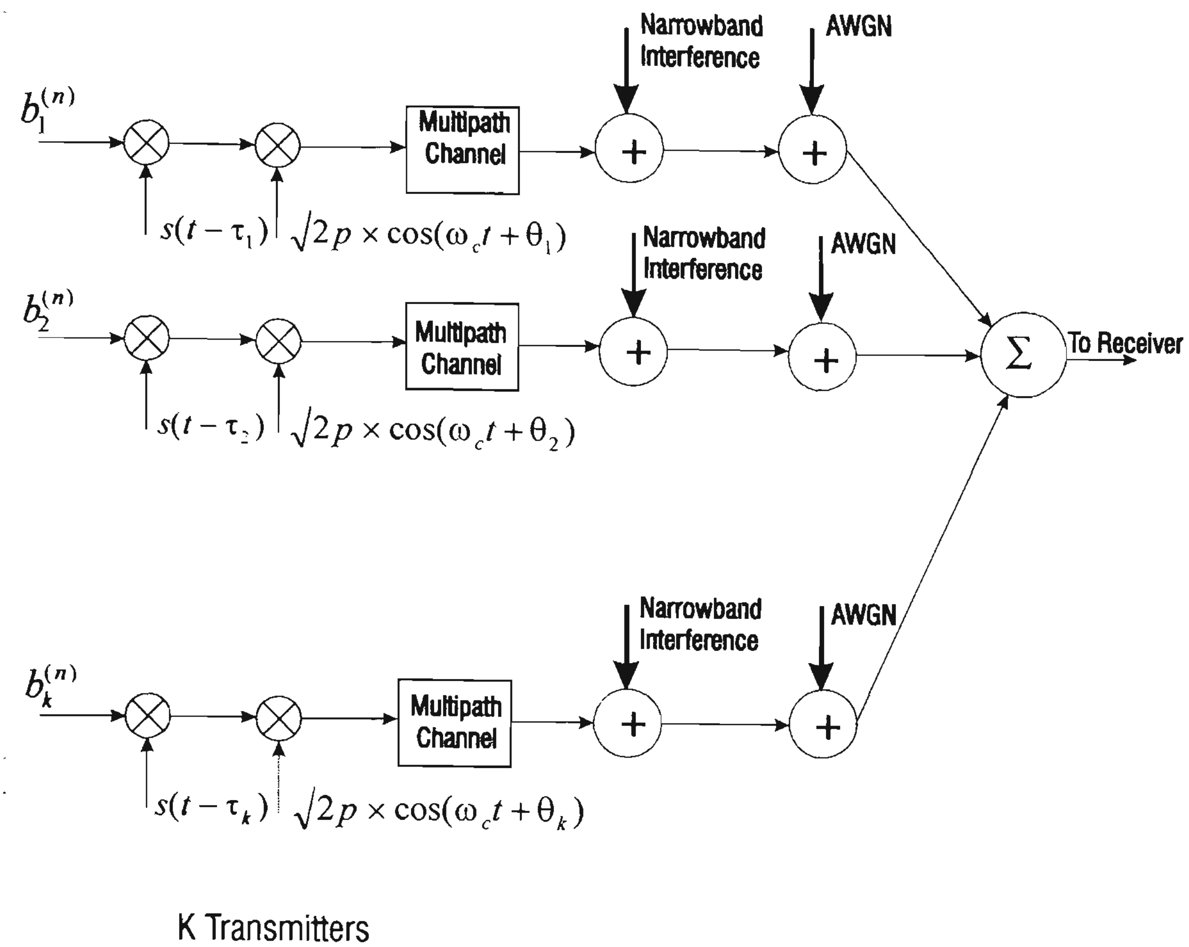


Figure 3. Model of a DS-CDMA system with *K* users

2.5 Transmitter Model

A user *k* from the set {1,2,3,..*K*} transmits in the *n*th symbol interval *t*, where $t \in [(n-1)T, nT]$, a complex signal :

$$x_k(t) = A_k b_k(n) s_k(t - \tau_k), \qquad (5)$$

where *T* is the length of the symbol period, $b_k(n) \in \Xi$ is the transmitted complex data symbol and Ξ is the modulation symbol alphabet. The complex amplitude, denoted by A_k, is given by $\sqrt{E_k} e^{j\phi_k}$, E_k is the energy per bit of the corresponding real bandpass signal, ϕ_k is the carrier phase and τ_k is the delay of the *k*th user's transmitted signal. The signature waveform of user k is given by $s_k(t)$, this can be normalised so that if t falls outside the symbol period then $s_k(t) = 0$ otherwise $\int_0^T \left|s_k(t)\right|^2 = 1$. In a DS-CDMA system the signature waveforms are of the form:

$$s_k(t) = \sum_{m=0}^{N_c-1} s_{k,m}\psi(t - mT_c), \tag{6}$$

where $s_{k,m}$ is the mth chip of user k, T_c is the length of the chip period, $N_c = \frac{T}{T_c}$ is the processing gain, and $\psi(t)$ is the binary chip waveform.

2.6 Channel Model

It is assumed that the channel of User k is a linear filter with the following impulse response:

$$c_k(t) = \sum_{l=1}^{L} \delta(t - \tau_{k,l}) \cdot g_{k,l}, \tag{7}$$

where $g_{k,l}$ is the complex channel coefficient, subject to the channel model. It could be, Rayleigh, Rappaport or Rican. The relative delay for the multipath component l is denoted by $\tau_{k,l}$.

Besides being subjected to the response of the channel, User k is exposed to several other forms of interference; these include Additive White Gaussian Noise (AWGN), Interference from other users commonly known as MAI and NarrowBand Interference (NBI).

2.7 Processing the received signal

The above model, (see Figure 3) can be expressed as:

$$r(t) = \sum_{n=0}^{N_b-1}\sum_{k=1}^{K} b_k(n)A_k s_k(t - nT - \tau_k)\cdot c_k(t) + n(t)$$

$$= \sum_{n=0}^{N_b-1}\sum_{k=1}^{K} b_k(n)A_k\sum_{l=1}^{L} g_{k,l}s(t - nT - \tau_{k,l}) + n(t), \tag{8}$$

where N_b is the number of symbols in the packet and $n(t)$ is AWGN.

In an AWGN channel $g_{k,1}$ can be incorporated into the amplitude A_k (see Figure 4) and the received signal becomes:

$$r(t)=\sum_{n=0}^{N_b-1}\sum_{k=1}^{K} b_k(n)A_k s_k(t-nT-\tau_k)+n(t). \tag{9}$$

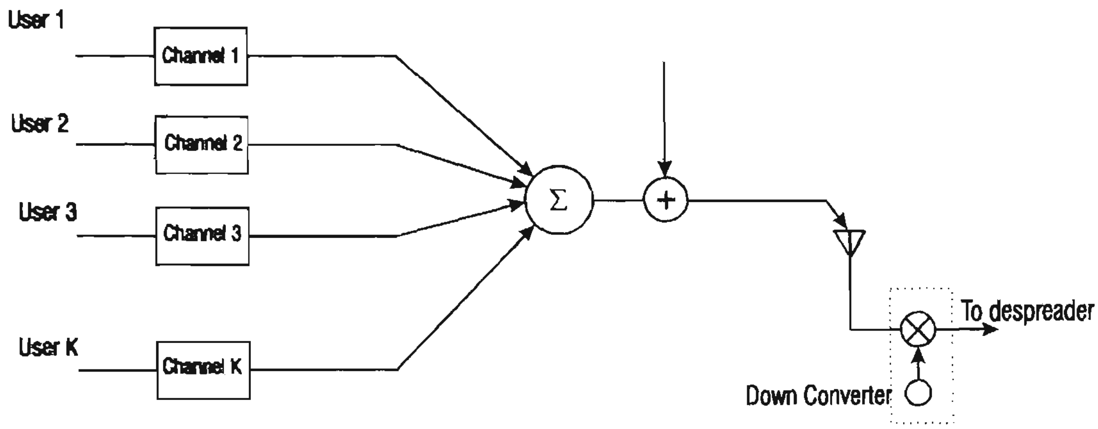


Figure 4. Diagram showing the signals of Users 1..K at the antenna input at the receiver.

2.7.1 Continuous Time Model

It has been shown that the set of matched filter (MF) outputs sampled once in a symbol interval forms sufficient statistics for the detection of the transmitted data [3]. The sampled output of the filter matched to the *j*th users' *l*th multipath component is

$$y_{j,l}(i)=\int_{iT+\tau_{j,l}}^{(i+1)T+\tau_{j,l}} r(t).s_{j_{rep}}(t-iT-\tau_{j,l})dt\ , \tag{10}$$

where $s_{j_{rep}}(t)$ is a replica of the desired user's spreading code at the receiver.

This can be expanded by inserting $r(t)$ from (8) to form :

$$y_{j,l}(i)=\sum_{n}\sum_{k=1}^{K}\sum_{l=1}^{L} b_k(n)A_k g_{k,l}\int_{iT+\tau_{j,l}}^{(i+1)T+\tau_{j,l}} s_k(t-nT-\tau_{k,l})\cdot$$

$$s_{j_{rep}}(t-iT-\tau_{j,l})dt+n_j(i)$$

(11)

where $y_{j,l}(i)$ refers to the output of the matched filter of User j in the l_{th} multipath channel and $n_j(i)$ is equal to :

$$n_j(i)=\int_{iT+\tau_{j,l}}^{(i+1)T+\tau_{j,l}} n(t).s_{j_{rep}}(t-iT-\tau_{j,l})dt\,. \qquad (12)$$

In an ideal channel with no multipath fading this reduces to:

$$y_j(i)=\int_{iT+\tau_j}^{(i+1)T+\tau_j} r(t).s_{j_{rep}}(t-iT-\tau_j)dt. \qquad (13)$$

In this case (9) can be substituted into (13) to be expanded as [4]:

$$y_j(i)=\sum_{n}\sum_{k=1}^{K} b_k(n)A_k\int_{iT+\tau_j}^{(i+1)T+\tau_j} s_k(t-nT-\tau_k)s_{j_{rep}}(t-iT-\tau_j)dt+n_j(i)$$

$$=\sum_{k=1}^{K}\sum_{n=i,i\pm 1} b_k(n)A_k\int_{0}^{T} s_{j_{rep}}(t)s_k(t+(i-n)T-\tau_k+\tau_j)dt+n_j(i) \qquad (14)$$

These equations can be simplified by letting $\boldsymbol{R}$ be the cross correlation matrix between the received signatures and the locally generated signatures. In this case $\boldsymbol{R}$ can be represented as :

$$
\boldsymbol{R} = \begin{bmatrix}
H_0^{(n)}(1) & H_0^{(n)}(0) & 0 & 0 & \cdots & \cdots & 0 \\
H_1^{(n)}(1) & H_1^{(n)}(0) & H_1^{(n)}(-1) & \cdots & & 0 & 0 \\
0 & \cdots & H_2^{(n)}(1) & H_2^{(n)}(0) & \ddots & 0 & \vdots \\
\vdots & & \ddots & & \ddots & \ddots & 0 \\
\vdots & & & \ddots & & \ddots & H_{K-2}^{(n)}(-1) \\
0 & \cdots & \cdots & 0 & & H_{K-1}^{(n)}(1) & H_{K-1}^{(n)}(0)
\end{bmatrix}, \tag{15}
$$

where

$$
H_{j,k}(i) = \int_0^T s_k(t)s_j(t + iT + \tau_j - \tau_k)dt. \qquad i = 0, \pm 1 \tag{16}
$$

Using the result of (16), (14) can be rewritten as [4]

$$
y_j(i) = \sum_{k=1}^{K} b_k'(i+1)H_{jk}(-1) + \sum_{k=1}^{K} b_k'(i)H_{jk}(0) + \sum_{k=1}^{K} b_k'(i-1)H_{jk}(1) + n_j(i). \tag{17}
$$

where

$$
b_k'(i) = b_k(i).A_k \tag{18}
$$

The output of the matched filter can be described in matrix notation as

$$
\boldsymbol{y} = \boldsymbol{RCAb} + \boldsymbol{n} \tag{19}
$$

were **R** is the correlation matrix, **C** is the channel matrix, **A** is the amplitude matrix and **b** is the symbol matrix. In an AWGN channel, the matrix **C** becomes the identity matrix.

2.7.2 Discrete Time Model

In most real communications systems, the input to the matched filter/Multiuser Detector is a sampled version of the signal at the input to the antenna. This received signal is passed through an analogue to digital converter, (A/D) and is sampled at N_s samples per chip, before it is decorrelated by the matched filter.

Figure 5 details a simplified block diagram of a discrete Multiuser Receiver for CDMA applications. We can observe that it contains a down converter, which translates the RF spectrum to a low frequency (10MHz or lower). This signal is then fed into an A/D converter and then into a digital signal processor which detects and despreads the desired signal.

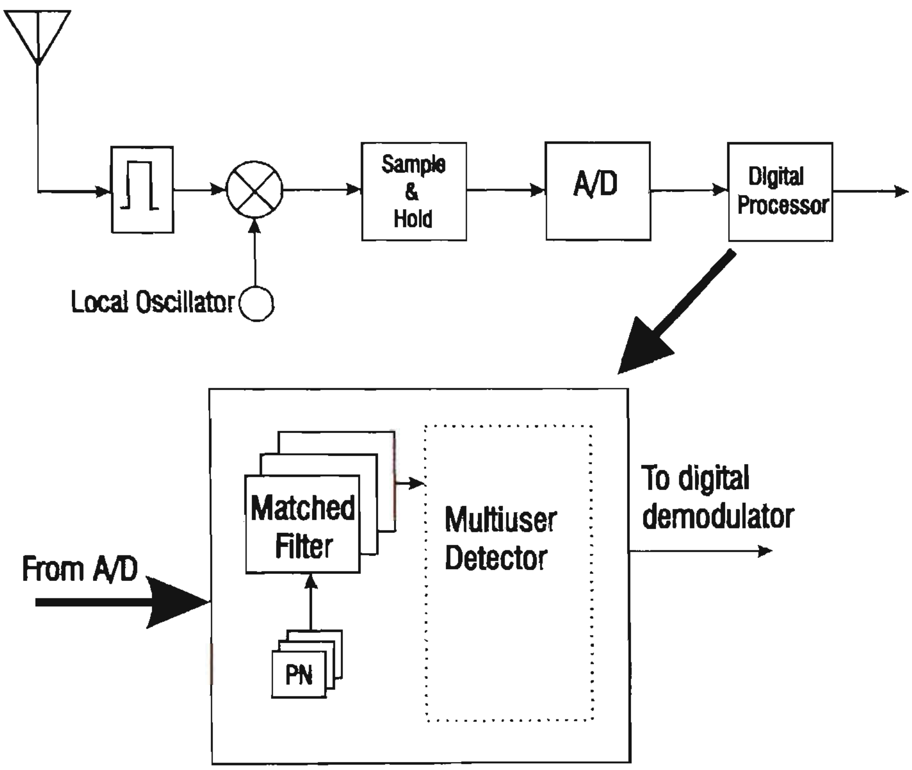


Figure 5. Block diagram of a discrete CDMA receiver.

The input to the receiver is given by :

$$r(t) = \sum_{n=0}^{N_b-1} \sum_{k=1}^{K} \sum_{l=1}^{L} b_k(n) A_k g_{k,l} s_k(t - nT - \tau_{k,l}) + n(t) \tag{20}$$

after down conversion and sampling, this input becomes.

$$r(mT_s) = \sum_{n=0}^{N_b-1} \sum_{k=1}^{K} \sum_{l=1}^{L} b_k(n) A_k g_{k,l} s_k(mT_s - nT - \tau_{k,l}) + n(mT_s) \tag{21}$$

This signal is then applied to the matched filter and Multiuser Detector. The symbol at the ith symbol interval is given by $y_{j,l}(i)$, the output of the matched filter:

$$y_{j,l}(i) = \sum_{n}\sum_{k=1}^{K} r_k(mT_s).s_{j_{rep}}(mT_s - iT - \tau_j) + n(mT_s)$$

$$= \sum_{n}\sum_{k=1}^{K}\sum_{l=1}^{L} A_k b_k(n) g_{k,l} s_k(mT_s - nT - \tau_{k,l}).s_{j_{rep}}(mT_s - iT - \tau_{j,l}) + n(mTs) \quad (22)$$

From the output of the matched filter the discrete signal is passed to the multiuser detection algorithm.

2.8 Performance Analysis of the Single User Receiver

In order to evaluate the benefits of any of the proposed Multiuser Receivers, we first need to accurately simulate the conventional receiver, and investigate its performance under various conditions. These results will form a reference for future simulations. Any improvements that are made with the multiuser receiver will be able to be compared directly against the conventional receiver.

The simulations were set up using the following parameters :

Gold Code Sequence	31
Modulation Scheme	BPSK
Spreading Gain	8.85dB
Data Length	10^4 bits
Samples per chip	4
Monte Carlo Simulations	10

Table 1 Parameters for the conventional Matched Filter Receiver.

Note that when the Multiuser receivers were simulated, the data length decreased to 10^3, to shorten the simulation execution time. In all simulations the system was asynchronous and it was assumed that there was no fading in the channel.

An abstract diagram of the complete CDMA system is depicted in the figure below, and this diagram shows the sequence of events that are used to analyse the proposed receivers that are used in the CDMA system. The performance of a digital communications system is evaluated by observing the Bit Error Rate (BER). The BER of a system is calculated by comparing the transmitted data stream with the received data stream, counting the number of errors and then dividing the result by the total number of bits that were transmitted.

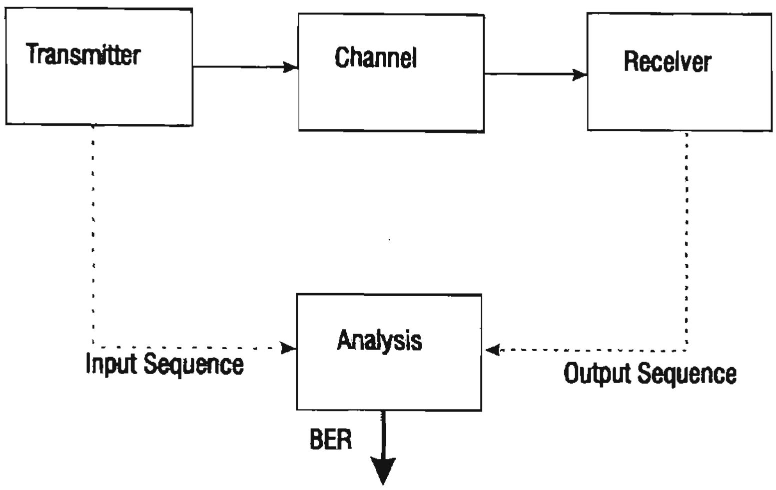


Figure 6. Block diagram of BER analysis.

The software package Matlab was used to simulate the receivers. The receivers were implemented as programs and then executed within the matlab environment. In all simulations common modules were used, these included functions to generate the random data, implement the Gold Code Sequences and perform the spreading. Specialised functions that implemented the actual despreading and data detection were added to these common modules to form the multiuser detector.

2.8.1 Conventional Single User Receiver

In this section the results of the single user conventional matched filter receiver are presented. We begin by investigating its theoretical performance with several different spreading codes and compare these against data obtained from simulations.

The Q function (23) is used to determine the BER response of the receiver. A constant amplitude is assumed for each simulation.

$$Q(x)= \frac{1}{\sqrt{2\pi}}\int_{x}^{\infty} e^{-\frac{y^2}{2}}dy. \qquad (23)$$

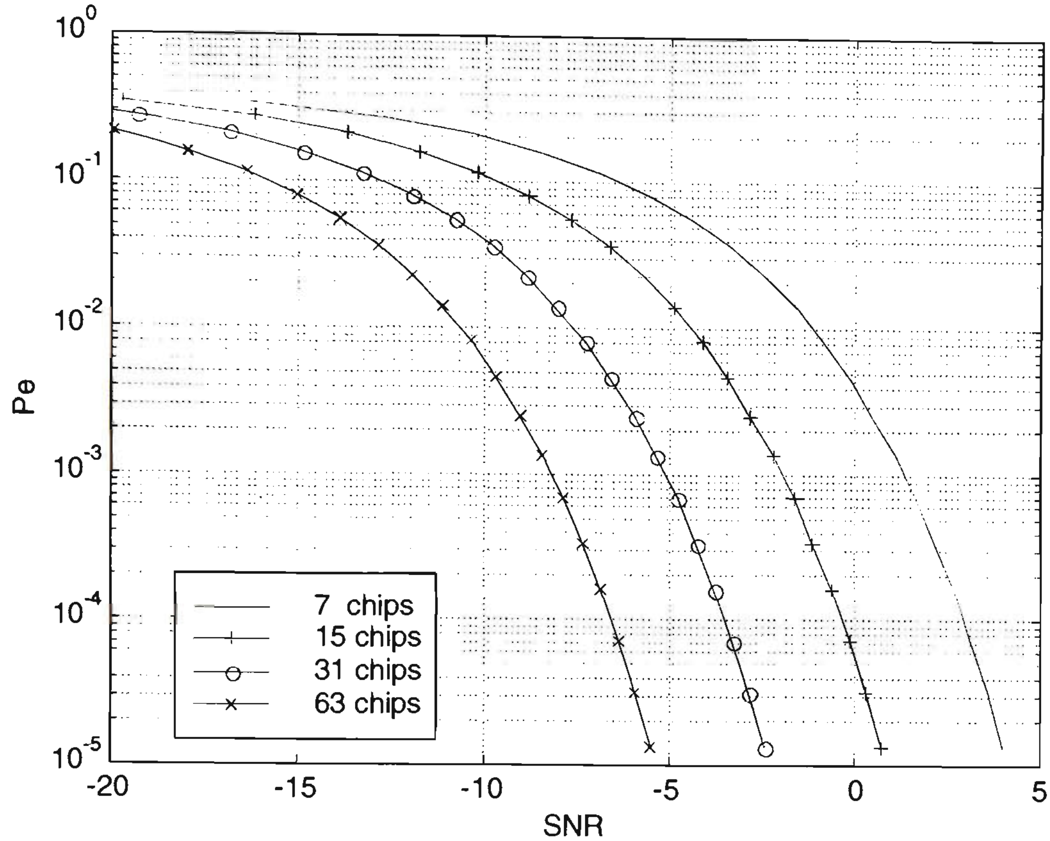


Figure 7. Theorectical BER Vs SNR for Codelengths of 63 31 15 7

The graph shown in Figure 7 details the theoretical BER for a single user DS-CDMA system using spreading gains of 63, 31, 15 and 7. Here we can observe that the users in this system can transmit data when the system is subject to a high level of noise. Figure 7 shows that for an SNR of -5dB, and a spreading code length of 31 chips, the received BER is equal to 10^{-3}.

To evaluate the BER of an experimental single user system, the simulations were set up so that it was representative of a practical spread spectrum system. User data were generated randomly and mapped to a BPSK format and then spread using spreading codes of several different lengths, which were 63, 31, 15 and 7 chips. To characterise the channel, AWGN was added; the noise level was varied according to the calculated SNR.

Figure 8 shows the results for the practical system, it can be observed that they correlate well with those in Figure 7 . This indicates that the experimental system was set up correctly and that the results are accurate. From table 1, as the data length is 10000 and the number of simulations is 10, this would provide a simulation accuracy of around 0.001. Figure 8 shows that for an SNR of -5dB the expected received BER would be 10^{-3}.

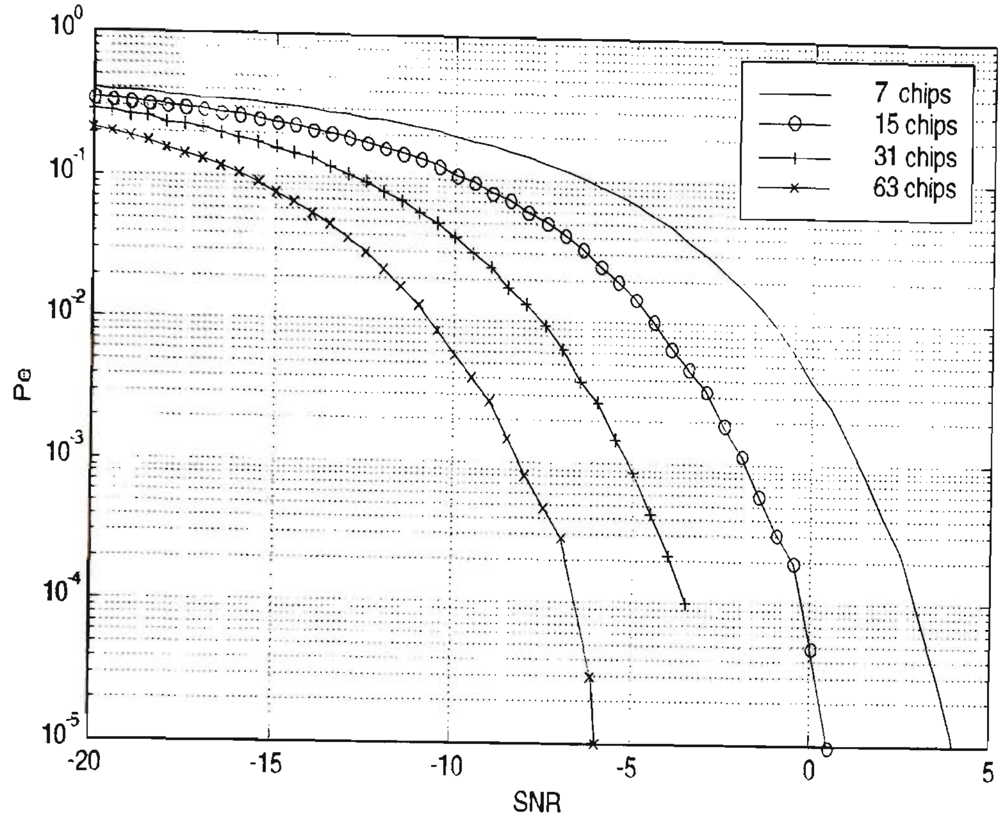


Figure 8. Experimental BER Vs SNR for CDMA System with Codelengths of 63, 31, 15 and 7

2.8.2 Simulation of a CDMA system with Near Far Effect

The NFE for the conventional receiver was simulated with 2, 4 and 8 users.

The effect of the NFE with multiple users is observed in Figure 9. Here the amplitude of one user(s), was held constant whilst the amplitudes of the other users, user 2,4 and 8, varied from -20dB to +20dB with respect to User 1. This simulated the effect of detecting the wanted signal when it is both far (-20dB) and near (+20dB).

This simulation is a very good illustration of how the performance of the conventional receiver using a matched filter correlator is limited by both the NFE and MAI. It shows that the receiver experiences difficulty in detecting weak wanted signals when a strong interfering signal is near by, and exceeds that of the spreading gain of any of the users in the system. It can be observed that as the number of users increased, the amplitude that was required by User 1 to achieve the same BER as the 2 or 4 user case, had to be increased also.

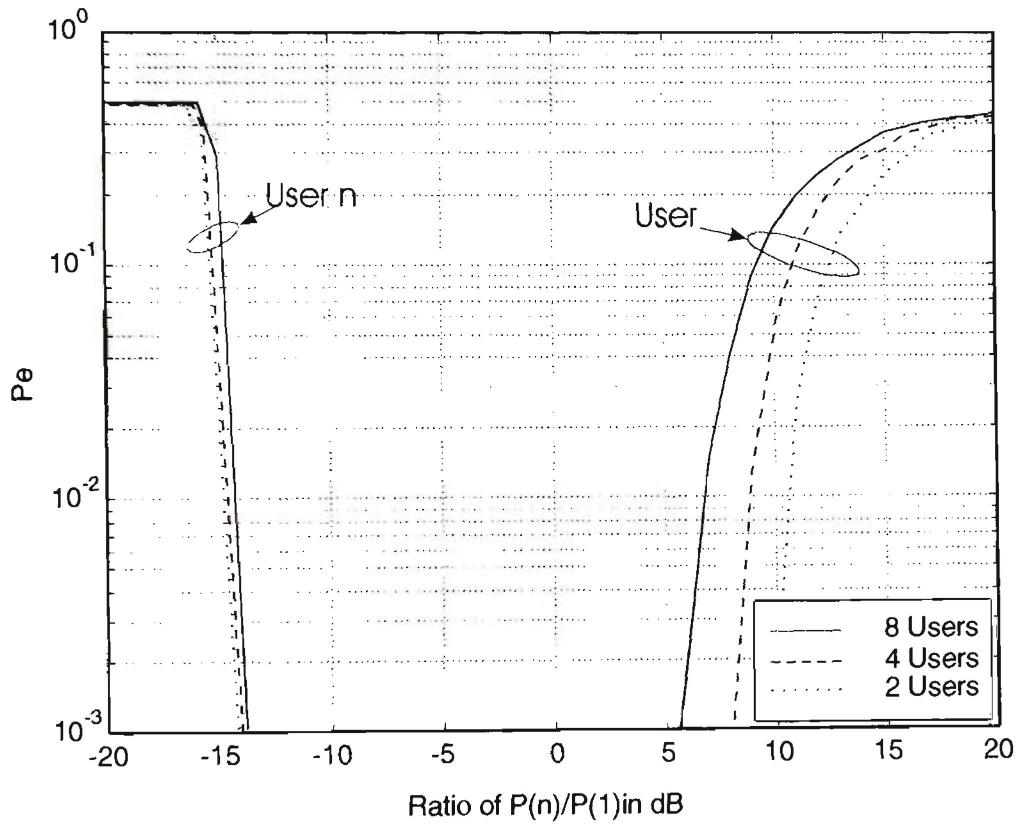


Figure 9. Near Far Effect of a multiuser conventional CDMA system

3 Multiuser Detection: an Overview

A review of the earlier and parallel work regarding multiuser receiver design and multiuser demodulation is presented in this chapter.

3.1 Conventional Receiver

In the conventional matched filter receiver, the decision device, which follows the matched filter, makes one shot decisions, see Figure 10. That is, it estimates the transmitted symbol on the basis of the received signal only in the interval corresponding to that symbol. Detection is not optimum in this approach as the information that relates to overlapping symbols from other users is ignored. Optimum detection of asynchronous DS/CDMA signals requires that the whole received waveforms for all users at the output of the matched filter be observed.

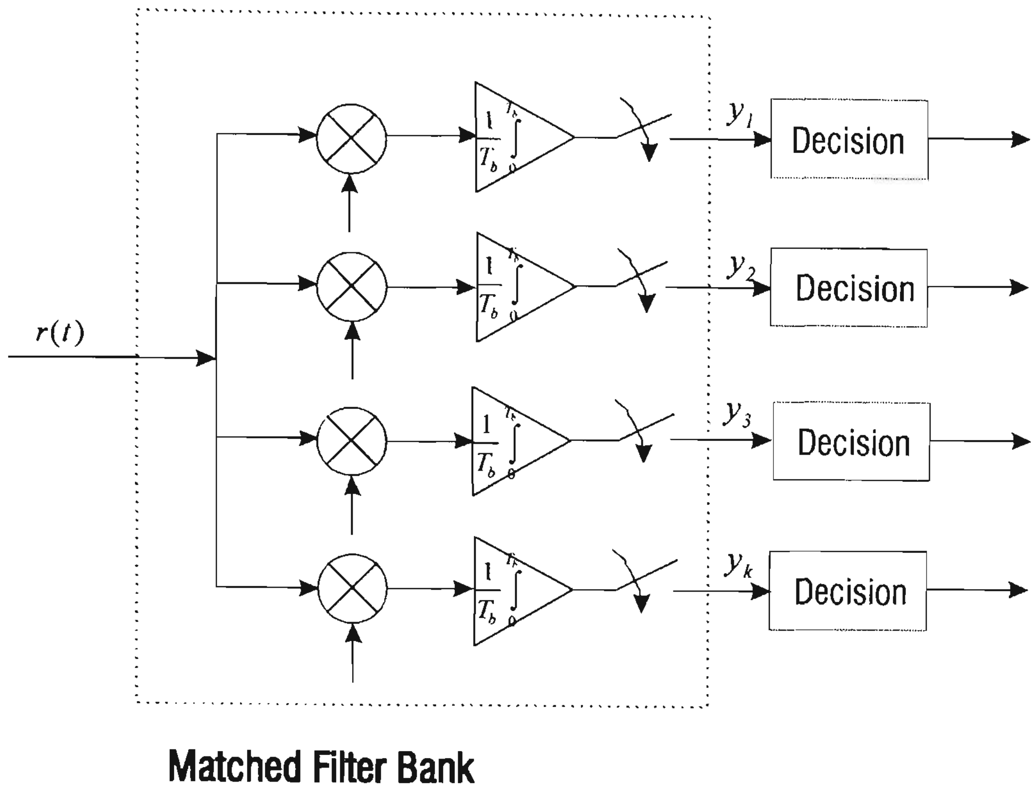


Figure 10. Block diagram of Matched Filter.

3.2 Linear Multiuser Detectors

Linear equaliser type of multiuser detectors processes the matched filter output vector by a linear operation. The performance of the decorrelating detector has been analysed by Verdu and Lupas in [5] and [6]. They show that it has some very attractive properties including the ability to achieve the same Near Far Resistance as the optimum Maximum Likelihood Sequence Detector (MLSD). It is due to these attractive features that the decorrelating detector has been the focus of much research, and a lot of effort has been made to modify the detector so that it can be implemented in a realistic communications environment. Zvonar and Brady in [7] studied the application of the decorrelating detector for frequency selective Rayleigh fading channels; Varansi [8] proposed to use it as the front end of his multi stage detector and Duel-Hallen studied the decorrelating decision feedback multiuser detector [9] .

Decorrelating receivers for quasi-synchronous CDMA systems in AWGN channels without precise delay estimation have been proposed in [10,11,12] and for code acquisition in quasi-synchronous CDMA in [13].

The use of a non adaptive receiver can result in wasted resources and unnecessary computations if only a subset of the possible users is active. In a practical communications environment, the set of users is dynamic as users enter and leave the network. Unfortunately in the presence of a new unknown user the performance of the decorrelating detector is severely degraded [14].

Adaptive decorrelating detectors were studied in [10] for asynchronous systems and implementations for synchronous systems were studied in [15]. Here they developed a simple adaptive decorrelating detector by placing constraints on the set of spreading codes that are to be used by the active users. Both of these papers investigate different methods for updating the receiver parameters to cope with the presence of new unknown users. They take a total systems approach to address the issue of learning and integrating the knowledge of a new transmitting user into the receiver structure. The paper discusses methods for determining the new users spreading codes with the use of training sequences. Blind algorithms are presented as a means of determining the new users spreading codes without the use of a training sequence. The paper also investigates further generalisation of the adaptive decorrelator to accommodate asynchronous transmissions by each of the active users.

Another linear equaliser that has also been researched extensively is the Linear Minimum Mean Squared Error (LMMSE) detector. This receiver takes into account the background noise and utilises the knowledge of the received signal powers. It then attempts to maximise the Signal to Interference plus Noise Ratio of the (SINR) [16]. Centralised LMMSE receivers have been proposed for AWGN channels in [17,18], for fading channels in [19, 20]. The bounds for the NFR and SINR of the LMMSE receiver in AWGN channels have been derived in [21]. An improved LMMSE receiver, less sensitive to the time delay estimation errors, has been proposed in [22]. Receivers suitable for blind adaptation utilising the minimum output energy (MOE) criterion have been studied in [23, 24]. It has been shown that the linear filter optimal in the MOE sense is equal to the linear filter optimal in the MMSE sense [25].

Other techniques for reducing the effects of MAI and Intersymbol Interference (ISI) were investigated by [26]. These include the : *zero forcing block linear equaliser (zf-ble), Minimum Mean Square Error Block, Zero Forcing Block decision Feedback Equaliser and the Minimum Mean Square Error Block Feedback Equaliser.* The authors detail the necessary computations that are required to execute each equaliser. Simulations are performed in a Multipath channel with Rayleigh fading. Observations where made on the system both with and without the application of Turbo Coding. Results showed an improvement in system performance when Turbo Coding was used.

3.3 Adaptive Receivers

Receivers that are based on an adaptive algorithm often have the ability to adapt to their environment and learn about the dynamic and changing nature of the interference and channel.

The convergence of the adaptive algorithms for the LMMSE multiuser receivers has been considered in [27, 28, 29]. A modified adaptive multiuser receiver applicable to relatively fast fading frequency-selective channels with channel state information has been proposed in [30]. Adaptive receivers do not require the signature or timing information of the other users [31].These types of receivers are trained before data is transmitted with a known training sequence and continually adjusts the signature weights during data transmission. Advantages of this type of receiver include timing recovery, multiple access elimination and frequency selective fading suppression.

Adaptive transmitter and receiver structures [32] for asynchronous CDMA systems again assume that the adaptive receiver and transmitter have no knowledge of the signature waveforms of the other users. The concept of an adaptive transmitter is based on feedback information from the corresponding receiver. The information obtained from the receiver is used to calculate the optimum transmitting signature and are adaptively adjusted according to the MSE criterion during the training period and during data transmission.

An adaptive receiver structure that is based on a matched filter followed by an adaptive equaliser is presented in [33]. This structure allows the receiver to adjust to its environment and reduce the effects of interference and noise. The receiver structure is shown to offer a two-fold increase in capacity relative to the conventional receiver with perfect power control.

3.4 Adaptive Narrowband Interference Cancellation Techniques

Spread Spectrum CDMA communications is inherently resistant to NBI which is caused by coexisting with conventional communications systems. It has been demonstrated that the performance of spread spectrum systems in the presence of narrowband signals can be enhanced significantly through the use of active NBI suppression prior to despreading [34]. Not only does active suppression improve error performance but it also leads to an increase in CDMA system capacity [35].

The adaptive notch filter is a technique that has been proposed to notch out or flatten the spectrum of the NBI signal. The adaptive notch filter places notches at the locations of the NBI, so to bring the level of the interference down to the level of the SS signal.

Estimation notch filters using adaptive techniques such as the Least Mean Squared (LMS) have been extensively researched. These filters are based on the knowledge that with wideband SS signals, the past values tend to be uncorrelated with the present or future values. With narrowband signals, the future values are correlated with the past values. Using this knowledge, the interfering signal can be estimated and subtracted out.

Narrow Band Interference rejection capabilities of the fractionally spaced equaliser is investigated by Davis and Milstein [36]. Here they describe an adaptive tapped delay line equaliser that operates in a DS-CDMA receiver, where the taps are adapted to minimise the mean squared error. The overall effect is to whiten the noise and mitigate the effects of both MAI and NBI.

The performance of both optimal and adaptive interference suppression filters for DSSS systems is simulated by Mammela [37]. The simulations include the linear M step prediction and interpolation filters and as well as the LMS, Kalman algorithms. It is demonstrated that linear filters work well if the interference bandwidth is a fraction of the signal bandwidth.

For the case of a sudden parameter jump or new interference, Lee and Lee [38] suggest a gradient - search fast converging (GFC) algorithm. The transient behaviour of the receiver using a GFC adaptive filter is investigated and compared with that of receivers using an LMS or a lattice adaptive filter. They maintain that the GFC is superior for suppressing irregular hostile jamming in DS-CDMA. For better stability, He, Lei, Das and Saulnier [39] discuss the modified LMS algorithm and lattice filter structures comparing their BER performance and convergence characteristics.

Using the minimum mean squared error (MMSE) criterion, Madow and Honig [40] consider interference suppression schemes for CDMA systems. They investigate N tap chip rate filters, the cyclically shifted filter bank and data symbol oversampling. These schemes have the virtue of being amenable to adaption and simple implementation (in comparison to multiuser detection) while at the same time alleviating the near-far problem to a large extent. The channel output is first passed through a filter matched to the chip waveform and then sampled at the chip rate. Due to the complexity and coefficient noise that is associated with systems with large spreading gains simpler structures with few components are proposed.

3.5 Interference Cancellation

Spread spectrum, by its very nature, is an interference tolerant modulation scheme. However, there are situations where the processing gain is inadequate and interference-rejection techniques must be employed. This is especially true for direct sequence spread spectrum, which suffers from the near far problem.

The idea of *interference cancellation* is to estimate the multiple access and multipath induced interference and then subtract the interference estimate from the MF output [41]. The interference cancellation can be derived as an approximation of the (optimum) MLSD receiver with the assumption that the data, amplitude, and delays of the interfering users are known.

3.5.1 Successive Interference Cancellation

The need for a simple but yet effective solution to the problem of multiuser detection (MUD) is presented in [42] by Holtzman. The author discusses a method known as successive interference cancellation as a relatively simple form of MUD and investigates how it can be practically implemented in a real system. Problems that impede its implementation are discussed and solutions for these problems are investigated.

Successive Interference Cancellation (SIC) takes a serial approach to reducing the interference that is caused by the MAI. This is done by recreating separate estimates of the multiple access and multi-path induced interference at the receiver, so that its influence on each user can be subtracted out. This is performed on a user by user basis [43] using either Soft Decision (SD) or Hard Decision (HD) techniques.

The ordering of the powers of the various users is a problem in relatively fast fading channels as they must be updated frequently. SD-SIC has been considered in [44], and HD-SIC in [45, 46]. The SIC for multi-rate CDMA communications has been studied in [47]. The SIC may not yield good enough performance in a heavily loaded CDMA system, where the performance of a conventional receiver is poor. The reason for that is that the SIC is initialised for User 1. If the MAI estimation is poor in the cancellations, the estimation errors will propagate to all users. The SIC has good performance in systems where the powers of the users differ significantly. The effects of delay estimation errors for the SIC has been considered in [48].

Johansson and Svensson [49] investigates interference cancellation in DS-CDMA systems that support multiple data rates. Two methods for implementing multiple data rates are considered, one is the use of mixed modulation and the other is the use of multi-codes. They introduce and analyse a new approach that combines these multiple data rate systems together with a single and multistage non-decision directed interference canceller. The IC schemes that are used by the authors are generalisations and extensions of the successive (serial) IC technique. The analysis was performed in an AWGN and Rayleigh fading channel. Their analysis indicated that the IC schemes used in the mixed modulation or multi-code systems yielded a performance that was close to the single BPSK user bound and gave the prospect of an overall system improvement compared to systems using conventional matched filter techniques.

The performance of an adaptive successive serial-parallel CDMA cancellation scheme in flat Rayleigh fading channels has been studied in [50]. They use a sliding block window ranker and a bank of successive serial - parallel cancellation receivers to reduce the effects of MAI. The serial canceller requires that the users amplitudes be ranked in order of signal strength. The proposed receiver uses a serial canceller followed by a parallel canceller. The authors consider the reverse channel of a single cell CDMA system with perfect power control, the system is simulated in a flat Rayleigh fading channel using both perfect and non-perfect channel estimation.

3.5.2 Parallel Interference Cancellation

Multistage receivers that are based on the Parallel Interference Canceller (PIC) [51], do not require prior knowledge of the cross correlation between the user sequences. During the early stages of cancellation, the PIC can leave some residual interference after cancellation, caused by errors in the initial decisions. One method of improving these decisions is to utilise the error correcting capability of the orthogonal convolutional codes [52]. The problem of multipath fading in a DS/CDMA environment can be overcome by implementing a PIC receiver that uses diversity techniques. The corresponding receivers for slowly fading channels have been studied in [53,54], and for relatively fast fading channels in [55,56,57]. Systems with multiple data rates have been studied in [58].

The application of the HD-PIC to multiuser delay estimation in relatively fast fading channels has been considered in [59]. The SD-PIC receiver has been considered in [60]. The effect of estimation errors on the performance of the HD-PIC receiver has been considered in [61].

An efficient feedback receiver structure for the coherent demodulation of K asynchronous CDMA signals is investigated in [62]. Through the use of feedback, the receiver provides protection for the synchronising loops against the effects of strong interfering signals.

Latva-aho and Lilleberg in [51] investigate Parallel interference cancellation (PIC) based channel parameter estimators for frequency selective fading channels for the uplink in code division multiple access (CDMA) mobile communication systems. They note that the performance of PIC based algorithms depends heavily on the quality of the multiple access interference estimates, which can be improved by using adaptive channel estimation filters. The performance of two adaptive complex channel estimation filters was verified in a fading channel by computer simulations. According to their results, the PIC based adaptive channel estimators outperformed the conventional matched filter method, the successive interference cancellation and decorrelation based adaptive channel estimators.

In [63], the authors describe a multistage PIC for the uplink of an asynchronous coherent DS/CDMA mobile radio system. An important design parameter of this IC is the low processing delay that is required to realise an accurate closed loop Transmission Power Control. Computer simulations reveal that the performance of the two stage PIC can enhance the cell capacity by approximately 2.2 times more than a Matched Filter in a multipath Rayleigh fading channel. The PIC includes moderation factors to control the interference replica to a level that is proportional to the reliability of that replica.

A new flexible multiuser detection scheme for DS-CDMA using multistage detection algorithms and maximum likelihood detection is proposed in [64]. It is based on initialising the multistage detection scheme with several different starting vectors. Using parallel computations the processing delay was able to be limited to reasonable values.

Tanaka et al in [65] analyses the performance of a multistage Interference Cancellation in multi-cell environments. They note that there is some degradation that is caused by not only interference from other users but also from other cells. It was observed that this other cell interference cannot be cancelled out using the MIC. A simulation model of the multi cell environment was constructed and the interference suppression factor, bit error rate and the capacity gain using the MIC was observed.

Kobayashi and Suzuki [66], proposes an interference cancellation method that incorporated a method for estimating the channel in a multipath environment for slotted ALOHA. They performed computer simulations using a frequency selective Rayleigh fading model.

Divsalar et al present an improved non linear PIC that is based on partial cancellation [67]. They show that in the early stages of interference cancellation, where the interference estimate is poor, the tentative data decisions are less reliable than those of the following stages. The authors note that it is preferable not to cancel out the entire amount of the estimated MAI, but only a fraction of it. As the interference canceller (IC) operation progresses, the estimates of the MAI improve and the amount of the "real" interference that is being removed also increases.

Rapport and Shan in [68] investigate the technique of partial parallel interference cancellation further. They compare the performance of both the partial PIC and the standard PIC under various near far effects and show how the BER performance varies with different fractions of cancellation.

4 Multiuser Detection: an in-depth analysis

The detection of W-CDMA signals can be improved by the use of Multiuser Detectors. In multiuser detection, the code (signature) and timing information and in some instances possibly amplitude and phase information of multiple users can be jointly used to improve the reliability of detection for each individual user. Multiuser Detectors are usually implemented at the base station where there is knowledge of all of the users' codes.

There have been many proposed receivers, they range from the optimal receiver structure that was proposed by Verdu; based on Maximum-Likelihood Sequence Detection, to the less complex, sub optimal structures that approximate the optimal receiver. Most of these detectors can be classified into one of three categories. These include:

1. Linear Detectors
2. Adaptive Linear Receivers
3. Subtractive Interference Cancellation
    - 3a. Serial Cancellation
    - 3b. Parallel Cancellation

Linear multiuser detectors apply a linear mapping at the output of the decision device of the conventional detector. This mapping produces new set of outputs which will provide better performance and reduce the effect of MAI that is seen by each user. The adaptive linear receiver uses an error term to control the weights that are applied to the linear FIR filter. The error term and weights are updated every symbol. Subtractive Interference Cancellation relies upon estimates of the users' amplitudes to generate replicas of the interference. This interference term is then subtracted out. Non linear detectors use neural networks or a nonlinear decision device like the hyperbolic tan function, tanh, to detect the spread spectrum signal.

In the remainder of this chapter we will firstly look at the optimum receiver and then briefly analyse each of the detectors that are listed above and finally discuss the advantages and disadvantages of each of them.

4.1 Optimum receiver

The conventional matched filter receiver, estimates the transmitted users signal on the basis of the received signal, only in the symbol interval. In this approach, the detection of desired users symbol is not optimum as the information that relates to the interference coming from the other users overlapping symbols is ignored. Optimum detection of asynchronous DS/CDMA signals, requires observation of the whole received waveform for all users at the output of the matched filter.

For optimum demodulation, it is assumed that the receiver has the information about the signature waveforms for each user as well as knowledge of the time delays, phase shifts and amplitudes.

The minimum error probability receiver must find the most probably transmitted data symbol for all users for all symbol intervals. Each minimisation computes a metric for all possible interfering data symbol combinations. Although a dynamic programming algorithm can be devised to implement the minimum probability of error detector, the required number of operations grows exponentially with the number of users.

The Maximum Likelihood Sequence Detector (MLSD) multiuser receiver minimises the probability of an erroneous decision on the bit vector **b** including the data symbols of all users on all symbol intervals.

Optimum receivers can then be designed to select the bit sequence

$$\hat{b} = \begin{bmatrix} \hat{b}_1(-M) & \cdots & \hat{b}_1(M) \\ \vdots & \ddots & \vdots \\ \hat{b}_k(-M) & \cdots & \hat{b}_k(M) \end{bmatrix} \tag{24}$$

which maximises the conditional probability [1]

$$P\left[\hat{b}\,|\,r(t)\right] \tag{25}$$

If we assume that the transmitted bits are independent and equip probable, maximising the probability in (25) is equivalent to maximising the likelihood function [1]:

$$P\left[r(t)\,|\,\hat{\mathrm{b}}\right] = Ce^{-\frac{1}{2\sigma^2}\int_0^T\left[y_k(t) - \sum_{k=1}^{K}\hat{b}\sqrt{E_k}s_k(t)\right]^2 dt} \quad \text{for } t \in [0,T] \tag{26}$$

where C is a constant and σ^2 is the noise power, $y_k(t)$ is the output of the kth matched filter, E_k is the amplitude of the users' waveform, $\hat{b}$ and $s_k(t)$ are the bit sequence and the signature respectively.

Although a dynamic programming algorithm can be devised to implement the likelihood function in (26), the required number of operations grows exponentially with the number of users. An example of dynamic programming is the Viterbi algorithm. The input to this algorithm is the output samples from the matched filters, it operates on a trellis and its complexity is proportional to 2^{k-1} users'. This algorithm selects a sequence 'b' so that the likelihood function in (26) is maximised.

As can be seen, optimum receivers make decisions by selecting the transmitted sequence to minimise either the sequence error probability or the symbol error probability. In the case of asynchronous DS/CDMA, the maximum likelihood detector consists of a front end matched filter that is followed by the maximum likelihood Viterbi decision algorithm.

4.2 Sub-Optimum Detectors

The optimum receiver is exponential in complexity in the number of users. For example, in a system with 50 users, the number of computations that are required per symbol would be in the order of magnitude of $O(2^{50})$, which is a very high number. For practical implementation this extreme complexity has to be reduced to a reasonable level even if the performance is somewhat degraded from the optimum one. A block diagram of a basic multiuser receiver is detailed in Figure 9.

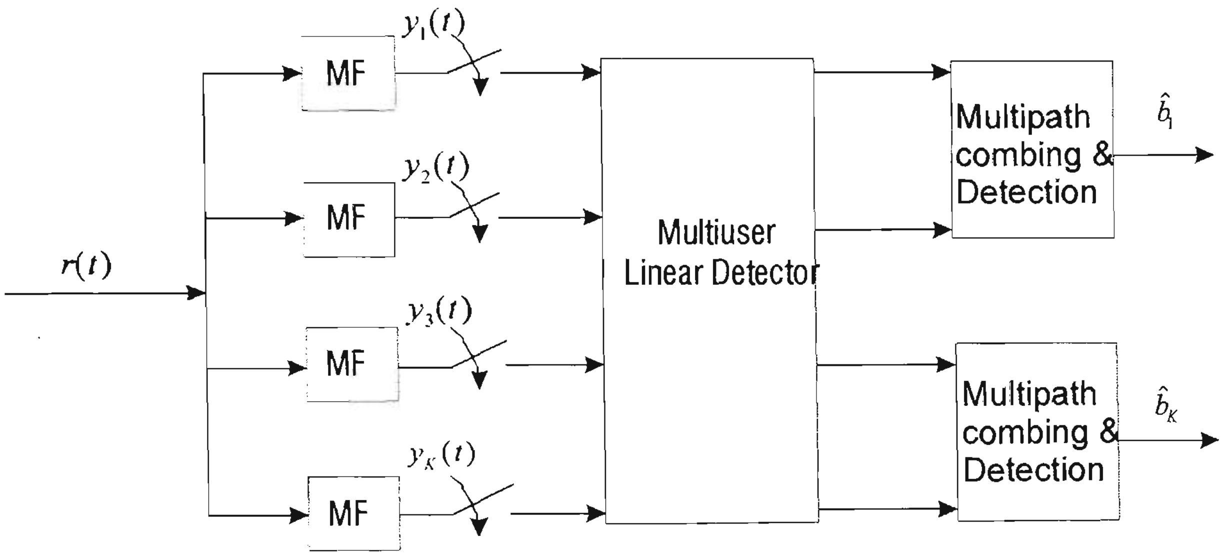


Figure 9. Diagram of a linear multiuser detector.

Multiuser receivers that process the matched filter output by a linear operation are called a Linear Multiuser receiver. The output of these types of receivers are given by the expression

$$L_{lin} = Ly. \tag{27}$$

were y is the output vector of the matched filter, L is the linear operation that is performed by the Multiuser Detector, and L_{lin} is the output of the matched filter. Different choices of the equalisation matrix L yield different types of multiuser receivers. For example, the identity matrix $L = I_{N_b KL}$ will be equivalent to the conventional matched filter.

4.3 The Decorrelating Detector

The optimum receiver achieves low bit error probability at the expense of high computational complexity. When the number of users is large, it is desirable to use a simple but reliable sub-optimum detector. The linear decorrelating detector can significantly outperform the conventional receiver. It is a multiuser receiver that is near-far resistant it has sub-optimum performance and its complexity increases linearly with the number of users. This detector applies a linear transformation, which is based on the inverse of the correlation matrix, $\boldsymbol{R}$, to each output vector of the conventional matched filter in order to decouple the users' data.

$$L_{dec} = \boldsymbol{R}^{-1} \quad . \tag{28}$$

Consider an asynchronous CDMA system that consists of K users, with each user being assigned a signature waveform, $s_k(t)$, where $k = 1 \dots K$, and each signature waveform is restricted to a symbol interval T, and is linearly independent. The input data from each user is a binary sequence, from the symbol alphabet, Ξ . If we assume that the input vector is given by $b = [b_1, \dots b_k]^T$ where $b_k \in \Xi$, then from (19) the output of a conventional matched filter detector for a K user CDMA system can be describes as follows :

$$\boldsymbol{y} = \boldsymbol{RAb} + \boldsymbol{n} \, . \tag{29}$$

Here, $\boldsymbol{R}$ is the crosscorrelation matrix, $\boldsymbol{A}$ is a diagonal matrix that contains the amplitude information of the K users', $\boldsymbol{b}$ contains the users' information bits and $\boldsymbol{n}$ is the noise vector.

Multiplying the output vector $\boldsymbol{y}$ by $\boldsymbol{R}^{-1}$, we get

$$\boldsymbol{R}^{-1}\boldsymbol{y} = \boldsymbol{R}^{-1}(\boldsymbol{RAb} + \boldsymbol{n})$$

$$= \boldsymbol{Ab} + \boldsymbol{z} \, . \tag{30}$$

The term $\boldsymbol{z}$ is a Gaussian noise vector with the autocorrelation matrix $\boldsymbol{R}_z = \sigma^2 \boldsymbol{R}^{-2}$. It represents an increase in the output noise of the decorrelating detector. This level is always greater than or equal to the power that is associated with the noise term that is present at the output of the matched filter. Near optimum performance is achieved when the users signals form a linearly independent set. For the case where there is no system noise, the decorrelating detector achieves perfect demodulation.

The probability that the *kth*, input is recovered correctly [9] is :

$$Pe_k(i) = Q\left(\sqrt{\frac{A_k}{\sigma^2 (R^{-1})_{(i-1)K+k}}}\right) \qquad (31)$$

where A_k is the amplitude of User k.

To mitigate the effects of MAI, the decorrelating detector does not require any amplitude information of any of the users in the system. It uses the information of the users signature waveforms to form the crosscorrelation matrix **R**.

A significant disadvantage of this type of detector is that the computations that are required to invert the correlation matrix **R** are difficult to perform in real time. There have been several suboptimal approaches to implementing the decorrelating detector. Many of them entail breaking up the detector into more manageable blocks so that the matrix inverse may be computed. However, whichever sub-optimal decorrelating detector technique is used, the computational effort that is required is still quite significant.

4.4 Minimum Mean Squared Error (MMSE) Detector

Another detector that performs a linear mapping of the output of the matched filter is the Minimum Mean Squared Error (MMSE) Detector. This detector takes into account the background noise and makes use of the received signal powers of all of the users [16]. The object of this receiver is to minimise the error between the actual output of the conventional matched filter and the soft output of the decision device or the linear mapper [18].

A mapping L is applied to the output y of the conventional detector. The MMSE based detector ensures that the performance criterion, $E\left\{\left(b - \hat{b}\right)^T \left(b - \hat{b}\right)\right\}$ is minimised, where the estimate $\hat{b}$ is given by:

$$\hat{b} = Ly. \qquad (32)$$

In other words this detector forces the expression

$$E\left(\left|b_k(n) - Ly\right|^2\right) \qquad (33)$$

to be zero.

The actual equalisation matrix that is used in the mapping of the output of the matched filter is

$$\boldsymbol{L}_{MMSE} = \left(\boldsymbol{R} + \frac{N_0}{2A^2} \right)^{-1}$$

$$= (\boldsymbol{R} + \xi)^{-1}. \qquad (34)$$

Where $\boldsymbol{R}$ is the cross correlation matrix, N_0 is the received noise power and A is the received signal power. In the absence of noise, the MMSE estimate becomes $\hat{\boldsymbol{b}} = \boldsymbol{R}^{-1}\boldsymbol{y}$. At the other extreme, i.e., when $N_0 >> A$, $\boldsymbol{L}$ reduces to the identity mapping and the MMSE detector reduces to the conventional receiver.

In (34) noise and amplitude information is added to the correlation matrix $\boldsymbol{R}$. This type of detector is similar to that of the decorrelator. In this case it is takes into account the background noise, the amount of modification is directly proportional to the background noise. With this detector, an inversion of **R** cannot be performed without some sort of noise enhancement and performance degradation. Hence the MMSE detector balances the desire to decouple users and completely eliminate any MAI with the desire to keep the effect of the background noise low.

The probability of the *k*th user's i^{th} bit is detected in error is given by [18]:

$$P_k^{MMSE} = \Pr\left\{ [\mathrm{sgn}(\boldsymbol{L}\boldsymbol{y})]_{(i-1)K+k} = -1 \middle| b_k(i) = 1 \right\}$$

$$= \Pr\left\{ \left[\left(\boldsymbol{R} + \frac{N_0}{2A^2} \right)^{-1} n \right]_{(i-1)K+k} \left\} \left[\left(\boldsymbol{R} + \frac{N_0}{2A^2} \right)^{-1} \boldsymbol{RAb} \right]_{(i-1)K+k} \right| b_k(i) = 1 \right\} \qquad (35)$$

Since the noise component $\left[\left(\boldsymbol{R}+\frac{N_0}{2A^2}\right)^{-1}n\right]_{(i-1)K+k}$ is Gaussian with zero mean and variance

equal to the $[(i-1)K+k]$th diagonal element of $E\left\{\left(\boldsymbol{R}+\frac{N_0}{2A^2}\right)^{-1}nn^T\left(\boldsymbol{R}+\frac{N_0}{2A^2}\right)^{-1}\right\}$

$$= \frac{N_0}{2}\left(\boldsymbol{R}+\frac{N_0}{2A^2}\right)^{-1}\boldsymbol{R}\left(\boldsymbol{R}+\frac{N_0}{2A^2}\right)^{-1}, \qquad (36)$$

$P_k^{MMSE}(i)$ can be expressed as a sum of Q functions . From [18], the $[(i-1)K+k]$th diagonal entry of the matrix in (36) can be denoted as $\frac{N_0}{2\alpha}$ and the $[(i-1)K+k]$th row of the matrix $\left(R+\frac{N_0}{2A^2}\right)^{-1}R$ as $[g_1, g_2, \ldots g_{MK}]$, then $P_k^{MMSE}(i)$ can be written as

$$P_k^{MMSE}(i) = 2^{-MK+1} \sum_{b_{k(j)}(\eta(j))\in\{-1,1\}} Q\left(\frac{g_{(i-1)K+k} + \sum_{j\neq(i-1)K+k} g_j b_{k(j)}(\eta(j))}{\sqrt{\frac{N_0}{2\alpha}}}\right). \qquad (37)$$

$\eta(j)$ is the integer part of the ratio of j/K, where K is the number of asynchronous users, $k(j)$ is equal to j mod K. This can be calculated if the mapping L is known.

Due to the fact that this type of detector takes the effect of background noise into account, the overall system performance and probability of error will be better than that of the decorrelating detector. As the background noise converges to zero the performance of this receiver will approach that of the decorrelating detector.

The linear decorrelating detector and the MMSE both have the ability to provide substantial system improvements over the conventional receiver for a CDMA environment. These improvements include the rejection of MAI and an increase in overall system capacity.

The linear operation of the decorrelating detector uses information contained in the correlation matrix to reduce the effects of MAI and as such no amplitude information is required. As the MMSE receiver reduces the effect of the background noise this type of linear receiver requires that the amplitudes or signal powers of the other users' are estimated. One common feature of the MMSE detector and the linear equaliser is that the preliminary estimate of the transmitted bit sequence obtained using the linear transformation $\boldsymbol{L}$ is biased and dependent on the estimation of the signal levels of the interfering users in the system . This complicates the analysis and leads to a performance that depends on the interfering signal power.

As these types of detectors implement a linear mapping at the output of the matched filter, the actual complexity of them is linearly proportional with the number of users. This is significantly lower than that of the optimum detector whose complexity increases exponentially with the number of users.

The MMSE detector that is described above operates on the entire sequence at once. When M is large, as it will almost always be in practice, the resulting detection delay will be unacceptably large. The detector can be modified to incorporate a practical constraint on the size of the delay. This can be implemented by dividing the entire sequence of M bits up into subsequences, by the insertion of reference symbols.

Due to the many attractive features of these types of detectors, these types of receivers have been the focus of much research. Unfortunately they also have several disadvantages including noise enhancement, the estimation of signal powers and the fact that it can be computationally expensive to perform matrix inversion.

5 Adaptive Linear Receivers

In this section the application of the Adaptive Linear Receiver as a multiuser receiver is investigated. To be able to implement this type of receiver, both in a computer simulation and in a Digital Signal Processor, the knowledge of adaptive algorithms is required. In the next section, the Least Mean Squared (LMS) algorithm is analysed and the coefficient update equations are derived. This then forms the basis of the adaptive receiver which is analysed in section 5.5.2.

5.1 Adaptive Filters

Adaptive Filters have the ability to operate satisfactory in unknown environments and conditions and can track time varying input data. They have a broad range of applications, including communications, radar and control. In communication receivers and systems, the adaptive filter finds extensive use in eliminating noise and interference as well as enhancing the desired signal and improving the BER. Interference rejection techniques are often required to be adaptive as they often have the ability to adapt to their environment and they can learn about the dynamic and changing nature of the interference and the channel.

Even though communication systems such as CDMA, which are based on spread spectrum techniques, are inherently resistant to many forms of interference, the introduction of adaptive based interference rejection techniques will provided a further increase in system capacity.

An adaptive filter is a device that is self designing, in that the adaptive filter makes use of a recursive algorithm, which makes it possible for the filter to perform satisfactorily in an environment where complete knowledge of the relevant signal characteristics is not available. The algorithm starts from some predetermined set of initial conditions, representing whatever is known about the environment. In a situation where the environment is stationary, we find that after successive iterations the algorithm converges to an optimum value. In a non-stationary environment, the algorithm offers a tracking capability in which it can track the time varying parameters of the input data, provided that the variations are sufficiently slow.

These filters work by using knowledge of the desired response and information from the input vector to compute an estimation error and converge to an optimum solution in the statistical sense. This is performed by minimising the mean square error value of the error signal, which is defined as the difference between some desired response and the actual filter output. This error is used to adjust the values of a set of filter taps/weights. The error term and filter coefficients are updated periodically, and in most cases after each input sample. As a direct consequence, the parameters of an adaptive filter are updated from one iteration to the next and hence become data dependent. To update the filter weights, well established algorithms like the LMS, RLS or Kalman algorithms are used.

An overview of the LMS Algorithm

The least mean squares (LMS) algorithm is a linear filter that consists of two basic processes

1. A *filtering process* : generating an output of a filter by using a set of tapped inputs and computing an error signal that controls the weights of the filter
2. An *adaptive process* : an automatic adjustment of the tapped weights in accordance with the estimation error.

A significant feature of this algorithm is its simplicity and the fact that it does not require to compute a matrix inversion. A typical block diagram of the LMS filter is shown in Figure 10.

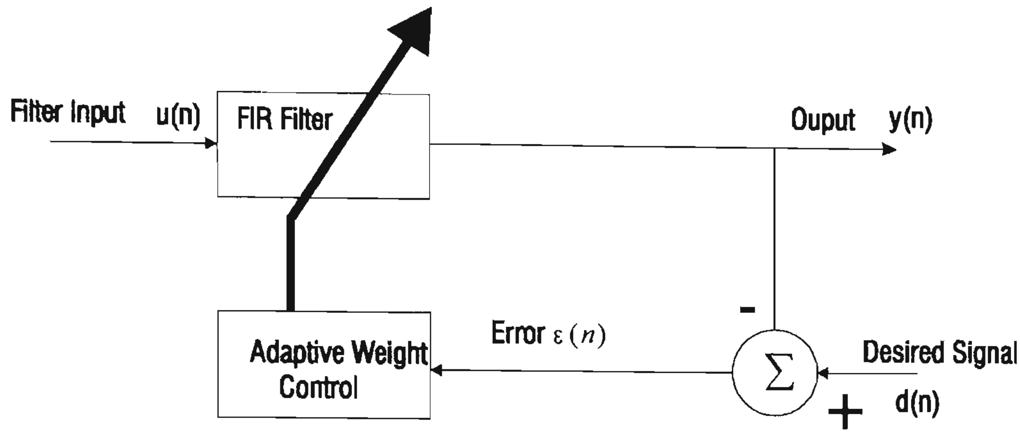


Figure 10. Block diagram of adaptive filter

During the filtering process, the desired response is supplied to the filter algorithm. Using knowledge of this desired response an error term can be computed. The error signal $\varepsilon(n)$ is defined as the difference between the filter output $y(n)$ and the desired response $d(n)$.

The LMS filter is based on a Minimum Mean Square error concept. The adaptive weight control adjusts the coefficients of the Finite Impulse Response (FIR) filter so that $E\left[\varepsilon^2(t)\right]$ is minimised, where $E[\cdot]$ denotes the expectation operator and drives the error term in the direction that makes it small and asymptotes to zero. When this happens the algorithm is said to have converged.

The output of almost all communications channels contain a desired signal, interfering signals and noise. After this signal has been down converted and sampled, it can be represented as :

$$y(n)= \alpha y_d(n)+ \beta y_i(n)+ \gamma z(n) \tag{38}$$

where $y_d(n)$ is the desired signal, $y_i(n)$ is the interference and $z(n)$ is the noise. The effect of the adaptive weight control on this signal is represented by α , β and γ .

If we assume that the desired signal, interference and the noise are all zero mean processes and are uncorrelated with each other, then the cross product terms such as $E\left[y_d(n)y_i(n)\right]$ will be zero. Hence $E\left[\varepsilon^2(n)\right]$ will be a minimum when α is almost unity and β and γ are small. So minimising the minimum mean squared error, $E\left[\varepsilon^2(n)\right]$, corresponds to maximising the signal to interference plus noise ratio (SINR).

It is of interest to determine the optimal weights that yield the minimum mean square error. The output of the filter can be described by :

$$\begin{aligned} y(n) &= \sum_{i=0}^{N-1} a_i(n)x(n-i) \\ &= \mathbf{a}_n^T \mathbf{x}_n, \end{aligned} \tag{39}$$

where

$$\mathbf{x}_n = \left[x(n), x(n-1) \ldots x[n-(N-1)]\right]^T \tag{40}$$

and

$$\mathbf{a}_n = \left[a_0(n), a_1(n) \ldots a_{N-1}(n)\right]^T \tag{41}$$

are the input signal and filter coefficient vectors, respectively, at instant nT .

The mean squared error *(MSE)* function is defined as

$$\Psi(\boldsymbol{a}_n) = E\left[\varepsilon^2(n)\right] \tag{42}$$

where $E[\cdot]$ is the expected value of $[\cdot]$ and the error term $\varepsilon(n)$ can be defined as

$$\varepsilon(n) = d(n) - y(n). \tag{43}$$

The *MSE* can be expanded as:

$$\Psi(\boldsymbol{a}_n) = E\left[\varepsilon^2(n)\right]$$

$$= E\left[\left(d(n) - y(n)\right)^2\right]$$

$$= E\left[d^2(n) - 2d(n)y(n) + y^2(n)\right] \tag{44}$$

We can substitute $\boldsymbol{a}_n^T \boldsymbol{x}_n$ for $y(n)$ to form

$$\Psi(\boldsymbol{a}_n) = E\left[d^2(n) - 2d(n)\boldsymbol{a}_n^T \boldsymbol{x}_n + \boldsymbol{a}_n^T \boldsymbol{x}_n \boldsymbol{x}_n^T \boldsymbol{a}_n\right]$$

$$= E\left[d^2(n)\right] - E\left[2d(n)\boldsymbol{a}_n^T \boldsymbol{x}_n\right] + E\left[\boldsymbol{a}_n^T \boldsymbol{x}_n \boldsymbol{x}_n^T \boldsymbol{a}_n\right], \tag{45}$$

and this can be rewritten as

$$\Psi(\boldsymbol{a}_n) = E\left[d^2(n)\right] - 2\boldsymbol{a}_n^T \boldsymbol{P}_n + \boldsymbol{a}_n^T \boldsymbol{R}_n \boldsymbol{a}_n \tag{46}$$

where

$$\boldsymbol{P}_n = E\left[d(n)\boldsymbol{x}_n\right] \tag{47}$$

and

$$\boldsymbol{R}_n = E\left[\boldsymbol{x}_n \boldsymbol{x}_n^T\right] \tag{48}$$

are the *cross correlation* between the desired and input signals and the *correlation* matrix of the inputs signals respectively at instant nT. It is clearly seen from (45) and (46) that $\Psi(\boldsymbol{a}_n)$ is a quadratic function of the weights or filter coefficients; the output of this function is a parabola or a bowl. The extremum of this quadratic surface is clearly a minimum which is well defined, hence there are no other "local minima".

The weight vector that yields a minimum value of $E\left[\varepsilon^2(n)\right]$ can be determined by taking the gradient of the MSE :

$$\nabla\left[\Psi\left(\boldsymbol{a}_n\right)\right]=\left[\frac{\partial\Psi\left(\boldsymbol{a}_n\right)}{\partial \boldsymbol{a}_0(n)},\frac{\partial\Psi\left(\boldsymbol{a}_n\right)}{\partial \boldsymbol{a}_1(n)},\cdots\frac{\partial\Psi\left(\boldsymbol{a}_n\right)}{\partial \boldsymbol{a}_{N-1}(n)}\right]^T, \tag{49}$$

$$\begin{aligned}\therefore\ \nabla\left[\Psi\left(\boldsymbol{a}_n\right)\right] &= \frac{\partial\left\{E\left[d^2(n)\right]-2\boldsymbol{a}_n^T\boldsymbol{P}_n+\boldsymbol{a}_n^T\boldsymbol{R}_n\boldsymbol{a}_n\right\}}{\partial \boldsymbol{a}_n(n)} \\ &= -2\boldsymbol{p}_n+2\boldsymbol{R}_n\boldsymbol{a}_n.\end{aligned} \tag{50}$$

If (50) is set to zero then it can be rewritten as

$$\boldsymbol{R}_n\boldsymbol{a}_n=\boldsymbol{P}_n\ , \tag{51}$$

hence the optimal weight vector is equal to :

$$\boldsymbol{a}_n=\boldsymbol{R}_n^{-1}\boldsymbol{P}_n\ . \tag{52}$$

Equation (52) is often referred to as the Wiener-Hopf equation. From this result, the minimum mean squared error can be determined as:

$$\begin{aligned}\min\left\{\Psi\left(\boldsymbol{a}_n\right)\right\} &= E\left[d^2(n)\right]-2\boldsymbol{a}_n^T\boldsymbol{P}_n+\boldsymbol{a}_n^T\boldsymbol{R}_n\boldsymbol{a}_n \\ &= E\left[d^2(n)\right]-2\boldsymbol{a}_n^T\boldsymbol{R}_n\boldsymbol{a}_n+\boldsymbol{a}_n^T\boldsymbol{R}_n\boldsymbol{a}_n \\ &= E\left[d^2(n)\right]-\boldsymbol{a}_n^T\boldsymbol{R}_n\boldsymbol{a}_n\end{aligned} \tag{53}$$

The shape, location and orientation of the MSE surface will depend on the signals that are present at the input to the filter. If the number of signals (both desired and interfering) or their power levels change with time, then the bowl and hence $\boldsymbol{a}_n$ will move around the weight plane. The problem of the adaptive filter is to make the weights track the bottom of the bowl. The most popular method of finding an approximate solution to equation (53) is the LMS algorithm. In this algorithm, the weights are controlled according to a gradient algorithm.

It is easily seen that from (53) that the gradient function of $\Psi\left(\boldsymbol{a}_n\right)$ is dependent on both $\boldsymbol{R}_n$ and $\boldsymbol{P}_n$, and implementing these variables is difficult due to the expectation operator and is difficult to implement in a real time processor. Hence it is necessary to estimate them instead. The simplest way of estimating them is to drop the expectation operator. Therefore we can estimate the gradient function by letting

$$\widetilde{\boldsymbol{P}}_n = d(n)\boldsymbol{x}_n \tag{54}$$

and

$$\widetilde{\boldsymbol{R}} = \boldsymbol{x}_n\boldsymbol{x}_n^T \tag{55}$$

be estimates of $\boldsymbol{P}$ and $\boldsymbol{R}$ respectively.

From (38), (42), (49), (53) and (54) we can write the gradient as

$$\begin{aligned} \widetilde{\boldsymbol{g}}_n &= \nabla\left[\Psi(\boldsymbol{a}_n)\right] \\ &= -2d(n)\boldsymbol{x}_n + 2\boldsymbol{x}_n\boldsymbol{x}_n^T a_n \\ &= -2\left[d(n) - \boldsymbol{x}_n^T\boldsymbol{a}_n\right]\boldsymbol{x}_n \\ &= -2\varepsilon(n)\boldsymbol{x}_n. \end{aligned} \tag{56}$$

This is known as the LMS algorithm.

For stationary input signals, the mean-square error in (45) is a second order function. The method of steepest descent will seek the minimum MSE by making each change in the weight vector proportional to the gradient of the MSE with respect to the weight vector. Using the steepest descent update algorithm which is

$$\boldsymbol{a}_{n+1} = \boldsymbol{a}_n - \alpha\widetilde{\boldsymbol{g}}_n \ , \tag{56}$$

we can obtain

$$\boldsymbol{a}_{n+1} = \boldsymbol{a}_n + 2\alpha\varepsilon(n)\boldsymbol{x}_n . \tag{57}$$

In summary the LMS algorithm comprises of three basic equations

1. Filter Output

$$y(n) = \boldsymbol{a}(n).x(n) \tag{58}$$

2. Estimation Error

$$\varepsilon(n) = d(n) - y(n) \tag{59}$$

3. Tap-Weight adaptation

$$\boldsymbol{a}(n+1) = \boldsymbol{a}(n) + 2\alpha\varepsilon(n)\boldsymbol{x}_n \tag{60}$$

A block diagram that implements these equations is shown in Figure 11.

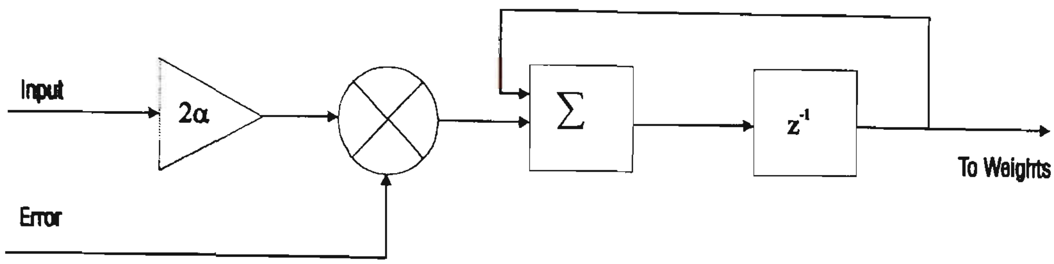


Figure 11. Diagram showing discrete implementation of adaptive filter

5.2 Adaptive Linear Receiver

The Adaptive Linear Receiver is based on the MMSE criterion that forces the error between the output of the decision device and the filter to zero. The receiver contains an MMSE filter which is implemented as an N-tap FIR filter to minimise the MSE between the transmitted and detected symbol, where N is the processing gain. It is this FIR filter that is responsible for performing the despreading of the spread spectrum signal. A decision device at the output of this filter determines the polarity of the recovered symbol; it could be based either on a hard decision or a soft decision, depending on the application. A major advantage of the MMSE based receiver is that explicit knowledge of the interference parameters is not required, as the filter parameters can be adapted to achieve a low minimum mean squared error.

Due to the adaptive structure of the filter and the fact that it is fractionally spaced, the receiver does not require any timing, carrier phase or signature information of either itself or of other users. Before any transmission of data, each user will transmit a training sequence. During this training period the receiver will adjust its weights to so that the error between the reference symbol sequence and the received symbol sequence asymptotes to zero. Not only will be the receiver determine the best weights for each user, it will also find the optimum spread sequence that is required to despread the received signal in the presence of any interfering signals, learn any necessary information about the channel to reduce the effect of channel noise and establish synchronisation.

It is the ability of the adaptive receiver to adapt and learn about their environment, particularly when the environment has time varying parameters that makes it more robust than the matched filter. The linear adaptive receiver uses the sampled signal as the observation vector rather than the output of the matched filter. This is important if the cyclo-stationary nature of the MAI is to be preserved, which is essential for its removal.

In the absence of MAI, the N tap detector reduces to the conventional matched filter. In this case the adaptive receiver will adjust its weights to mitigate the effects of channel noise and multipath fading to accomplish an optimum BER

The complexity of these types of receivers is independent of the number of users and is slightly higher than that of a matched filter. If the time varying parameters of the multipath channel are slower than that of the adaptive algorithms' convergence speed then it will be resistant to multipath fading. The receiver is resistant to the near/far effect and thus does not require strict power control. As a result of being near far resistant, a significant improvement in the capacity over that of the matched filter is achieved.

5.2.1 Method of Operation

This section describes the system operation of the adaptive receiver and the derivation of the LMS update algorithm for the filter coefficients is presented.

A diagram of the adaptive receiver is shown in Figure 12 .

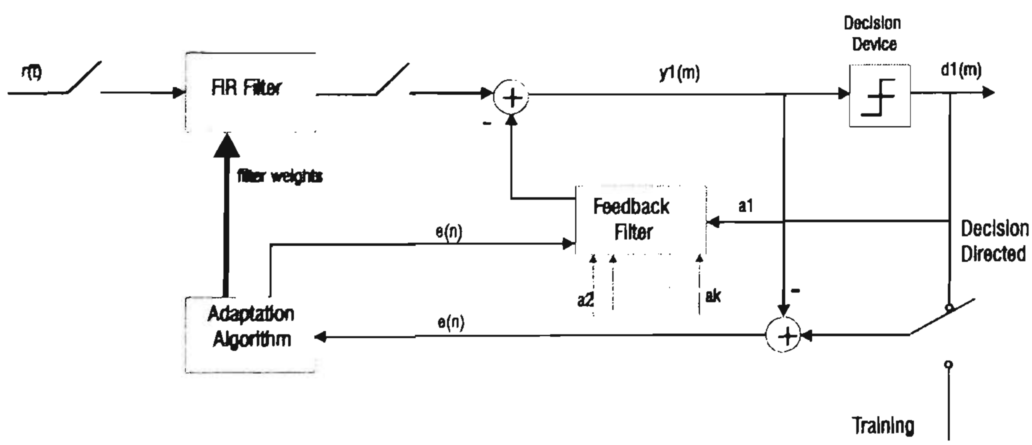


Figure 12. Adaptive Receiver

The received signal is down converted to baseband by a down converter and then filtered. The filtered signal is sampled at a rate of T_s where $T_s < T_c$ and T_c is the period of each chip. The relationship between T_s and T_c is given by $T_s = \frac{T_c}{p}$ where p is the number of samples per chip. As each chip is over sampled, no timing information of any of the users in the system including the desired users is required.

The signal at the input to the adaptive receiver at instant m, is given by

$$r(mT_s) = \sum_{n=0}^{N-1} \sum_{k=1}^{K} b_k(n) A_k s_k(mT_s - nT - \tau_k) + z(mT_s) \qquad (61)$$

The output of the adaptive filter, at time n, is equal to the convolution of the filter coefficients and the received signal this is denoted by $f(nT)$ hence

$$f(nT) = \sum_{q=0}^{Q-1} c(q) r(nT - qT_s) \qquad (62)$$

where $c(q)$ is the FIR filter coefficients and Q is the length of the FIR filter.

The input signal to the receiver will contain the summation of all of the users signals. This can be represented by the diagram in Figure 13. This figure details the wanted user and the interference from the other users' overlapping packets.

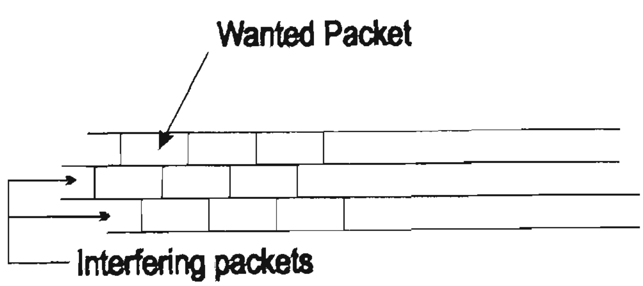


Figure 13. Diagram showing interference from other users symbols

The output of the correlator will contain some residual interference from the other users. To remove it, the feedback filter or data aided interference canceller will cancel this residual interference from the wanted signal. The output of this canceller is obtained by subtracting the decision aided interference estimate from the adaptive FIR filter output. The symbol estimate is given by:

$$\hat{a}(mT) = f(mT) - \boldsymbol{D}_1 \boldsymbol{A}_D . \tag{63}$$

Where $\boldsymbol{D}_1$ is the matrix of the decision aided coefficient sequence and $\boldsymbol{A}_D$ represents the symbols that are already known to the receiver. This estimate, $\hat{a}(mT)$ is applied to the decision device which determines the polarity of the received symbol.

The object of the adaptive receiver is to minimise the mean squared error between the output and the input to the decision device. The input to the decision device is the symbol estimate from the data aided interference canceller.

The minimum mean squared error (MMSE) of the adaptive receiver can be represented mathematically by the following equations:

$$\begin{aligned} \Psi &= E\left[e(m)^2\right] \\ &= E\left[\left|\hat{a}(mT) - a(m)\right|^2\right] \\ &= E\left[\left|f(mT) - \boldsymbol{D}_1 . \boldsymbol{A}_d^T - a(m)\right|^2\right] \\ &= E\left[\left|\boldsymbol{c}^T \boldsymbol{r} - \boldsymbol{D}_1 . \boldsymbol{A}_d - a(m)\right|^2\right] \end{aligned} \tag{64}$$

For the case where there is no feedback, we can rewrite (65) as

$$\Psi = E\left[\left|\boldsymbol{c}^T \boldsymbol{r} - a(m)\right|^2\right] \tag{65}$$

and using the result of (66) and [31], the minimum MSE becomes

$$\Psi_{opt} = \sigma^2 - z^H \boldsymbol{c}_{opt} \tag{66}$$

where $\sigma_a^{\ 2}$ is equal to the power of the symbol $a_k(m)$ given by $E\left[\left|a_k(m)\right|^2\right]$ and z is given by $R\boldsymbol{c}_{opt}$.

The output Signal to Noise Ratio (SNR) in dB of the receiver with K users is given by [31]

$$SNR_{out} = 10\log_{10}\left(\frac{\sigma_a^2}{\varepsilon_K}\right) \tag{67}$$

where ε_K is the MSE obtained for K users, whilst the input SNR is given by [31]

$$SNR_{in} = 10\log\left(\frac{\sigma_a^2}{\sigma^2}\right). \tag{68}$$

5.2.2 The LMS update algorithm for the Adaptive Receiver

The coefficient adjustment algorithm can be described by the following equations and implemented using the LMS algorithm.

FIR filter coefficient update algorithm:

$$c(n+1) = c(n) + \alpha e(n) r(n). \tag{69}$$

Decision aided coefficient update algorithm:

$$D(n+1) = D(n) + \alpha_d e(n) A_D. \tag{70}$$

5.2.3 Convergence properties of the Adaptive Receiver

The mean squared error will converge to a steady state value $\Psi(\infty)$, if and only if the step size parameter satisfies the following two conditions [69] and [31]

Condition 1

$$0 < \alpha < \frac{2}{\lambda_{\max}} \tag{71}$$

Condition 2

$$\sum_{i=1}^{N} \frac{\alpha\lambda_i}{2-\alpha\lambda_i}, \tag{72}$$

where α is the step size and λ_i are the eigenvalues of the multiple access channel correlation matrix.

The convergence speed of the adaptive receiver is dependent on the total input power at the receiver input and the number of users in the system. When the number of users becomes large, an increase in the cross correlation between the signatures occurs and the convergence of the adaptive algorithm in the receiver increases and the overall performance will deteriorate. The length of the spreading code will also effect the convergence time. Obviously the longer the spreading code the greater time the algorithm will take to converge. If a receiver with a faster convergence speed relative to the LMS algorithm is required, alternative algorithms such as the RLS or Kalman filters can be investigated.

Each time when users dynamically enter and leave the system, the communications environment also changes. In each case the receiver will have to be retrained so that it can learn about its new environment.

5.3 Adaptive Receiver Structures

Two adaptive receiver structures are presented in this section: the single user detector and the multiuser receiver.

5.3.1 Single user detection

Single user detection, requires that only one user's spreading code and delay are known at the receiver. Information of the other users' spreading codes, delays and the amplitude are assumed to be unknown. The complexity of the single user detector is generally much smaller than that of multiuser detection.

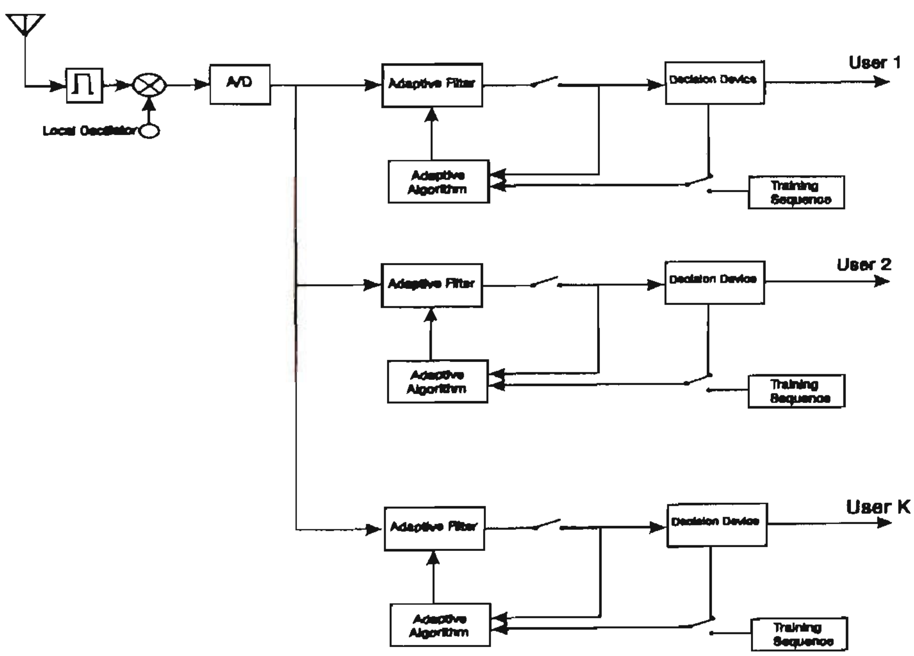


Figure 14. Diagram of a Single User Receiver.

They can be adaptive or fixed. Adaptive techniques allow the receiver to optimise the spreading code in such a way as to mitigate the effect of Multiple Access Interference MAI and hence improve its resistance to the near far effect. The adaptive receiver, whilst having knowledge of only one of the user's spreading code, can be used and utilised in the downlink, to reduce the effect of the other users and aid in the suppression of multiple access interference.

5.3.2 Multiuser detection

With multiuser detection, the knowledge of all of the users' spreading codes is used and information from all of the users is feedback to each adaptive receiver structure to mitigate the effect of multiple access interference.

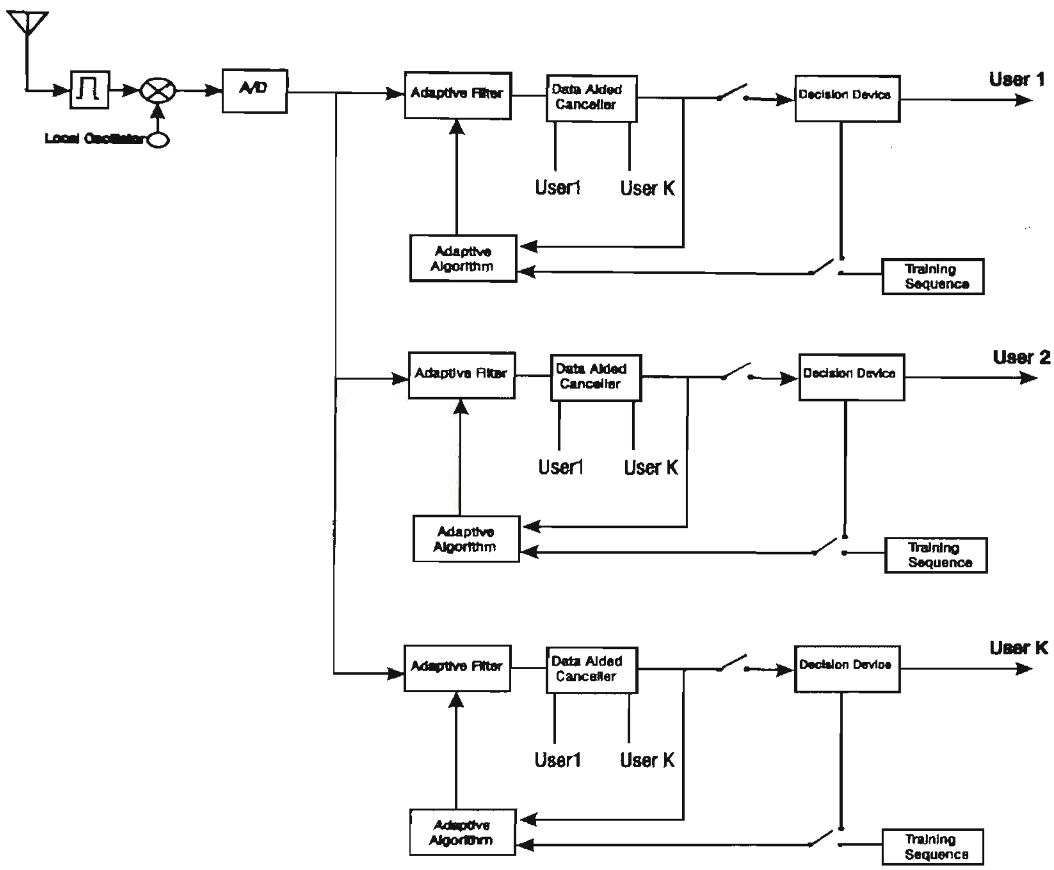


Figure 15. Diagram of an Adaptive Multiuser Receiver.

Cell capacity of a DS-CDMA system is realised by maintaining equal relative amplitudes amongst users at the base station receiver input. Multiuser detection and interference cancellation techniques aid in the relaxation of the specified received power by each user. As a direct result, multiuser receivers help in the maximisation of the system capacity and reduces the demands on power control requirements.

5.4 Adaptive Linear Receiver performance analysis and results

The effect of MAI can be removed to a great extent by the use of an adaptive receiver.

In this section the results of the adaptive linear receiver are presented. The adaptive filter uses an LMS algorithm to adjust its weights so that the output MMSE was reduced. Each user in the CDMA system had its own adaptive receiver that was trained to the users spreading code.

We begin by simulating the convergence of the adaptive receiver under various conditions. These include a system with the same power levels and a system with different signal levels.

5.4.1 Convergence of the Adaptive Receiver

To achieve a low MMSE, the rate of convergence of the adaptive algorithm is quite important especially in a real time system. Here different users enter and leave the system randomly and there are usually differences in the received signal strengths; this leads to the near far effect. It is important for any adaptive receiver to adapt quickly to these changes. During the training period the receiver learns about the type of environment that it is operating in. This would include information about its own channel as well as other users.

To gain a brief insight into this problem the convergence of the error was plotted and observed for different conditions.

- i Figure 15 details 1 user, 5 users and 15 users operating with the same power levels.
- ii. Figure 16 details 4 users with different power levels.
- iii. Figure 17 details 31 users operating with the same power levels.

Both the analytical and experimental results for the rate of convergence of the LMS algorithm and the other adaptive algorithms are discussed extensively in [69]. They conclude that the mean square error e of the adaptive algorithm will converge to a steady state value if the step size parameter satisfies the two conditions that are specified in (72) and (73).

Equations (72) and (73) can be rewritten as [31]:

$$\alpha < \frac{2}{\sum_{i=1}^{2M+1} \lambda_i} < \frac{2}{P_t} \tag{74}$$

Where λ_i is the eigenvalues of the correlation multiple access channel matrix, $2M+1$ is the number of filter coefficients and P_t is the total power at the receiver input.

It can be seen that from (74) that the rate of convergence is limited by power of each user. If we observe the results from Figure 15 it can be clearly seen that the convergence of the adaptive algorithm is also limited by the number of users in the system. When one user is present in the system, it takes in the order of 10's of samples to achieve a low Mean Squared Error (MSE), but as the number of users increases we can observe an decrease in the rate of convergence. For five users this increases to 100 samples to achieve an MSE of 0.1 whereas fifteen users takes up to 200 samples to achieve approximately the same MSE as that for the five user case.

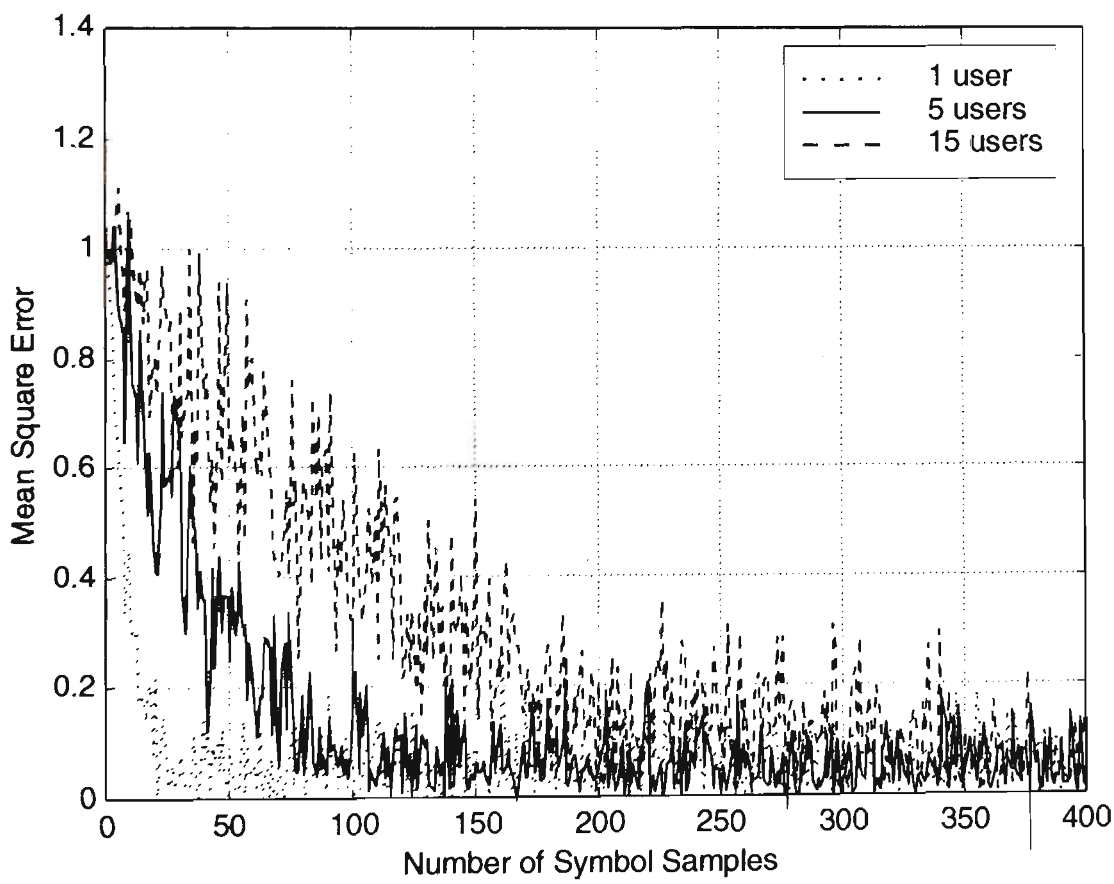


Figure 15. Convergence of MSE Vs Samples for an Adaptive CDMA Receiver

To observe the effects of convergence of the adaptive linear receiver with different received signal strengths, a system was setup with four users, and the power of each user was as follows:

User	Transmitted Power Level dBm
User 1	34
User 2	18
User 3	10
User 4	7

Table 2 Power Levels of the four users in the simulation of Figure 16.

The results of this simulation are shown in Figure 16 below.

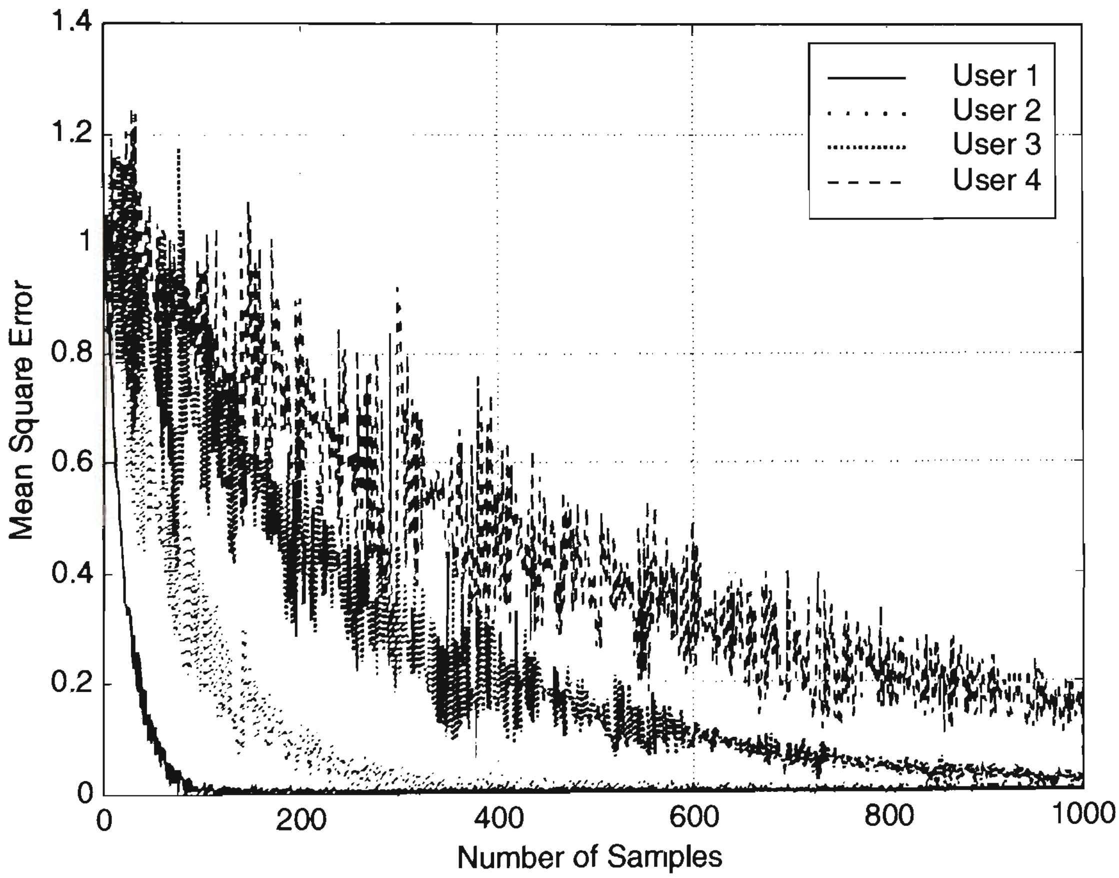


Figure 16 Convergence of Adaptive CDMA Rx with 4 users different power levels

It is quite clear that from this simulation as well as Equation (89) that the convergence rate of the LMS algorithm is indeed dependent on the signal strength of the users. For the case with User 4 it can be observed that it takes 800 samples for it to converge, whereas User 1 has a faster convergence time and has an amplitude that is seven times the size of that of User 4.

As the number of users approaches the theoretical limit of the system; in this case 31 users for a spreading code length of 31, it will converge but some noise will be present and this is illustrated in Figure 17.

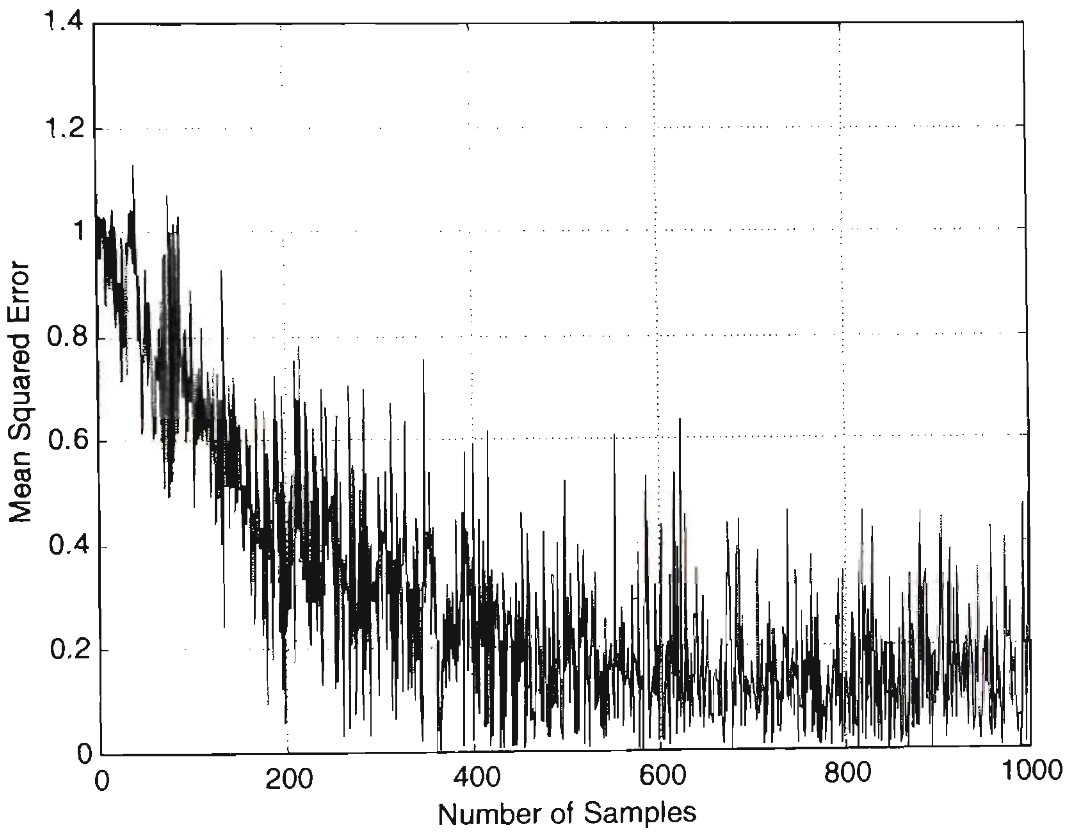


Figure 17 Convergence of 31 users, with the same power level

In a real time system, the system is dynamic and this means that the number of users increases and decreases as new users enter and leave the network. In these situations the algorithm must be able to handle the dynamic nature of the network and as such the receiver must converge to an acceptable level of error in a reasonable time. If it cannot converge, the performance of the receiver will be worse than that of the conventional matched filter. The convergence speed of the adaptive receiver is an important parameter as it determines not only the tolerable multipath fading rate but it gives an indication of how many users can be present at any one time and what sort of differences in power the algorithm can handle at the receiver antenna input at any one time.

5.4.2 Performance of Adaptive Receiver in Near Far Environment

In a Spread Spectrum environment that uses Code Division Multiple Access techniques to allow multiple users to access the network, strict power control is used so that the users arrive at the antenna input at the same signal strength. In a real practical environment the signals strengths of the users at the receiver input will vary 2dB between the weakest and strongest user.

To allow an increase of users it is important that the users arrive at the receiver input with no more than 2dB difference between the weakest and strongest user. In some instances this will prove to be a very difficult task to perform. This stringent requirement can be overcome through the use of a Near-Far resistant receiver.

The adaptive linear receiver is NF resistant and it can be used to relax these stringent requirements. Simulations show the NF resistant properties of the adaptive linear receiver and how well it performs in such environments.

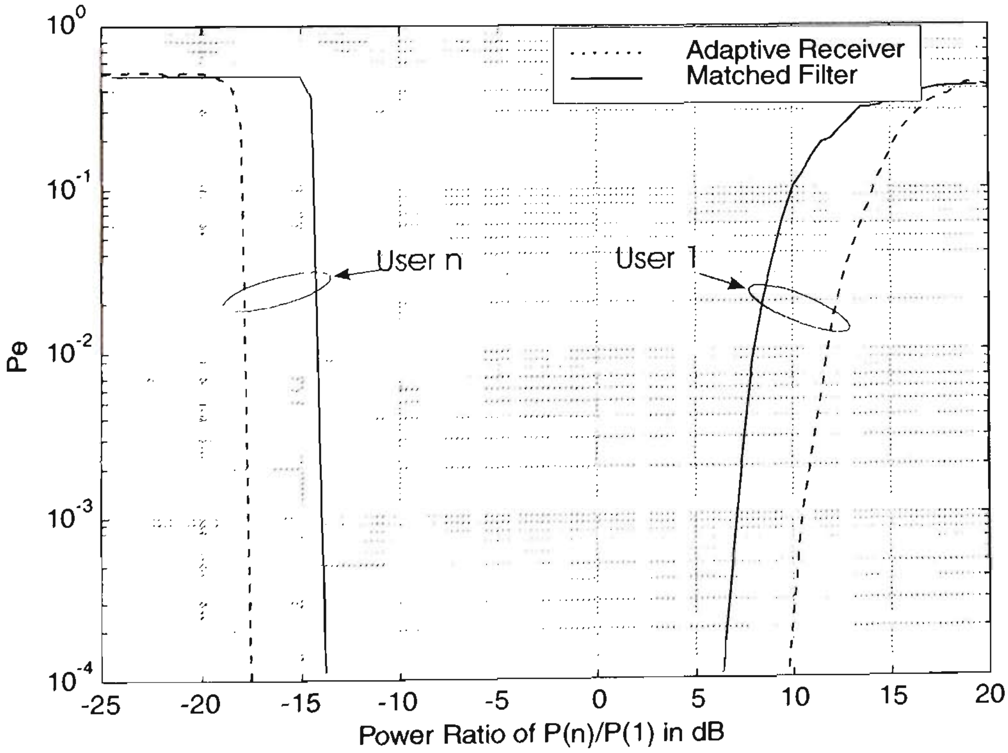


Figure 18. Near Far Effect of 4 users in an adaptive CDMA Receiver

Figure 18 details the Near Far Effect in a 4 user CDMA system using an adaptive receiver. From this graph it can be observed that there is an improvement of approximately 4dB over the standard matched filter receiver. At 10^{-3} BER the receiver can still receive and despread signals that are present with hostile signals that are 18dB greater than its own signal level (interfering signals are near) and at 10dB when the interfering signals are far.

The table below summarises the performance of both the adaptive receiver and the conventional matched filter.

	Wanted Signal : Far			Wanted Signal : Near		
	Power Ratio at 10^{-3}	Amplitude of user 1 (dBm)	Amplitude of users 2 to 4 (dBm)	Power Ratio at 10^{-3}	Amplitude of user 1 (dBm)	Amplitude of users 2 to 4 (dBm)
Adaptive Rx	-18dB	32	0.5	10dB	0.05	0.5
Conventional Rx	-14dB	12	0.5	7dB	0.1	0.5
Improvement	4dB or 2.5 times			3dB or 2 times		

Table 3 Performance of the 4 user CDMA system for both Adaptive and Matched Filter Receivers.

6 Interference Cancellation

The basic principle underlying the interference Canceller is the recreation of separate estimates of the multiple access and multipath induced interferences, so that its influence on each user can be subtracted out from the received signal. There are two main structures that are used for subtractive interference cancellation. The interference can be cancelled simultaneously from all users leading to *parallel* interference cancellation (PIC), or on a user by user basis leading to *successive (serial)* interference cancellation (SIC), so that in each stage, the remaining users see less and less MAI. As the PIC has minimal throughput delay, its processing time will be much faster than that of the SIC at the expense of hardware complexity. Such detectors are often implemented with multiple stages where the expectation is that the decision will improve at the output of each successive stage and a significant performance improvement over the conventional MF receiver will be obtained; this will be reflected in improvements in the overall system capacity and reliability.

For practical communications systems, the interference cancellation receiver appears to be a realistic solution for the multiuser detector. They have lower complexity than the more optimal schemes such as the decorrelator and the MMSE based receivers and they can be applied to systems with long spreading codes in which an adaptive MMSE type detector will be difficult to apply.

Since subtractive interference cancellation receivers require the information of spreading code, timing, amplitude and phase of all of the users in the system, they can only be implemented in the base station and as such they cannot suppress out of cell interference in a multi cell environment.

6.1 Parallel Interference Cancellation

Parallel Interference Cancellation receivers have multiple stages of interference estimation and cancellation. It is based on simultaneously processing all of the users in parallel, calculating the contribution of MAI from each user and then removing it from the received signal. This process can be repeated for several stages. At each stage, better estimates of each user are produced, allowing more effective interference cancellation. As the interference cancellation is performed in parallel for all users, the processing delay required to complete the operation, is at most, a few bits at a time and is proportional to the number of stages that are implemented. The diagram of a typical parallel interference cancellation system is shown in Figure 19.

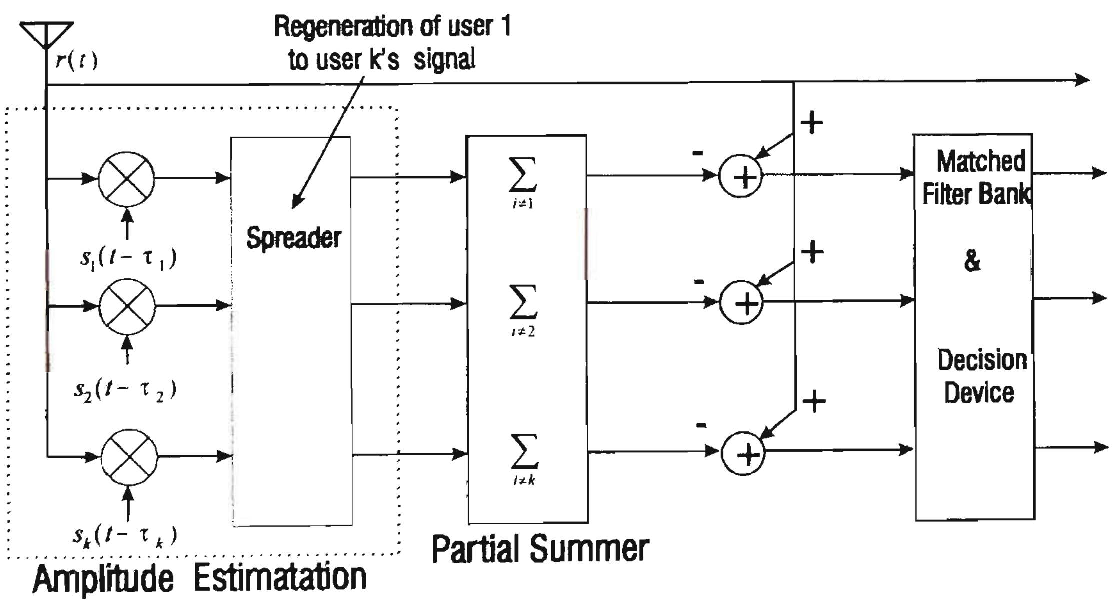


Figure 19 A Parallel Interference Canceller.

We can redraw the PIC to show the inter-connectivity between the various stages, and can observe that as the number of users increases, the inter-connectivity between the various stages increases in a linear manner. This structure indeed shows that the interference of the other users is summed and then subtracted out from the received signal. This is illustrated in Figure 20.

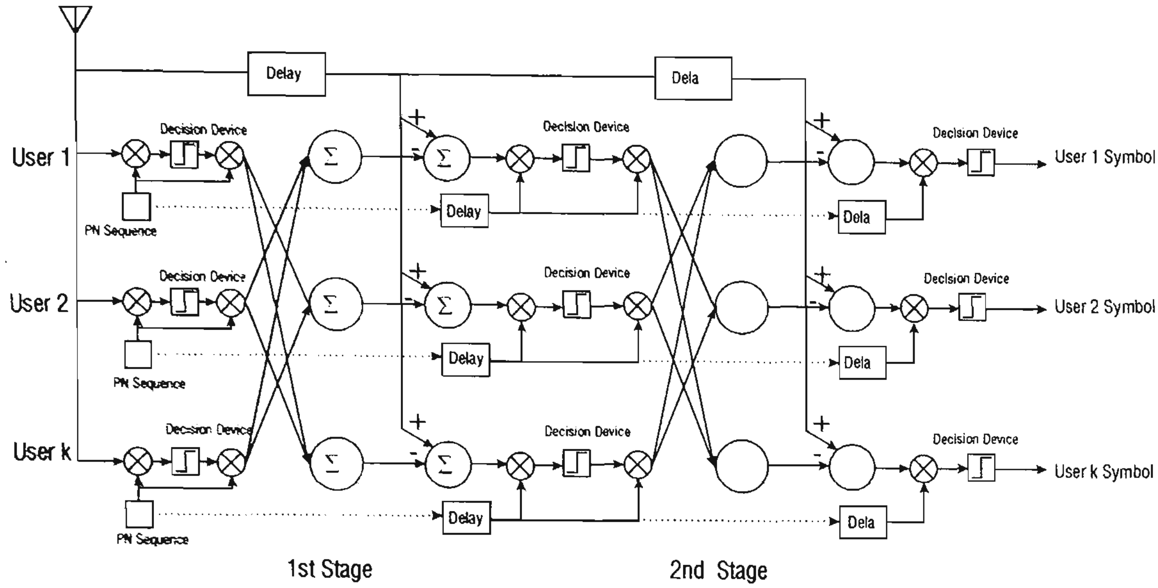


Figure 20 A block diagram showing inter-connectivity of the PIC.

Parallel interference cancellation is a simple yet effective method of parameter estimation in the presence of MAI. It incorporates some of the known methods of data detection, channel coefficient estimation and as well as delay tracking. To successfully remove the interference that is produced through multiple users accessing the channel, the PIC requires information of all of the users' spreading codes, timing, channel parameters and amplitude information. Here we can see in Figures 19 and 20 that the matched filter is used to estimate this information. A decision is then made on the polarity of the symbol, after which it is respread and then partially added. This forms the basis for the estimated MAI for each user in the system. The estimated MAI for each user can be improved by replacing the matched filter receiver with the decorrelating receiver, the linear MMSE receiver or the Adaptive MMSE receiver.

6.1.1 A model of the PIC receiver

From (9), the received signal is given by :

$$r(t) = \sum_{n=0}^{N_b-1} \sum_{k=1}^{K} A_k b_k(n) s_k(t - nT - \tau_k) + n(t), \tag{75}$$

where A_k is the amplitude of User k's signal , n the users' symbol index, s_k is User k's spreading code, τ_k is the random time delay associated with the kth user and n(t) is Additve White Guassian Noise (AWGN).

The received signal is passed through the matched filter bank. At the output of the matched filters we have:

$$z_k(n) = \int r(t).s_{k_{rep}}dt$$

$$= \sum_{n=0}^{N_b-1}\sum_{l=1}^{K}\left\{\int_{nT+\tau_k}^{(n+1)T+\tau_k} A_l b_l(n)s_l(t-mT-\tau_l)s_{k_{rep}}(t-nT-\tau_k)dt\right\}+v(t), \tag{76}$$

$$= \sum_{n=0}^{N_b-1}\sum_{l=1}^{K} A_l b_l(n)\int_{nT+\tau_k}^{(n+1)T+\tau_k} s_l(t-mT-\tau_l)s_{k_{rep}}(t-nT-\tau_k)dt+v(t)$$

where s_{rep_k} is the replica of User k's spreading sequence at the receiver.

Output $z_k(.)$ is then respread to form a replica of User k's transmitted signal:

$$w_k(n) = z_k(n).s_{j_{rep}}(t-nT-\tau_k). \tag{77}$$

The output of the jth stage of the PIC receiver is:

$$PIC_j^k = r(t) - \sum_{\substack{l=1 \\ l\neq k}}^{K} w_l(n), \tag{78}$$

The desired users' data are obtained by multiplying this new estimate by the locally generated spreading sequence:

$$d_k(n) = \int_{nT+\tau_k}^{(n+1)T+\tau_k} PIC_j^k.s_{rep_k}(t-nT-\tau_k)dt. \tag{79}$$

The above steps are carried out in each stage and normally several stages need to be accomplished before an acceptable result can be achieved.

Parallel interference estimation utilises tentative data decisions. This scheme is called hard decision interference cancellation (HD-PIC). If tentative data decisions are not used, the scheme is then called soft decision interference cancellation (SD-PIC). The interference cancellation process iteratively improves the interference estimates.

The decision directed algorithm applies hard tentative decisions of the desired user's signal to generate the interference replica. This method requires the estimation of the channel parameters and a coherent detection scheme. The non decision directed algorithm uses the output of the correlated receivers to generate the interference replica, and does not need channel parameter estimation. Non decision directed IC schemes have a simpler structure compared to that of the decision directed IC scheme. However, since this type of canceller cannot utilise the advantage of RAKE path diversity in the replica generation, it requires that a relatively large number of stages be used to achieve a sufficient performance, which results in increased capacity.

A hybrid successive and parallel configuration or a hybrid decision directed and non decision directed algorithm have been reported to mitigate the respective drawbacks [70].

6.2 Partial Parallel Interference Cancellation using Soft Detection

The performance of PIC based algorithms depends heavily on the quality of the multiple access interference estimates, which are formed by using knowledge of the channel coefficients and estimates of the data and timing delays of all the users. Degradations in any of these parameters impairs the performance of PIC based algorithms, this is highlighted in the inability of the PIC based detector to reproduce accurate estimates of the MAI caused by an incomplete knowledge of all of the users parameters.

In practical applications, the MAI estimates are used due to the lack of an exact knowledge of MAI. By introducing multiple stages of estimation, the MAI estimates can be improved in an iterative way. However, this is not always true for a conventional multistage PIC receiver, where at the earlier stages a poor estimate of the MAI, especially when the BER in the previous stage is sufficiently high, will lead to an unsatisfactory cancellation of the MAI. This will result in an increase in the interference power, hence causing further degradation in the latter stages of the PIC receiver.

An improvement to the conventional PIC receiver can be made by taking into account that the tentative decisions at the earlier stages are less reliable than those in the following stages and let the earlier stages of cancellation be more conservative. This means that only part of the estimated MAI will be subtracted out during the initial stages of the PIC receiver.

This idea was originally proposed by by Divsalar, Simon and Raphaeli in [67], where they investigated the design of a improved parallel interference canceller for CDMA systems using the partial interference cancellation technique. In this type of receiver the initial estimate of the interference is not accurate, so instead of removing it completely at the first stage, it is only partially removed. Hence in the early stages of cancellation it is not preferable to cancel out the entire amount of estimated MAI. As the operation progresses, the estimates of the MAI improve and thus in the later stages of the PIC it becomes more desirable to increase the amount of interference being removed by increase the weight. The inter-connectivity of this type of receiver is shown below.

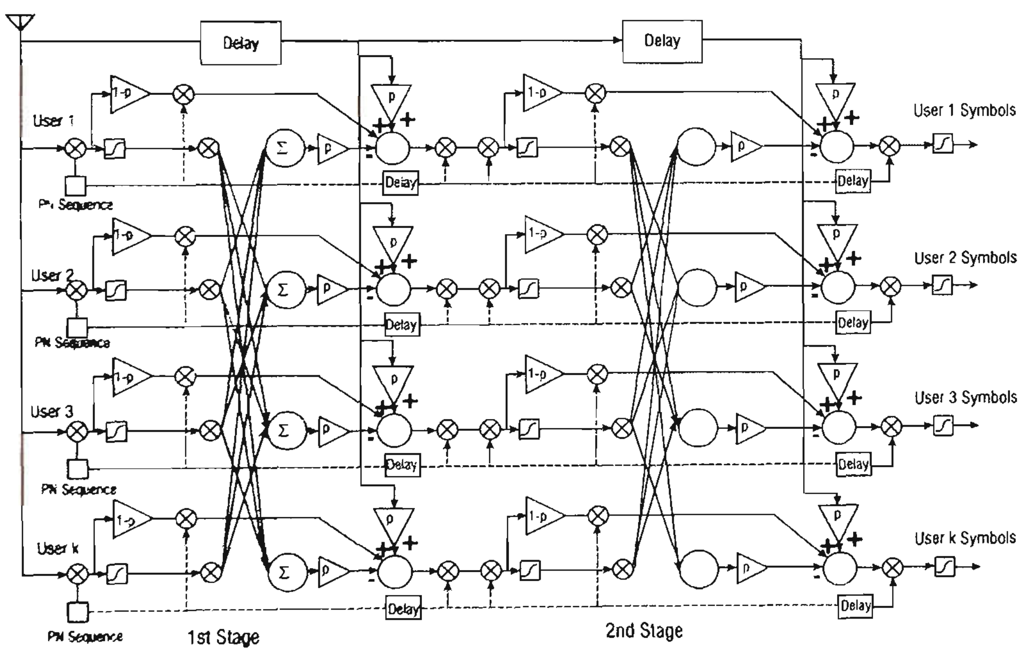


Figure 21 Interconnectivity of the Partial Parallel Interference Canceller.

By including information of the current symbol (bit) and feeding this forward, the overall IC technique can be improved [67]. Information of this bit is fed from the additional signal path from the previous section (just before the decision device). This signal path, of User k, has a gain component in it $(1-\ \rho)$ to control the amount of signal from this user that is being added to partially estimated MAI and the received signal.

Like most IC algorithms it is necessary to estimate the power and delays of each user in the system. The amplitude of the correlator output will provide an estimate of the received signal strength (from the square of the amplitude) of the user. The estimate will contain noise, but its value will be sufficiently accurate.

Instead of feeding the partial weights forward, as per Figure 22, the same result [68] can be obtained if the received signal is scaled by a factor of $\frac{1}{\rho_i}$ where ρ_i is the partial weight of the *ith* stage. The standard PIC can be considered a special case of the partial PIC receiver where ρ_i is equal to 1.

From (78), the output of the jth stage of the partial PIC can be rewritten as:

$$PIC_j^k = \frac{1}{\rho_j}.r(t) - \sum_{\substack{l=1 \\ l \neq k}}^{K} w_l(n) \tag{80}$$

Initially the received signal is detected with a standard CDMA correlator that is implemented as a bank of matched filters for all users. Using the bit decisions and amplitude estimates from the previous stages, the users signals are regenerated with the known spreading codes. For each user, the MAI is then estimated by summing all of the other users signals. The MAI estimates are then subtracted from the original received signal that is scaled by the fraction $\frac{1}{\rho}$ and then finally fed to the matched filter based detector for an improved decision for each user. Errors in the amplitude estimation and the error of the previous decision will contribute to the MAI estimation error.

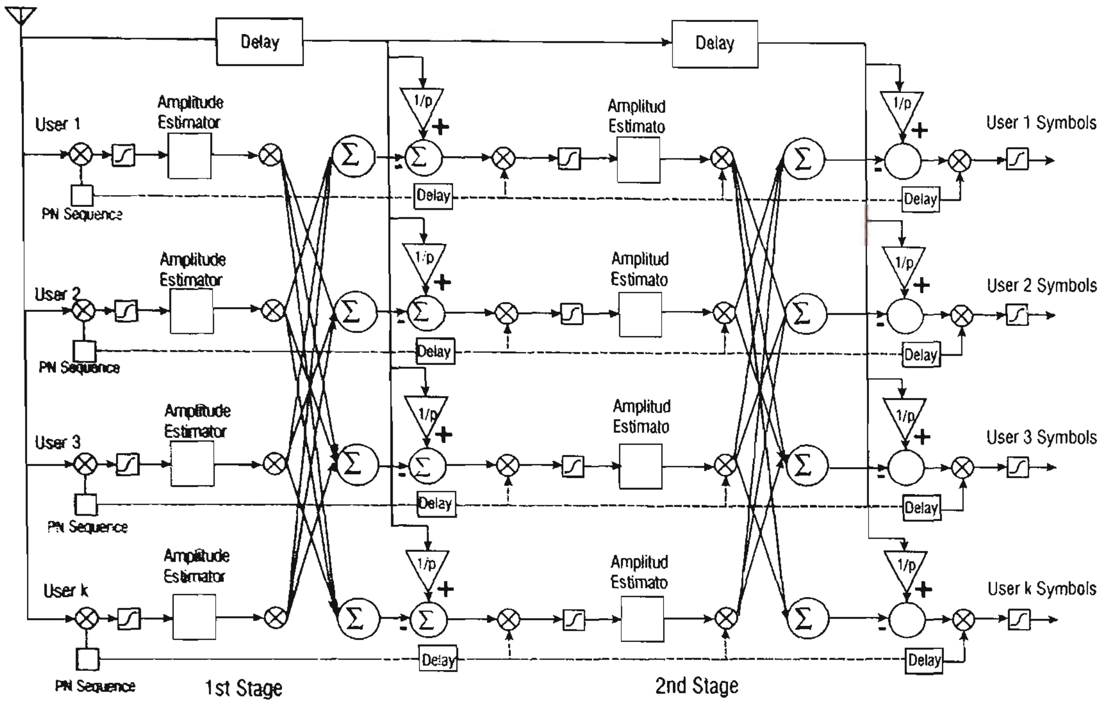


Figure 22. Block diagram of the partial parallel interference canceller.

6.2.1 Synchronisation Errors in the partial PIC

In any practical asynchronous CDMA communications system, perfect chip synchronisation may not be possible [4]. Synchronisation errors will affect the performance of the receiver and will result in an increase of errors. In a real system, the time delays, carrier phase and frequency are subjected to some sort of estimation error, no matter how accurate the acquisition and tracking mechanism is. Timing misalignment affects interference cancellation in two ways [61]. First, the correlation between the desired user's signal and the locally generated spreading code is imperfect, resulting in reduced received signal power. Second, the cancellation will be imperfect due to the time offset between the actual interfering signal and the estimated signal.

During the reception of information the receiver can also loose synchronisation, this can vary from a fraction of a chip up to one chip.

The received signal was given in (75) and the locally generated spreading code at the receiver can be expressed as:

$$s_{k_{rep}} = s_k(t - nT_b - \tau_k + \varepsilon_k), \qquad (81)$$

where $0 \leq |\varepsilon_k| \leq T_c$ represents the timing error associated with User k and Tc is the period of each chip.

At the output of the matched filters we have:

$$z_k(n) = \int s_{rep_k} . r(t) dt$$

$$= \sum_{m=0}^{M-1} \sum_{l=1}^{K} \left\{ \int_{nT+\tau_l}^{(n+1)T+\tau_l} \left(A_l b_l(n) s_l(t - mT - \tau_l) + n(t) \right) dt \right\} . s_{rep_k}(t - nT - \tau_k + \varepsilon_k)$$

$$= \sum_{m=0}^{M-1} \sum_{l=1}^{K} A_l b_l(n) \int_{nT+\tau_k}^{(n+1)T+\tau_k} s_l(t - mT - \tau_l) s_{rep_k}(t - nT - \tau_k + \varepsilon_k) dt + v(t). \quad (82)$$

Output $z_k(.)$ is then respread to form a replica of User k's transmitted signal:

$$w_k(n) = z_k(n) . s_{k_{rep}}(t - nT - \tau_k + \varepsilon_k). \quad (83)$$

The output of the jth stage of the partial PIC was given in (80) and is repeated here for convenience.

$$PIC_j^k = \frac{1}{\rho_j} . r(t) - \sum_{\substack{l=1 \\ l \neq k}}^{K} w_l(n) \quad (84)$$

Here ρ_j is the fraction of cancellation of the partial PIC. The desired users' data are obtained by multiplying this new estimate by the locally generated spreading sequence:

$$d_k(n) = \int_{nT+\tau_k}^{(n+1)T+\tau_k} PIC_j . s_{rep_k}(t - nT - \tau_k + \varepsilon_k) dt \quad (85)$$

Existing methods, reported by Divsalar et al [67] and Rapaport [68] have used constant weights for each section and for all of the users, however this does not take into account that the MAI for each user will be different and uncorrelated with each other. A better method [71] that considers the multipath fading effects of the channel and uses a set of weights that reflects the reliability of the bit estimations from the previous stage. Here an adaptive LMS algorithm is used to minimise the mean squared error between the received signal and the weighted sum of the estimates of all of the users' signal during a bit interval with respect to the weights.

6.2.2 Performance of the Partial PIC in presence of time offset errors

In this section the performance of the Partial Parallel Interference Canceller is investigated in a near far environment. The effects of time offset errors are examined and compared against that of the standard Parallel Interference Canceller. The scaling factors that were used in the partial PIC were 0.65 for the first stage and 0.85 in the second stage based on the results in [5]. Four users are assumed, 3 of which have the same SNR. The 4th user represents a user that is near and far; its power is swept over a large range of values. At the receiver, ten samples per chip were introduced into each of the users signals to simulated the effect of the A/D converter. This enabled synchronisation errors to be introduced into the system by offsetting the locally generated PN sequence at the receiver by fractions of a chip.

Shown in Figure 23, is a system which is simulated with no tracking errors. Here a near far ratio, *Pi/P4*, of -21dB can be tolerated for a BER of 10^{-3} for the partial PIC and approximately -18dB for the standard PIC. When the situation was reversed, a power ratio of +14dB was required by the partial PIC for a BER of 10^{-3}. The standard PIC required approximately +11dB to achieve a BER of 10^{-3}.

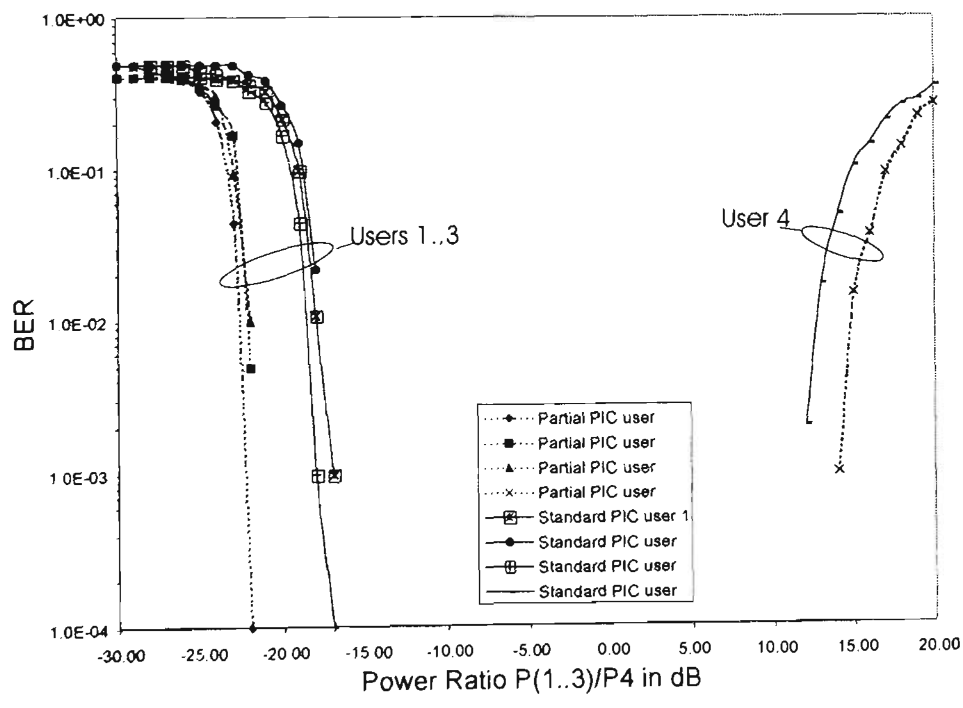


Figure 23 An asynchronous DS-CDMA system with 4 users, no time offset errors, and Eb/No = 7dB.

A system where a timing error equal to 0.1Tc is then introduced, as shown in Figure 24. To achieve a BER of 10^{-3}, the power ratio had to be decreased to −19dB for the partial PIC and −16dB for the standard PIC. When the Near Far situation was reversed, to achieve a BER of 10^{-3} a power ratio of approximately +11dB and +8dB for the partial PIC and standard PIC respectively were required.

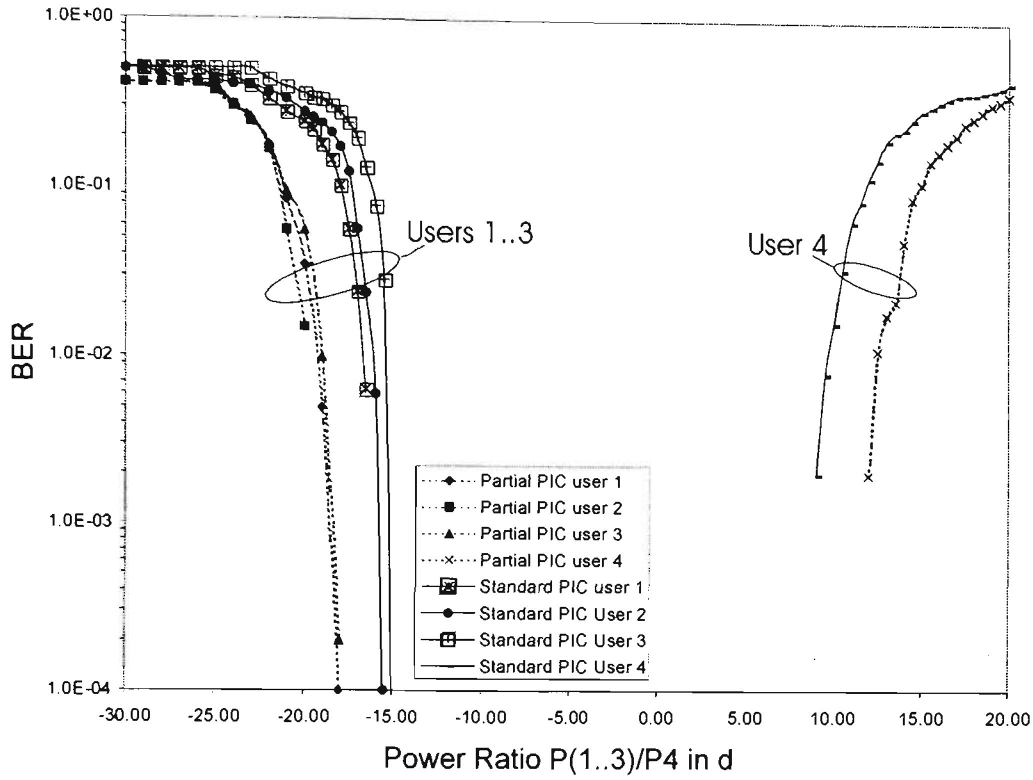


Figure 24 Asynchronous DS-CDMA system with 4 users, time offset error = 0.10Tc, and Eb/No = 7dB

As the time offset errors were increased to 0.2Tc, both the partial PIC and the standard PIC become more sensitive to the NF ratio, this can be observed in Figure 25. To achieve the BER of 10^{-3}, the power ratios had to be decreased to −15dB for the partial PIC. Due to the asynchronous nature of this system, it can be observed that the BER curves for the 3 users do not exactly coincide with each other. To achieve the BER of 10^{-3} when the 3 users were near, the power ratio had to be decreased to approximately +8dB and +5dB for the partial PIC and the standard PIC respectively.

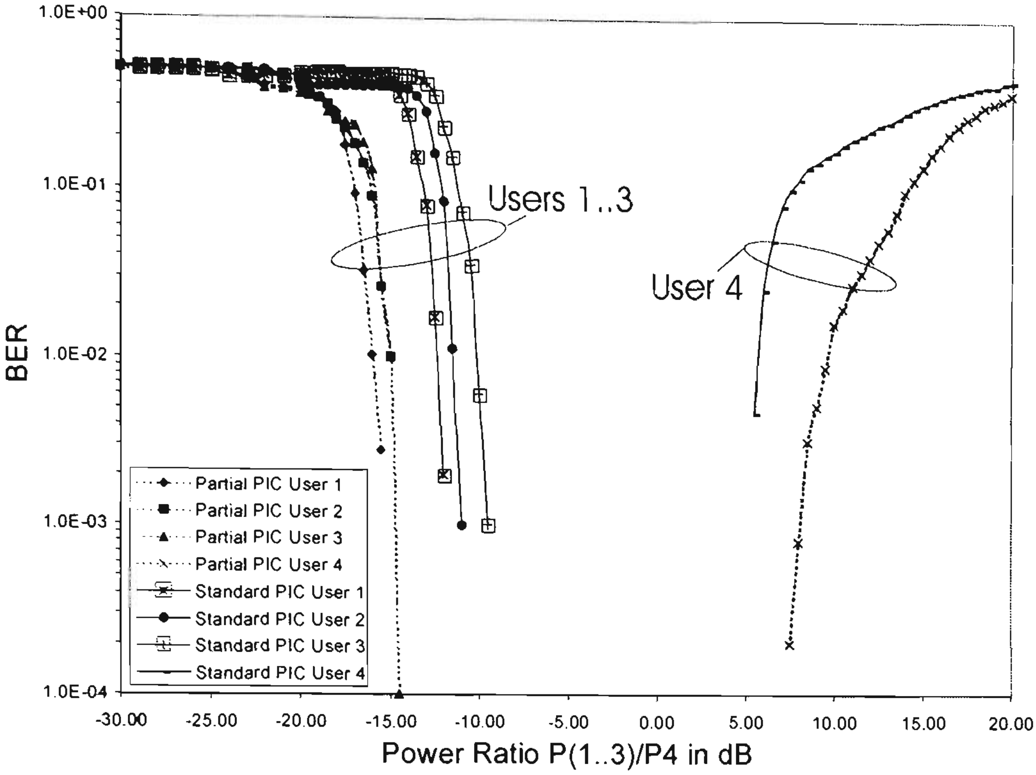


Figure 25 Asynchronous DS-CDMA system with 4 users, time offset error = 0.20Tc, and Eb/No= 7dB

As shown in Figure 26, a system with a time offset error of 0.4Tc is simulated. To achieve BER performance of 10^{-3}, the power ratio decreased to –8dB for the partial PIC and –5dB for the standard PIC. For the near situation, the power ratio was decreased to approximately +2dB and 0.5dB for the PIC and the partial PIC respectively.

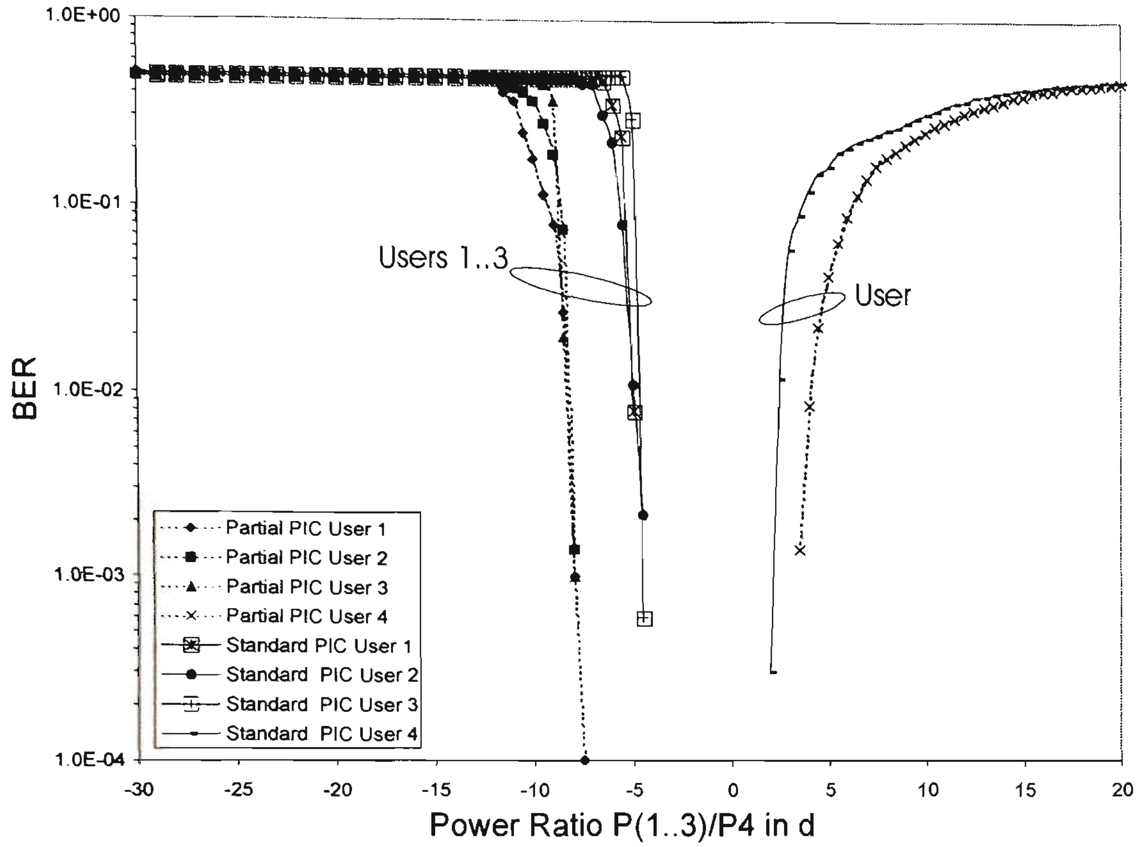


Figure 26 Asynchronous DS-CDMA system with 4 users, time offset = 0.40 of a chip, and Eb/No= 7dB

In all cases the partial PIC outperformed the standard PIC. It was able to operate in a near far environment with signal levels that were at least 3dB greater than those of the standard PIC. This type of performance is particularly attractive when the receiver is operating in a hostile environment that is affected severely from the NFE.

	Wanted Signal : Far			Wanted Signal : Near		
	Power Ratio at 10^{-3}	Amplitude of user 1 (dBm)	Amplitude of users 2 to 4 (dBm)	Power Ratio at 10^{-3}	Amplitude of user 1 (dBm)	Amplitude of users 2 to 4 (dBm)
Partial PIC	-22dB	79	0.5	13dB	0.05	0.5
Standard PIC	-18dB	32	0.5	11dB	0.04	0.5
Conventional Rx	-14dB	12	0.5	7dB	0.1	0.5
Improvement	8dB			6dB		

Table 4 Performance of the 4 user CDMA system for standard and Partial PIC and Matched Filter Receivers

From this table it is quite clear that the partial PIC is able to operate in a Near Far Environment and still provide a satisfactory BER performance. When the wanted signal was far for the partial PIC and a BER of 10^{-3} there was an overall improvement of 8dB compared with that of the conventional receiver and 4dB compared to the standard PIC receiver. For the case when the wanted signal was near and a received BER of 10^{-3} is required, the partial PIC receiver gave an improvement of 2dB with respect to the standard PIC and 7dB compared with the conventional receiver.

6.3 Channel Estimation

Improvements to the partial interference cancellation technique can be made by noting that for a CDMA system operating in a multipath channel, the MAI will be time variant. It will vary from one user to another and from bit to bit. The power level of each user at a particular time instant, in the channel, will also change with time and according to the delays in the channel. Inaccurate complex channel coefficient estimates have a dramatic impact on the quality of the MAI estimates. A crucial part in channel estimation is the filtering of the rough channel coefficient estimates. Since the signal to noise ratio and the rate of fading are time-variant and different for each user, the channel estimation filters should be adaptive and can be improved by using adaptive channel estimation filters.

As with any adaptive filter or equaliser there will be a trade off between convergence time and the size of the spreading code. These adaptive channel estimation algorithms are obviously more robust than existing methods as they are able to operate successfully in a multipath fading environment.

6.4 Serial Interference Cancellation

The successive interference cancellation detector takes a serial approach to reducing the interference that is caused by MAI. As each stage cancels out one additional user from the received signal, the contribution due to MAI on the remaining users decreases. With this detector, it is important to cancel out the strongest signal first before the detection and cancellation of the other users, as it has the most severe effect in producing interference and it is also easier to acquire both synchronisation and perform the despreading operation.

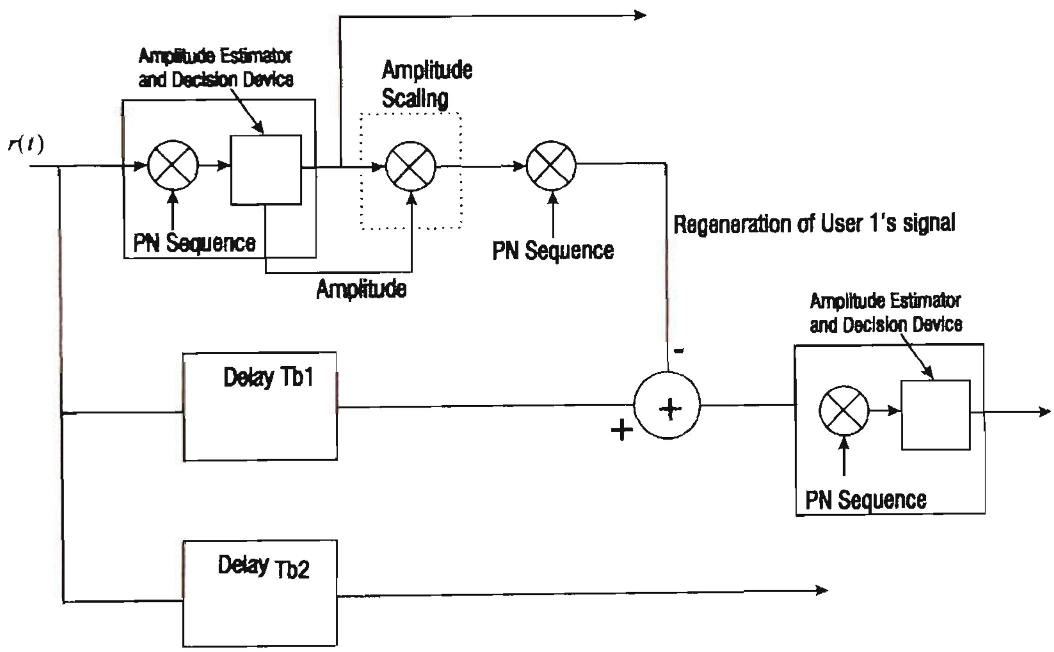


Figure 27. Serial Interference Canceller.

The operation of the SIC consists of: (1) Using the conventional detector to detect the strongest user, this is user 1. (2) Make either a soft or hard decision on user 1 and (3) using the knowledge of its amplitude and chip sequence, regenerate the spread spectrum signal x(t). (4) Cancel this interferer and (5) repeat until all users' signals are detected.

As all of the users are detected independently by the conventional receiver, the signatures timing and phase information of each user must be known. In order to achieve sufficient cancellation of the MAI, the amplitude information of each user must be accurately estimated. By cancelling only a number of the strongest signals, the complexity of the receiver can be simplified.

The complexity of this receiver is linear with the number of users. It requires only a minimal amount of additional hardware and it has the potential to provide significant improvement over the conventional detector.

Despite the advantages, there exist some implementation difficulties, and these include the additional delay that is required for each stage, hence a trade off between the amount of delay that can be tolerated and the number of users that are cancelled must be made. Each time the power profile of the system changes there is a need to reorder the signals and the difficulty of reordering the users' signal powers in real time must be taken into consideration.

Even if the amplitude and timing information is correct but the initial data estimates are inaccurate, there could be a degradation of performance. In this case there must be a minimal level of performance requirement for the conventional detector.

7 Conclusions

In this thesis, multiuser demodulation algorithms for receivers of asynchronous DS-CDMA systems were considered.

Two multiuser algorithms are examined, the Adaptive Centralised Receiver and interference cancellation techniques based on the Parallel Interference Canceller (PIC). Different digital baseband realisations of multiuser receiver structures have been modelled, and their performance has been analysed. Special attention has been given to these receivers in a near far environment.

The partial PIC receiver was demonstrated to achieve better performance than the adaptive receiver in a near far environment. This is easily observed by comparing Figures 23-26. Here an improvement of approximately 3dB is achieved with the Partial PIC over the adaptive algorithm. This results could be further improved through the use of an adaptive Partial PIC where the partial weights of each stage of the PIC are calculated adaptively. Further improvements can also made by employing Forward Error Correction techniques and Channel Estimation.

A major concern is the time it takes for the adaptive receiver to converge when it is subject to an hostile environment where each user arrives at the receiver with non uniform amplitudes. From Figure 16 it is clear that the receiver will converge at different rates depending upon the received signal strengths. In a real time system this would cause unacceptable delays in processing the users' data whilst the algorithm converges for each user. As the number of users increase, the time it takes for the adaptive receiver to converge also increases. For the situation where there were 31 users, 400 symbol samples where required for the receiver to converge. Depending on the sample time of each symbol, this could take up to half a second. For a high performance system, algorithms like the Recursive Least Squares and the Kalman filter could be used.

This type of receiver is suitable for CDMA systems with short spreading codes or in a stationary network in which there is a fixed number of users. During the setup each user is trained only once, not only to the desired spreading code, but also to the channel parameters.

Time offset errors for both the parallel interference canceller and the partial parallel interference canceller were studied. It was observed that the partial PIC was not as sensitive to these errors as the standard PIC and can provide significant performance improvement to a CDMA system. In an NF environment the partial PIC gave an average of 3 to 5 dB improvement over the standard PIC and as such can be employed to relax the power control requirements in CDMA system design.

There multiuser receivers are near far resistant and have the ability to provide substantial improvements to the system capacity as well as relaxing the specifications of the power control.

There are several interesting open problems in multiuser receivers requiring further study. Some of them are discussed here in short.

The performance of the partial parallel interference cancellation receivers can still possibly be improved in some cases. However, a simple and robust way to measure the reliability and to determine the cancellation weights remains to be found. Since the reliability depends on the state of the communication channel, the weights should be adapted to the changes in complex channel coefficients. That poses strict requirements to the speed of such weight determination. Thus, simple adaptive weighting, as proposed in [71], may not be fast enough in fading channels.

From the data detection point of view channel encoding should be taken into consideration. The encoded transmission and reception for CDMA systems utilizing multiuser receivers are important research problems. The overall signal design (design of modulation and coding) for multiuser channels with some efficient low complexity joint decoding algorithms for all users would be of major interest.

The impact of several system level aspects to the multiuser receiver performance would be worth investigating. The impact of multiuser receivers on the overall system capacity has not been analysed thoroughly yet. For example, the effect of the existence of multiple cells is often neglected in the multiuser receiver analysis. Multiuser receivers could naturally handle the intra-cell MAI by exploiting some ordinary multiuser receiver, e.g., a PIC receiver. The inter-cell MAI, on the other hand, could be compressed by some decentralized receiver technique. Multiuser receiver design and receiver performance in CDMA systems with multiple data rates in realistic fading channels have been studied very little. The application of group wise multiuser receivers, where grouping could be based on the data rates of the users, appears as an interesting alternative [72]. The performance of multiuser receivers with antenna arrays should also be taken into consideration in the studies.

A severe problem is the fact that there are convergence problems associated with most adaptive receivers due to the large number of taps required by direct form FIR filters. Therefore, there is room for further work on dimension reduction techniques to reduce the number of filter taps needed, as well as for work on efficient adaptive algorithms to enhance the convergence. More work on the performance of different adaptive algorithms is required. In general, the impact of various practical non idealities (e.g., delay estimation errors and quantization in DSP hardware) on the performance of the receivers should be considered.

8 References

[1] S. Glisic, B. Vucetic, "*Spread Spectrum CDMA systems for Wireless Communications*", Artech House London 1997

[2] R. Dixon, "*Spread Spectrum Systems with Commercial Applications*", Wiley Interscience, New York 1994.

[3] S Verdu, "Minimum probability of error for asynchronous Gaussian multiple-access channels.", *IEEE Transactions on Information Theory*, vol 32, no. 1, pp.85–96 1986.

[4] F-C Zheng and S. Barton "On the performance of Near Far Resistant CDMA Detectors in the Presence of Synchronisation Errors", *IEEE Transactions on Communications*, vol. 43, no. 12, pp. 3037-3045, Dec. 1995

[5] R. Lupas and S. Verdu, " Near-Far Resistance of Multi-User Detectors in Asynchronous Channels", *IEEE Transactions on Communications*, vol 38, no 4, pp. 496-508, April 1990.

[6] R. Lupas and S. Verdu, "Linear Multi-user Detectors for Synchronous Code Division Multiple Access Channels", *IEEE Transactions on Information Theory*, vol 35 no 1, pp 123-136, Jan. 1989.

[7] Z. Zvonar and D. Brady, "Suboptimum multiuser detector for synchronous CDMA frequency selective Rayleigh fading channels", *Proceedings of the IEEE Globecomm communications Theory Mini Conference*, pp. 82-86, 1992.

[8] M. Varanasi and B. Aazhang, "Multistage detection in asynchronous CDMA communication", *IEEE Transactions on Communications*, vol 38, no. 4, pp. 509 - 519, April 1990

[9] A. Duel-Hallen, "Decorrelating decision feedback multiuser detector for synchronous code division multiple access channel", *IEEE Transactions on Communications*, vol 41, no. 2, pp. 285-290, Feb. 1993.

[10] A. Kajiwara and M. Nakagawa, "Microcellular CDMA system with a linear multiuser interference canceller", *IEEE Journal on Selected areas in Communications*, vol. 12, no. 4, pp. 605-611, April 1994.

[11] F. Van Heeswyk, D. Falconer and A. Sheikh, "Decorrelating detectors for quasi-synchronous CDMA", *Wireless Personal Communications*, Kluwer Academic Publishers, vol 3, no. 1-2, pp. 129-147, 1993.

[12] R. Iltis and L. Mailaender, "Multiuser detection of quasisynchronous CDMA signals using linear decorrelators", *IEEE Transactions on Communications*, vol 44, no. 11, pp. 1561-571, Nov. 1996.

[13] R. Iltis, "Demodulation and code acquisition using decorrelator detectors for QS-CDMA", *IEEE Transactions on Communications,* vol 44, no. 11, pp. 1553 - 1560, Nov. 1996.

[14] U. Mitra, H.V. Poor, "Adaptive Decorrelating Detectors for CDMA Systems", *Wireless Personal Communications Journal*, Kluwer Academic Publishers, Vol. 2, No. 4, pp 415 - 440, 1992.

[15] U. Mitra, H.V. Poor, "Analysis of an adaptive decorrelating detectors for synchronous CDMA channels", *IEEE Transactions on Communications,* vol 44, no. 2, pp. 257-268, Feb. 1996.

[16] U. Madhow and M.L. Honig, "MMSE interference suppresion for direct sequence spread spectrum CDMA", *IEEE Transactions on Communications*, vol 42, no. 12, pp. 3178 - 3188, Dec. 1990.

[17] Z. Xie, R. Short and C. Rushforth, "A family of suboptimum detectors for coherent multiuser communications", *IEEE Journal on Selected Areas in Communications* , vol 8, no. 4, pp. 683 - 690, Aug. 1996.

[18] U. Madhow, M.L Honig, "MMSE Interference Suppressioin for Direct Sequence Spread Spectrum CDMA", IEEE Transactions on Communications, vol. 42, no. 12, Dec. 1994, pp . 3178-88

[19] A. Klein, G. Kaleh and P. Baier , "Zero forcing and minimum mean-square-error equalisation for multiuser detection in code-division multiple access channels", *IEEE Transactions on Vehicular Technology* vol. 45, no. 2, pp. 276–287, Feb. 1996.

[20] W. Wu and K. Chen, "Linear multiuser detectors for synchronous CDMA communication over Rayleigh fading channels", *Proceedings IEEE International Symposium on Personal, Indoor, and Mobile Radio Communications (PIMRC) 1996*, Taipei, Taiwan, pp 578–582.

[21] M. Honig and W. Veerakachen, "Performance variability of linear multiuser detection for DS/CDMA", *Proceedings IEEE Vehicular Technology Conference (VTC)*, vol. 1, pp. 372-376. Atlanta, Georgia, USA 1996.

[22] S. Verdu and H.V. Poor, "Probability of error in MMSE multiuser detection", *IEEE Transactions on Information Theory*, vol 43, no. 3, pp. 858–871, March 1997.

[23] X. Wang and H.V. Poor, "Multiuser diversity receivers for frequency-selective Rayleigh fading CDMA channels", *Proceedings IEEE Vehicular Technology Conference (VTC)*, Phoenix, Arizona, USA, vol 1, pp. 198–202, 1997.

[24] M. Honig, U. Madhow and S. Verdu, "Blind adaptive Multiuser Detection", *IEEE Transactions on Information Theory*, vol 41, no. 3, pp. 944-960, March 1995.

[25] X. Wang and H.V. Poor, "Multiuser diversity receivers for frequency selective Rayliegh Fading CDMA Channels", *Proceedings IEEE Vehicular Technolgy Conference (VTC)*, Phoenix Arizona USA, vol 1 pp. 198-202, 1997.

[26] E.L Kuan and L. Hanzo, "Joint Detection CDMA Techniques for Third Generation Transceivers", *Proceedings ACTS 98*, Rhodes, Greece 1998.

[27] S.L. Miller, "Training Analysis of adaptive interference suppression for DS-CDMA multiple access systems", *IEEE Transactions on Communications*, vol 44, no. 4, pp 488-495, April 1998.

[28] K.B. Lee, "Orthogonalisation based adaptive algorithm for DS-CDMA multiple access systems", *IEEE Transactions on Communications*, vol 44, no. 9, pp. 1082–1085 September 1998.

[29] G. Woodward, P. Rapajic and B.S. Vucetic, "Adaptive Algorithms for aysnchronous DS-CDMA receivers", *Proc IEEE International Symposium on Personal Indoor and Mobile Radio Communications 1996*, pp. 583-587, Taipei, Taiwan.

[30] M. Latva-aho and M. Juntti, "Modified adaptive LMMSE receiver for DS-CDMA systems in fading channels", *Proc IEEE International Symposium on Personal Indoor and Mobile Radio Communications PIMRC, 1997*, Helsinki Finland vol 2, pp. 554-558.

[31] P.B. Rapajic and B.S. Vucetic, "Adaptive receiver structures for asynchronous CDMA systems", *IEEE Journal on Selected Areas in Communications*, vol 12, no. 4, pp. 685–697, May 1994.

[32] P. Rapajic PB and B.S. Vucetic, "Linear adaptive transmitter-receiver structures for asynchronous CDMA systems", *European Transactions on Telecommunications*, vol. 1, no. 6, pp. 21–27, 1995.

[33] S.L. Miller, "An adaptive direct-sequence code-division multiple-access receiver for multiuser interference rejection", *IEEE Transactions on Communications*, vol. 43, no. 2, pp. 1746-1755, 1995.

[34] L.B. Milstein and J. Wang, "Interference suppression for CDMA overlays of narrowband waveforms", *IEEE Third International Symposium on Spread Spectrum Techniques and Applications*, pp. 61-88, July 4-6, 1994.

[35] L.B. Milstein, "On the Feasibility Study of a CDMA Overlay for Personal Communications," *IEEE Journal on Selected Areas in Communications*, vol 10, no 5, pp. 655-668, May 1992.

[36] M.E. Davis and L.B. Milstein L. B., "Anti Jamming Properties of a DS-CDMA equalisation filter", *Proceedings of the 12th Annual IEEE Military Communications Conference*, Boston, USA, vol 3, pp. 1008-1012, 1993.

[37] A. Mammela, "The performance of adaptive interference suppression filters used in PN sequence spread spectrum systems." *EUROCON 1988 8th European Conference on Electrotechnics. Conference Proceedings on Area of Communications*, pp. 126-129, 1988.

[38] J. Lee and C. Lee, "Adaptive Filters for suppressing irregular hostile jamming in Direct Sequence Spread Spectrum Systems" *MILCOM 87, IEEE Military Communications Conference*, vol. 1, pp. 118-122, October 1987.

[39] Y. He, S. Lei, P. Das, and G. Saulnier, "Suppresion of narrowband jammers in a DS spread spectrum receiver using modified adaptive filtering techniques", *Globecom 88 IEEE Global Communications Conference and Exhibition,* vol. 1, pp. 550-545, December 1988

[40] U. Madhow and M. Honig, "MMSE Interference Suppression for CDMA", *IEEE Transactions on Communications*, vol. 42, no. 12, pp. 3178-3188, December 1994

[41] R. Buehrer, N.S. Correal and Woerner BD, "A comparison of multiuser receivers for cellular CDMA", *Proc. IEEE Global Telecommunication Conference (GLOBECOM)*, London, U.K. vol. 3, pp. 1571–1577, 1996.

[42] P. Patel and J.Holtzman, "Analysis of a Simple Successive Interference Cancellation Scheme in a DS/CDMA System", *IEEE Journal on Selected Areas in Communications*, vol. 12, no. 5, June 1994.

[43] L.B. Nelson and H.V. Poor, "Iterative multiuser receivers for CDMA channels: An EM-based approach", *IEEE Transactions on Communications*, vol 44, no 12, pp. 1700-1710, December 1996.

[44] A. Soong and W. Krzymien, "Performance of a reference symbol assisted multistage successive interference cancelling receiver in a multi-cell CDMA wireless systems", *Proc. IEEE Global Telecommunication Conference (GLOBECOM)*, Singapore, vol. 1, pp. 152–156, 1995.

[45] A. Soong and W. Krzymien, "A novel CDMA multiuser interference cancellation receiver with reference symbol aided estimation of channel parameters", *IEEE Journal on Selected Areas in Communications*, vol. 14, no. 8, pp. 1536-1547, August 1996.

[46] O. Nesper and P. Ho, "A reference symbol assisted interference cancelling hybrid receiver for an asynchronous DS/CDMA system", *Proc. IEEE International Symposium on Personal, Indoor, and Mobile Radio Communications (PIMRC)*, Taipei, Taiwan, vol. 1, pp. 108–112, 1996.

[47] A.L. Johansson and A. Svensson, "Multistage interference cancellation in multirate DS/CDMA on a mobile radio channel", *Proc. IEEE Vehicular Technology Conference (VTC)*, Atlanta, Georgia, USA, vol. 2, pp. 666–670, 1996.

[48] A. Soong and W. Krzymien, "Robustness of the reference symbol assisted multistage successive interference cancelling receiver with imperfect parameter estimates", *Proc. IEEE Vehicular Technology Conference (VTC)*, Atlanta, Georgia, USA, vol. 2, pp. 676–680, 1996.

[49] A. Johansson, A. Svensson, "On Multirate DS-CDMA Schemes with Interference Cancellation", *Wireless Personal Communications*, Kluwer Academic Publishers vol. 9, no. 1, pp. 1-29, Jan. 1999.

[50] T.B. Oon, R. Steele and Y. Li, "Performance of an adaptive successive serial-parallel CDMA cancellation scheme in flat Rayleigh fading channels", *Proc. IEEE Vehicular Technology Conference (VTC)*, Phoenix, Arizona, USA, vol. 1, pp. 193–197, 1997.

[51] M. Latva-aho and J. Lilleberg, "Parallel interference cancellation in multiuser CDMA channel estimation", *Wireless Personal Communications*, Kluwer Academic Publishers 1998.

[52] Y. Sanada Q. Wang, "A co-channel interference cancellation technique using orthogonal convolutional codes", *IEEE Transactions on Communications*, vol. 44, no. 5, pp. 549-556, May 1996.

[53] Y. Sanada and Q. Wang, "A Co-channel interference cancellation technique using orthogonal convolutional codes on multipath Rayleigh fading channel", *IEEE Transactions on Vehicular Technology*, vol 46, no. 1, pp. 114–128, January 1997.

[54] U. Fawer and B. Aazhang, "A multiuser receiver for code division multiple access communications over multipath channels", *IEEE Transactions on Communications*, vol 43, no. 4, April 1995.

[55] H. Holma, A. Toskala and A. Hottinen, "Performance of CDMA multiuser detection with antenna diversity and closed loop power control", *Proc. IEEE Vehicular Technology Conference (VTC)*, Atlanta, Georgia, USA, vol 1, pp. 362–366, 1996.

[56] M. Latva-aho and J. Lilleberg, "Parallel interference cancellation in multiuser detection", *Proc. IEEE International Symposium on Spread Spectrum Techniques and Applications (ISSSTA)*, Mainz, Germany, vol. 3, pp. 1151–1155, 1996.

[57] H. Holma, A. Toskala and A. Hottinen, "Performance of multistage multiuser detection in a fading multipath channel", *Proc. IEEE Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC)*, Toronto Canada, vol. 3, pp. 960-964, 1995.

[58] H. Holma, A. Toskala and A Hottinen, "Multiuser Detection for multirate CDMA communications", *Proc. IEEE International Conference on Communications*, Dallas Texas, USA, 1996.

[59] M. Latva-aho and J. Lilleberg, "Delay trackers for multiuser CDMA receivers", *Proc IEEE International conference on universal personal communications ICUPC*, Boston Massachusetts, USA, vol. 1, pp. 326-330, 1996.

[60] V. Vanghi and B. Vojcic, "Soft interference cancellation in multiuser communications", *Wireless Personal Communications*, Kluwer Academic Publishers, vol 3. pp. 111-128, 1996.

[61] R.M. Buehrer, A. Kaul, S. Striglis and B.D. Woerner, "Analysis of DS-CDMA parallel interference cancellation with phase and timing errors", *IEEE Journal on Selected Areas in Communications*, vol 14, no. 8, pp. 1522-1535, 1996.

[62] B.S. Abrams, A.E. Zeger and T.E. Jones, "Efficiently structured CDMA receiver with near-far immunity", *IEEE Transactions on Vehicular Technology*, vol. 44, no.1, pp. 1–13, January 1995.

[63] H. Seki, T. Toda and Y. Tanaka, "*Low Delay Multistage Parallel Interference Canceller for Asynchronous CoherentDS/CDMA Systesm and its Performance with Closed Loop TPC*", *Proc. APCC,* pp. 832-837, 1997.

[64] U. Jetzek, J. Erbs, U. Sorger. "A new flexible multiuser detection scheme for DS-CDMA Systems", *Proc. IEEE Globecomm 98*, Sydney Australia, pp. 3144-3150, 1998.

[65] Futami T, Seki H, and Tanaka Y, "Performance Evaluation of DS-CDMA Multistage interference canceller in MultiCell Envirnoments", *Proc. IEEE Globecomm 98*, Sydney Australia. pp. 896-901, 1998.

[66] S. Suzuki and K. Kobayshi, "Interference Cancellation Methods for DS-CDMA Multicode Packet Networks", *Proc. IEEE Globecomm 98*,, Sydney Australia, pp. 3162-3167, 1998,

[67] D. Divsalar, M. Simon M and D. Raphaeli, "Improved Parallel Interference Cancellation for CDMA", *IEEE Transactions on Communications*, vol 46, no. 2, pp. 258-268, Feb. 1998.

[68] P. Shan and T. Rappaport, "Parrallel Interference Cancellation (PIC) Imporovement for CDMA multiuser receivers using Partial Cancellation of MAI Estimates", *Proc. IEEE Globecomm 98*, Sydney Australia, pp. 2615-2620, 1998.

[69] Haykin S. "*Adaptive Filter Theory*", Prentice Hall New Jersey, 1996

[70] Y. Li and R. Steel R., "Serial interference cancellation method for CDMA", *Electronic Letters*, vol 30, pp. 1581-1583, Sept. 1994

[71] G. Xue, J. Weng, L. Ngoc and S. Tahar, "Adaptive Multistage Parallel Interference Cancellation for CDMA over Multipath Fading Channels", *Proc IEEE VTC Conference 99*, Houston, Texas, USA.

[72] Juntti MJ, "Performance of multiuser detection in multirate CDMA systems". *Wireless Personal Communications,* Kluwer Academic Publishers, Vol 11. No. 3, pp. 293-311, Dec. 1997.

Appendix A: Published Paper

K. Anderson, F.C. Zheng, M. Faulkner, "A Study on the performance of the Partial PIC CDMA detector in the presence of time offset errors" *IEEE International Symposium on Signal Processing and its Applications, ISSPA 99*, pp.709-712, Brisbane Australia.

A Study on the performance of the Partial PIC CDMA detector in the presence of time offset errors

Kevin Anderson, Fu-Chun Zheng, Michael Faulkner
Victoria University of Technology
P.O. Box 144228 MCMC
Melbourne, VIC, 8001

Abstract - *This paper investigates the effect of time offset errors on the partial (PIC) and compares the performance of it against that of the standard (PIC). The BER performances of the standard and partial interference cancellers are simulated in a near far environment with varying time offset errors. These simulations indicate that whilst timing errors significantly affect the performance of both these schemes, they do not diminish the gains that are realised by the partial PIC over that of the standard PIC.*

Keywords :- Multiuser Detection, Parallel Interference Canceller, Time offset errors

I. Introduction

The third generation of the cellular mobile communications network will be based on wideband code division multiple access (W-CDMA). The capacity of a DS-CDMA system is limited by the multiple access interference (MAI) caused by users sharing the same channel (especially in the asynchronous reverse link). To realise the high capacity requirement of this next generation, techniques such as multiuser detection that reduce MAI are required.

Multiuser detectors have been the focus for the last decade as a means of reducing MAI. They track and demodulate all user waveforms simultaneously and the receiver then makes a symbol decision based on the observation of the whole received waveforms for all users [1].

Recently, studies have focused on interference cancellation techniques, which have lower complexity than the decorrelating detectors and can be applied to systems with a long spreading code. Interference cancellers estimate the MAI and this estimate is then subtracted from the received signal and passed onto the next stage where the bit decision of the wanted signal is improved [2].

Interference cancellation can be derived as an approximation of the maximum likelihood sequence detector (MLSD) receiver with the assumption that the data, amplitude and delays of the interfering users (or a subset) are known. There are several strategies for estimating the interference, leading to different IC techniques. The interference can be canceled either simultaneously from all users, termed *parallel* interference cancellation (PIC) [2], or on a user by user basis, called *serial* (or successive) interference cancellation (SIC) [2].

The successive interference cancellation (SIC) detector takes a serial approach to canceling interference. Users are first estimated and then ranked according to their received powers. They are then cancelled out in order from the strongest user to the weakest user. For each stage of cancellation, one additional bit delay is required, hence there must be a trade off between the number of users that are cancelled and the amount of delay that can be tolerated [3]. There is also a need to reorder the signals whenever the power profile changes.

In contrast to the SIC, the PIC does not need to reorder each user's signal according to its power and doesn't suffer from the large delays that are associated with the SIC. For users with equal power, the SIC scheme performs significantly worse than the PIC, but at the same time the PIC requires more hardware. However, as the user powers gets more diverse, the relative performance of the successive scheme improves.

Simon and Divaslar [4] proposed improvements to the PIC by using partial cancellation at each stage of the Interference Canceller. Shan and Rappaport further investigated this technique of partial cancellation using a synchronous system with no time offset chip errors [5]. The effect of the time offset errors on a synchronous CDMA system using a standard PIC was studied by Buehrer et al in [6]. As there are always acquisition and tracking errors in a practical CDMA system, an interesting question to ask is: how will the partial PIC perform in an asynchronous system in the presence of time offset errors? This paper is to present an answer to such a question.

II. Partial Parallel Interference Cancellation

In the early stages of interference cancellation, where the interference estimate is poor, the tentative data decisions at the earlier stages are less reliable than those of the following stages. It is therefore preferable not to cancel out the entire amount of the estimated MAI, but only a fraction of it. As the interference canceller (IC) operation progresses, the estimates of the MAI improve and the amount of the "real" interference that is being removed also increases. This is the basic principle of the partial PIC [4][5].

The partial PIC structure that is used in this paper is based on [5]. The scheme multiplies the original received signal by $1/p$ before subtracting the MAI estimate where, $0 < p < 1$ is the scaling factor for MAI estimate. Clearly the partial PIC will reduce to a standard PIC if $p = 1$. A block diagram of the partial PIC is shown in Figure 1.

The performance of the partial PIC in an error free synchronous system was investigated by [4] and [5]. However a perfect chip timing and phase synchronisation is rarely possible in a practical spread spectrum system, due to limited resolution at the sliding correlator and jitter in the received signal. The task of this paper is to examine the performance of the above partial PIC detector in an asynchronous system in the presence of time offset errors. Mathematically this can be described as follows.

The received signal is given by :

$$r(t) = \sum_{m=0}^{M-1} \sum_{k=1}^{K} A_k b_k^{(m)} s_k(t - mT_b - \tau_k) + n(t). \qquad (1)$$

Where A_k is the amplitude of the signal at the output of user k's transmitter, m the users' symbols index, S_k is the user k's spreading code, T_b is the period of bit, τ_k is the

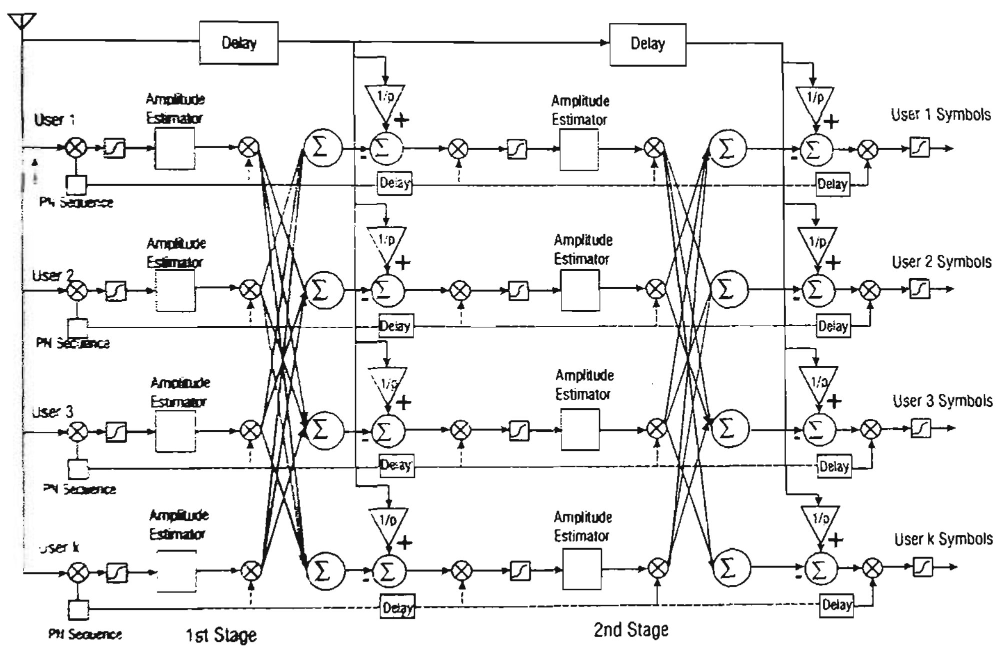


Figure 1.Partial Parallel Interference Canceller

As shown in Figure 1, the partial PIC includes a standard CDMA detector, which is implemented as a bank of matched filters. At this stage the initial data and amplitude estimates are obtained. This information, along with the known users' spreading sequences, is used to regenerate the DS-SS signal associated with each particular user. The MAI for each user is the sum of all the other users. The original received signal is multiplied by the scaling factor and the estimated MAI is then subtracted from this scaled version of the received signal to form the partial estimates of this stage.

random time delay associated with the kth user and n(t) is Additve White Guassian Noise (AWGN).

The locally generated spreading code at the receiver can be expressed as:

$$s_{rep_k} = s_k(t - nT_b - \tau_k + \varepsilon_k), \qquad (2)$$

where $0 \le |\varepsilon_k| \le T_c$ represents the timing error associated with user k and Tc is the period of each chip.

At the output of the matched filters we have:

$$z_k(n) = s_{rep_k}.r(t) \tag{3}$$

$$= \sum_{m=0}^{M-1}\sum_{l=1}^{K}\left\{A_l b_l^{(m)} s_l(t - mT_b - \tau_l) + n(t)\right\}.s_{rep_k}(t - nT_b - \tau_k + \varepsilon_k)$$

$$= \sum_{m=0}^{M-1}\sum_{l=1}^{K} A_l b_l^{(m)} s_l(t - mT_b - \tau_l) s_{rep_k}(t - nT_b - \tau_k + \varepsilon_k) + v(t).$$

Output $z_k(.)$ is then respread to form a replica of User k's transmitted signal:

$$w_k(n) = z_k(n).s_{rep_k}(t - nT_b - \tau_k + \varepsilon_k). \tag{4}$$

The output of the jth stage of the partial PIC is:

$$PIC_j = \rho_j.r(t) - \sum_{\substack{l=1\\ l\neq k}}^{K} w_l(n)\ , \tag{5}$$

where ρ_j is the fraction of cancellation of the partial PIC. The desired users' data are obtained by multiplying this new estimate by the locally generated spreading sequence:

$$d_k(n) = PIC_j.s_{rep_k}(t - nT_b - \tau_k + \varepsilon_k)\quad . \tag{6}$$

The above steps are carried out in each stage and normally several stages need to be accomplished before an acceptable result can be achieved (2 stages are run in the simulations in this paper, as shown next).

As the value of ε_k is seldom zero in practice, the mismatch described in equations (3) and (6) always exists, which causes the performance of the partial PIC to deteriorate. This will be illustrated by using simulation in the following section.

III. Simulations

In this section, an asynchronous DS-CDMA system with BPSK modulation is assumed, a 31chip gold code generator is used to generate the spreading sequence and 10 samples are taken for each chip. The noise is additive white Gaussian noise (AWGN) and it was assumed that there was no fading in the channel. The scaling factors that were used in the partial PIC were 0.65 for the first stage and 0.85 in the second stage based on the results in [5]. Four users are assumed, 3 of which have the same SNR. The 4th user represents a near far (NF) environment: its power swept over a large range of values. Synchronisation Errors were introduced into the system by offsetting the locally generated PN sequence at the receiver by fractions of a chip as described in section II.

In the first example, shown in Figure 2, a system with no tracking errors is simulated. A near far ratio, Pi/P4, of -21dB is required for a BER of 10^{-3} for the partial PIC and approximately -18dB for the standard PIC. When the situation was reversed, a power ratio of +14dB was required by the partial PIC for a BER of 10^{-3}. The standard PIC required approximately +11dB to achieve a BER of 10^{-3}.

Figure 3 shows a system with a timing error equal to 0.1Tc. To achieve a BER of 10^{-3}, the power ratio had to be decreased to –19dB for the partial PIC and –16dB for the standard PIC. When the Near Far situation was reversed, the power ratio of approximately +8dB and +11dB for the partial PIC and standard PIC respectively were required to achieve a BER of 10^{-3}.

The third example in Figure 4 shows that as the time offset errors were increased to 0.2Tc, both the partial PIC and the standard PIC become more sensitive to the NF ratio. To achieve the BER of 10^{-3}, the power ratios had to be decreased to –15dB for the partial PIC. Due to the asynchronous nature of this system, it can be observed that the BER curves for the 3 users' don't exactly coincide with each other. To achieve the BER of 10^{-3} when the 3 users were near, the power ratio had to be decreased to approximately +8dB and +5dB for the partial PIC and the standard PIC respectively.

In the final example, shown in Figure 5, a system with a time offset error of 0.4Tc is simulated. To achieve BER performance of 10^{-3}, the power ratio decreased to –8dB for the partial PIC and –5dB for the standard PIC. For the near situation, the power ratio was decreased to approximately +2dB and 3.5dB for the PIC and the partial PIC respectively.

Timing errors (or the effect of chip misalignment) cause degradation in the receivers' BER and a reduction in the anti MAI ability of the interference canceller. Whilst timing errors significantly affect the IC's performance it does not diminish the gains that are realised by the partial PIC.

IV. Conclusions

The impact of time offset errors on a Partial PIC in a asynchronous CDMA system was investigated. It was observed that the partial PIC was not as sensitive to these errors as the standard PIC and can provide significant performance improvement to a CDMA system. In an NF environment the partial PIC gave an average of 3 to 5 dB improvement over the standard PIC and as such can be employed to relax the power control to some degree requirements in CDMA system design.

References

[1] R. M. Buehrer, N.S. Correal and B. D. Woerner, "A comparison of multiuser receivers for cellular

CDMA", *IEEE Globecomm '96*, vol 3, pp. 1571-1577, London, Nov 1996.

[2] S. Moshavi, "Multi-user detection for DS-CDMA communications", *IEEE Magazine*, Vol 34, No.10, pp 124-136, Oct 1996.

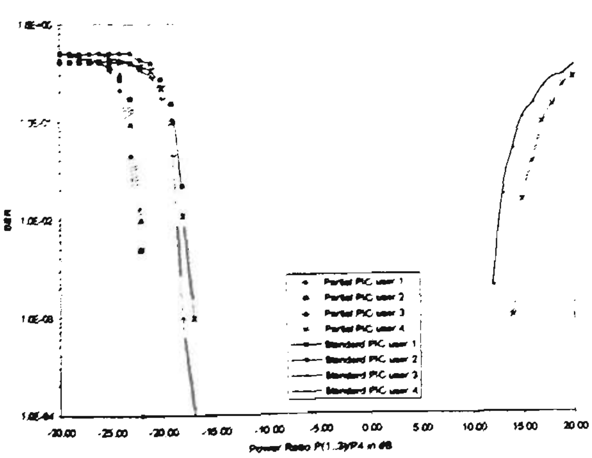
[3] D. Divsalar, M. Simon and D Raphaeli, "A new approach to parallel interference cancellation for CDMA", *IEEE Globecomm '96*, vol 3, pp. 1452-1457, London, Nov 1996.

[4] P. Patel, J.Holtzman, "Analysis of a Simple Successive Interference Cancellation Scheme in a DS/CDMA System". *IEEE Journal on Selected Areas in Communications*, Vol 12 No.5, June 1994.

[5] P. Shan, T.S Rappaport, 'Parallel Interference Cancellation (PIC) improvements for CDMA Multiuser Receiver's using partial cancellation of MAI estimates". *IEEE Globecomm '98*, Sydney, Nov 1998.

[6] R.M. Bruehrer, A. Kaul, S. Striglis, B.D. Woerner, "Analysis of DS-CDMA Parallel Interference Cancellation with Phase and Timing Errors", *IEEE Journal on Selected Areas in Communications*, Vol. 14 pp1522-1535, Oct 1996.

Figure 2: An asynchronous DS-CDMA system with 4 users, no time offset errors, and Eb/No = 7dB



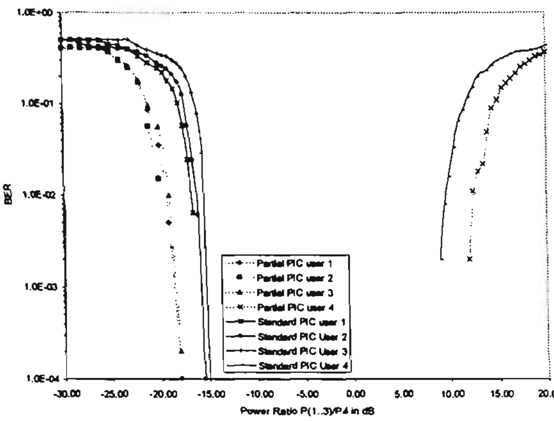


Figure 3: An Asynchronous DS-CDMA system with 4 users, time offset error = 0.10 Tc, and Eb/No = 7dB

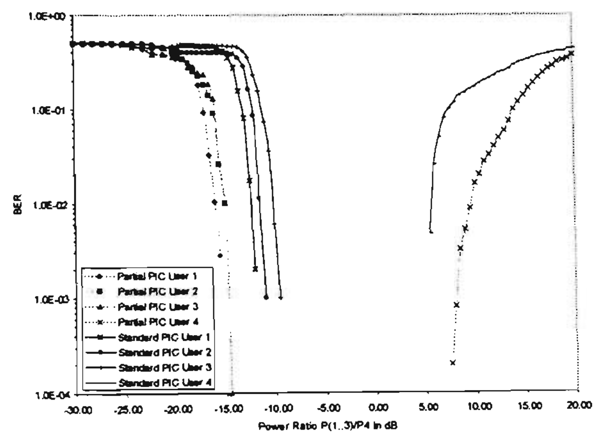


Figure 4: An asynchronous DS-CDMA system with 4 users, time offset error = 0.20Tc, and Eb/No= 7dB

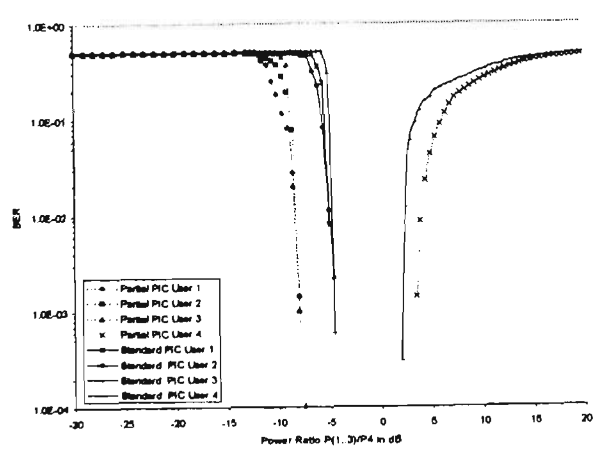


Figure 5: An asynchronous DS-CDMA system with 4 users, time offset = 0.40 of a chip, and Eb/No 7dB