**"Error estimation & analysis of the performance of CDMA due to various parametric impacts using Matlab."**

A DISSERTATION

SUBMITTED IN PARTIAL FULFILLMENT OF THE REQUIREMENT FOR THE

DEGREE OF

MASTER OF ENGINEERING

IN

ELECTRONICS & COMMUNICATION ENGINEERING

BY

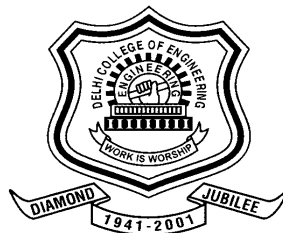SANDEEP SHARMA

UNDER THE GUIDANCE OF

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CERTIFICATE

This is to certify that the thesis entitled **"Error estimation & analysis of the performance of  CDMA due to various parametric impacts using MATLAB."** being submitted by **Sandeep Sharma** in the partial fulfillment of the requirement for the degree of Master of Engineering in Electronics & Communication in the Department of Electronics & Communication, Delhi College of Engineering, University of Delhi is a record of bonafide work done by him under my supervision and guidance.To the best of my knowledge & belief, this dissertation has not been submitted elsewhere for any other degree.

**Rajesh  Rohilla**                                                          **Prof. A. Bhattacharyya**

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The satisfaction and euphoria that accompanies the successful completion of any task would be incomplete without a mention of the people who made it possible, whose constant guidance and encouragement crown all the efforts with success.

I am grateful to Dr. A.BHATTACHARYYA (HOD, ECE Deptt.) for providing me an opportunity to undertake this project. I am highly indebted to Mr. Rajesh Rohilla (ECE Deptt.), my project guide for taking keen interest in my work and for his constant monitoring and invaluable guidance and support throughout the course of my project. I profusely thank him for having patience to clear my doubts and channelise my efforts. His cheerful disposition made my work all the more enjoyable.

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*ABSTRACT*

This dissertation presents the performance evaluation of a multi-user CDMA system in terms of Probability of error (Pe) v/s SNR for wide variety of vital system parameters & components. The study has been motivated by the fact that CDMA greatly simplifies the complexity of a system & enables efficient and reliable multiple access. It also contravenes the need of elaborate network synchronization in traditional approaches like TDM, FDM.


As an alternative to traditional approach a simulation has been envisaged in Matlab which is a powerful technical computing tool. The variable parameters are:

The system has been tested for an optimal combination of the following components:

- Error detecting/correcting codes-Cyclic codes, convolution codes,
- Spreading codes- Hadamard-Walsh codes.
- QPSK modulation.
- Channel- AWGN
- Receiver – integrate & dump followed by a correlation detector.
- Pulse shaping-rectangular.

Various graphs have been plotted between Pe & SNR as a measure of performance appraisal. Exhaustive discussion of Spread Spectrum technology has been done along with analysis of the future scope of CDMA technology.

## *INDEX*

# 1. Definition of Spread Spectrum (SS)

A transmission technique in which a pseudo-noise code, independent of the information data, is employed as a modulation waveform to "spread" the signal energy over a bandwidth much greater than the signal information bandwidth. At the receiver the signal is "despread" using a synchronized replica of the pseudo-noise code.

# 2. Basic principle of Spread Spectrum Systems: DSSS & FHSS

## *Direct Sequence Spread Spectrum (DSSS)*

A pseudo-noise sequence $pn_t$ generated at the modulator, is used in conjunction with an M-ary PSK modulation to shift the phase of the PSK signal pseudorandomly, at the chipping rate Rc ($=1/T_c$) a rate that is an integer multiple of the symbol rate Rs ($=1/T_s$).
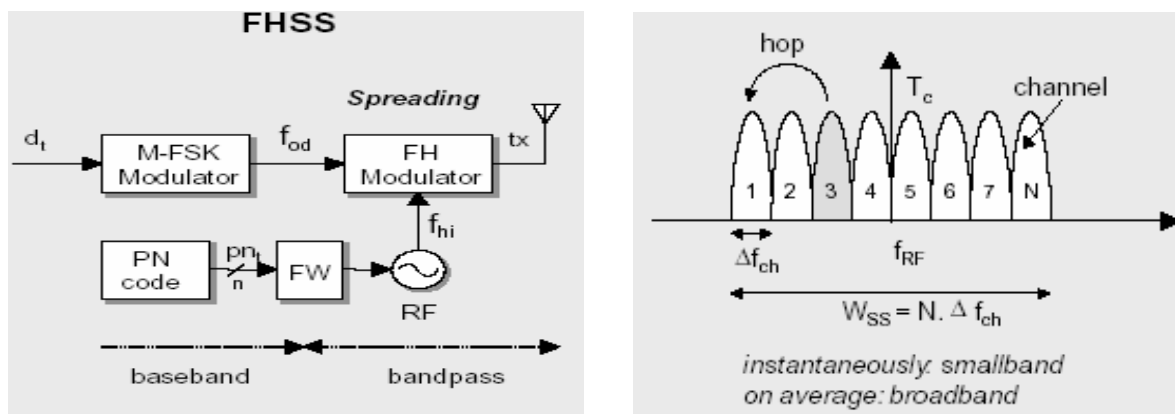
The transmitted bandwidth is determined by the chip rate and by the base band filtering. The implementation limits the maximum chiprate $R_c$ (clock rate) and thus the maximum spreading. The PSK modulation scheme requires a coherent demodulation.
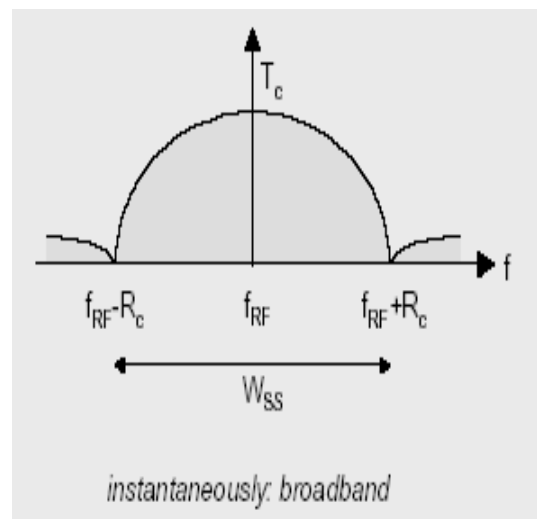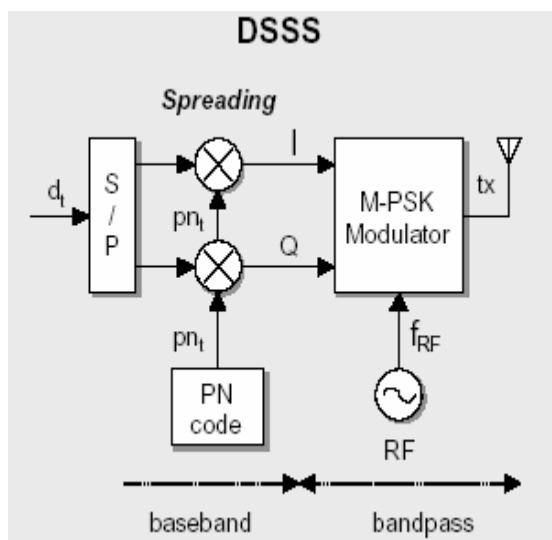
A short-code system uses a PN code length equal a data symbol. A long-code system uses a PN code length that is much longer than a data symbol, so that a-different chip pattern is associated with each symbol.

## Frequency Hopping Spread spectrum

A pseudo-noise sequence $pn_i$ generated at the modulator is used in conjunction with all M-ary FSK modulation to shift the carrier frequency of the FSK signal pseudo randomly, at the hopping rate $R_h$ .The transmitted signal occupies a number of frequencies in time, each for a period of time $T_h$ (=$1/R_h$), referred to as **dwell time**. FHSS divides the available bandwidth into N channels and hops between these channels according to the PN sequence. At each frequency hop time. The PN generator feeds the frequency synthesizer a frequency word FW (a sequence of n chips) which dictates one of $2^n$ frequency positions $f_{hi}$ transmitter and receiver follow the same frequency hop pattern.



The transmitted bandwidth is determined by the lowest and highest hop positions and by the bandwidth per hop position ($f_{ch}$).  For a given hop, the instantaneous occupied bandwidth is identical to bandwidth of the conventional M-FSK, which is typically much smaller than $W_{ss}$. So the FSSS signal is a narrowband signal, all transmission power is concentrated on one channel. Averaged over many hops, the FH/M-FSK spectrum occupies the entire spread

spectrum bandwidth. Because the bandwidth of an FHSS system only depends on the tuning range, it can be hopped over a much wider bandwidth than a DSSS system.Since the hops generally result in phase discontinuity (depending on the particular implementation) a non coherent demodulation is done at the receiver. With slow hopping there are multiple data symbols per hop and with fast hopping there are multiple hops per data symbol.

# 3. Basic principle of Direct Sequence Spread Spectrum (DSSS)

For QPSK modulation the building blocks of a DSSS system are:
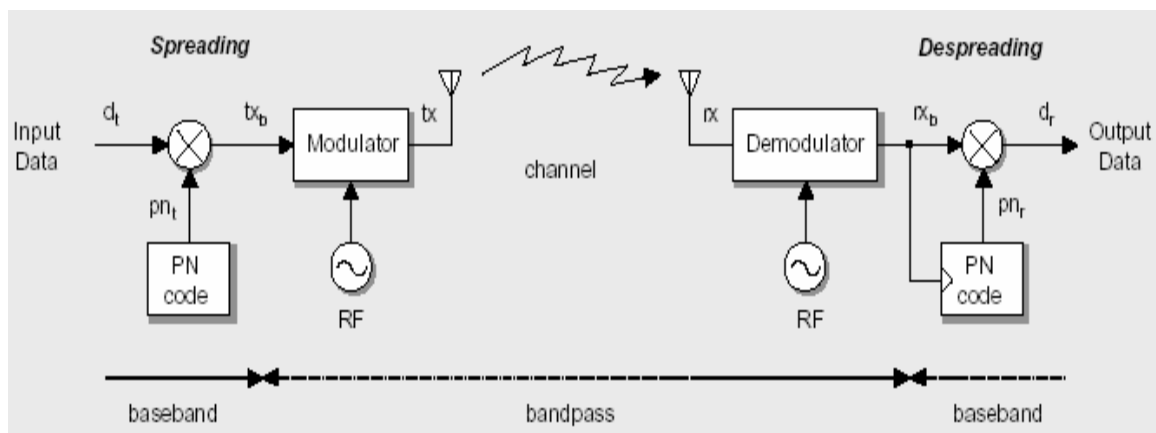


*Spreading:*
In the transmitter, the binary data $d_t$ is 'directly' multiplied with the PN sequence $pn_t$, which is independent of the binary data, to produce the transmitted baseband Signal $tx_b$.

$$Tx_b = d_t.pn_t$$

The effect of multiplication of $d_t$ with a PN sequence is to spread the baseband bandwidth $R_s$ of $d_t$ *to* a baseband bandwidth of $R_c$.
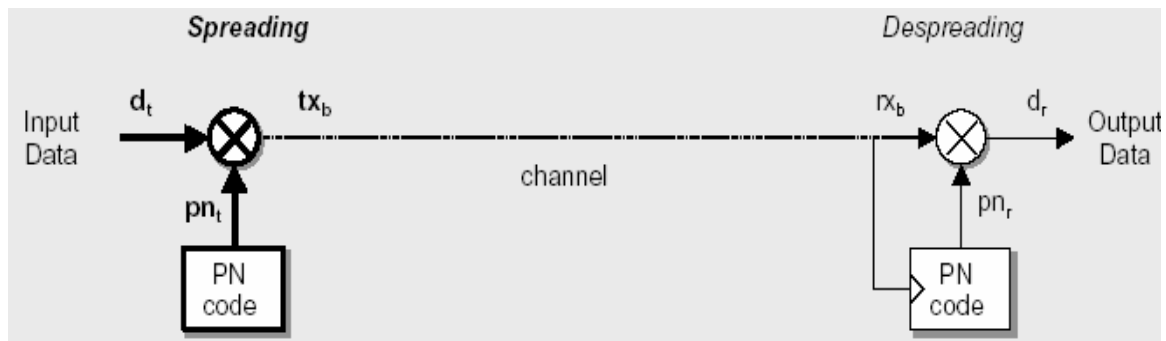
*Despreading:*
The spread spectrum signal cannot be detected by a conventional narrowband

receiver. In the receiver, the received baseband signal rx$_b$, is multiplied with the PN sequence pn$_r$.

- If pn$_r$ = pn$_t$ and synchronized *to* the PN sequence in the received data, than the recovered binary data is produced on d$_r$. The effect of multiplication of the spread spectrum signal rx$_b$ with the PN sequence pn$_t$ used in the transmitter is *to* despread the bandwidth of rx$_b$ to R$_s$.
- If pn$_r$ ≠ pn$_t$ , than there is no despreading action. The signal d$_r$ has a spread spectrum. A receiver not knowing the PN sequence of the transmitter cannot reproduce the transmitted data.

## 3.1 Modulation



Spread spectrum systems are spreading the information signal d$_t$ which has a BW$_{info}$, over a much larger bandwidth BW$_{ss}$:

$$BW_{info} \cong R_s << \quad BW_{ss} \cong R_c$$

The SS-signal spectrum is white noise-like. The amplitude and thus the power in the SS-signal tx$_b$ is the same as in the original information signal d$_t$. Due to the increased bandwidth of the SS-signal the power spectral density must be lower. The band width expansion factor, being the ratio of chip rate R$_c$ and the data symbol R$_s$, is usually selected to be an integer in practical SS system:

$$SF = G_p = \frac{BW_{SS}}{BW \text{ inf } o} = \frac{Rc}{R_s} = \frac{T_b}{T_c} = N_c$$

To demodulate, the received signal is multiplied by $pn_r$, this is the same PN sequence as $pn_t$ (the pseudo-noise code used in the transmitter), synchronized to the PN sequence in the received signal $rx_b$. This operation is called (spectrum) despreading, since the effect is to undo the spreading operation at the transmitter. The multiplier output in the receiver is then (since $pn_r$ = synchronized $pn_t$)

$$d_r = rx_b.pn_r = (d_t.pn_t).\ pn_t$$

The PN sequence $pn_t$ alternates between the levels $-1$ and $+1$, in the example

$$pn_t = +1 +1+1 -1+1-1-1$$

The alternation is destroyed when the PN sequence $pn_t$ is multiplied with itself (perfectly synchronized), because:

$$pn_t .pn_t = +1 \text{ for all } t$$

Thus,

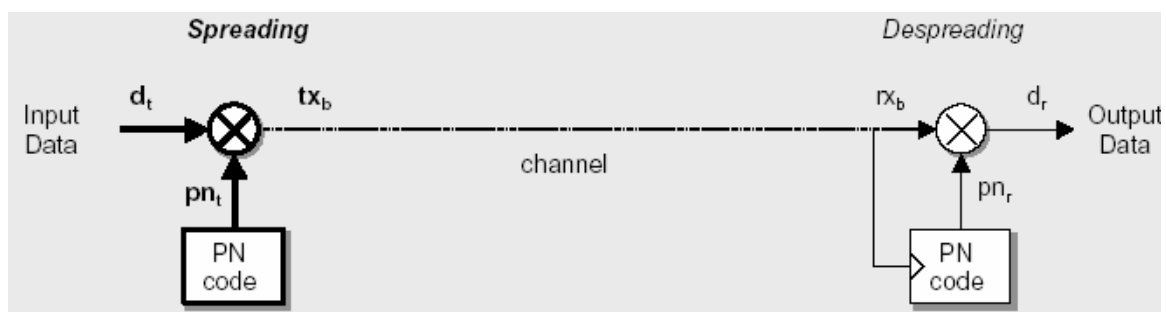$$\text{Autocorrelation } Ra(\tau = 0) = \text{average } (pn_t.pn_t) = +1$$

The data signal is produced at the multiplier output:

$$d_r = d_t$$

If the PN sequence at the receiver is not synchronized properly to the received signal, the data cannot be recovered.

## 3.2 Demodulation



## 3.2.1 $pn_r \neq pn_t$

If the received signal is multiplied by a PN sequence $pn_r$, different from the one

used in the modulator, the multiplier output becomes:

$$d_r = rx_b \cdot pn_r = (d_t \cdot pn_t) \cdot pm_r$$

In the receiver, detection of the desired signal is achieved by correlation against a local reference PN sequence. For secure communication in a multi-user environment, the transmitted data $d_t$ may not be recovered by a user that does not know the PN sequence $pn_t$ used at the transmitter. Therefore:

crosscorrelation $R_c(\tau)$ = average $(pn_t \cdot pn_r) \ll 1$      for all $\tau$

is required. This orthogonal property of the allocated spreading codes means that the output of the correlator used in the receiver is approximately zero for all except the desired transmission.

### 3.2.2 $pn_r = pn_t$
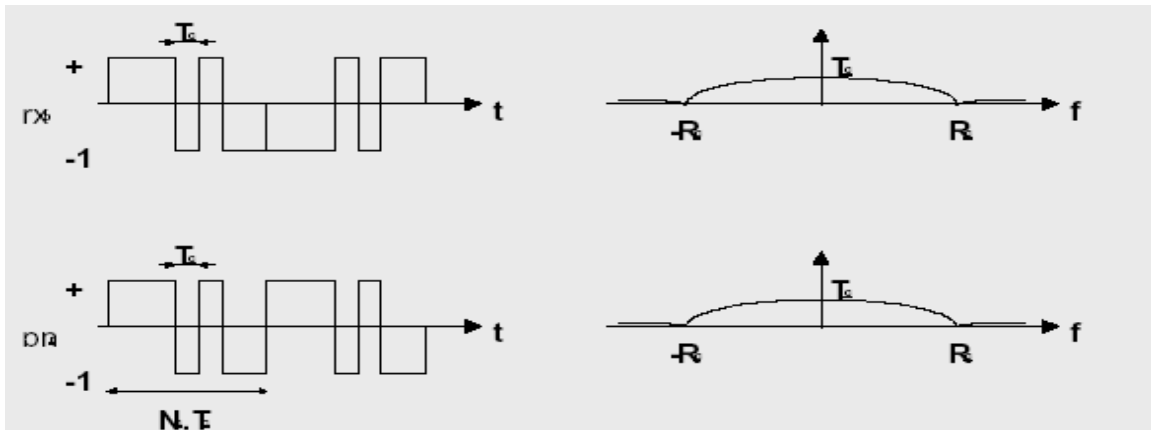


# 4. Performance in the presence of interference

Considering the spread spectrum system for baseband QPSK communication (without filtering).

The received signal rx$_b$ consists of the transmitted signal tx$_b$ plus an additive interference I (noise, other users, jammer ...):

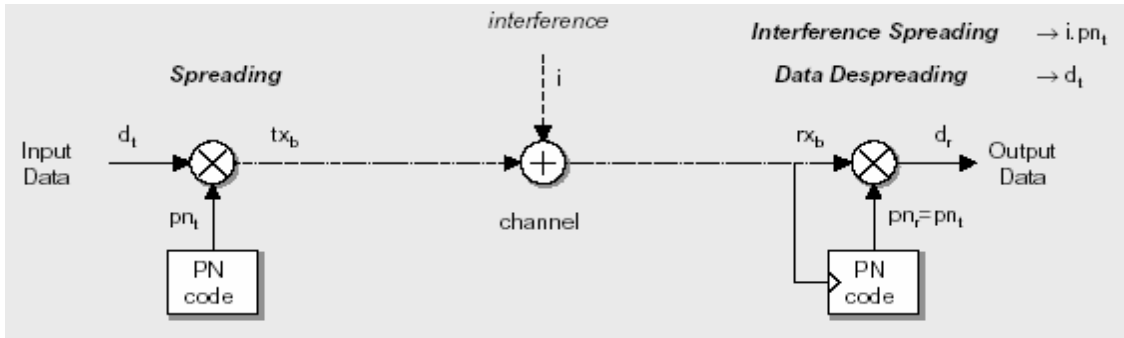$$rx_b = tx_b + i = d_t . pn_t + i$$

To recover the original data d$_t$, the received signal rx$_b$ is multiplied with a locally generated PN sequence pn$_r$ that is an exact replica of that used in the transmitter (that is pn$_r$ = pn$_t$ and synchronized). The multiplier output is therefore given by:

$$d_r = rx_b . pn_t = d_t . pn_t . pn_t + i.pn_t$$

The data signal d$_t$ is multiplied twice by the PN sequence pn$_t$, whereas the unwanted interference 'i' is multiplied only once. Due to the property of the PN sequence:

pn$_t$. pn$_t$ = +1 for all t

The multiplier output becomes:

$$d_r = d_t + i .pn_t$$

The data signal d$_t$ is recovered at the multiplier output in the receiver, except for the interference represented by the additive term 'i. pn$_t$'. Multiplication of the interference 'i' by the locally generated PN sequence means that the spreading code will affect the interference just as it did with the information bearing signal at the transmitter. Noise and interference, being uncorrelated with the PN sequence, become noise–like, increase in bandwidth and decrease in power density after the multiplier.

After despreading, the data component d$_t$ is narrow band (R$_s$) whereas the interference component is wideband (R$_c$). By applying the d$_r$ signal to a baseband (low-pass) filter with a bandwidth just large enough to accommodate the recovery of the data signal, most of the interference component 'i' is filtered out. The effect

of the interference is reduces by the processing gain ($G_p$).

## 4.1 Narrowband interference

The narrowband noise is spread by the multiplication with the PN sequence $pn_r$ of the receiver. The power density of the noise is reduced with respect to the despread data signal. Only $1/G_p$ of the original noise power is left in the nformation baseband ($R_s$).

Spreading and despreading enables a bandwidth trade for processing gain against narrow band interfering signals. Narrowband interference would disable conventional narrowband receivers.

The essence behind the interference rejection capability of a spread spectrum system: the useful signal (data) gets multiplied twice by the PN sequence, but the interference signal gets multiplied only once.
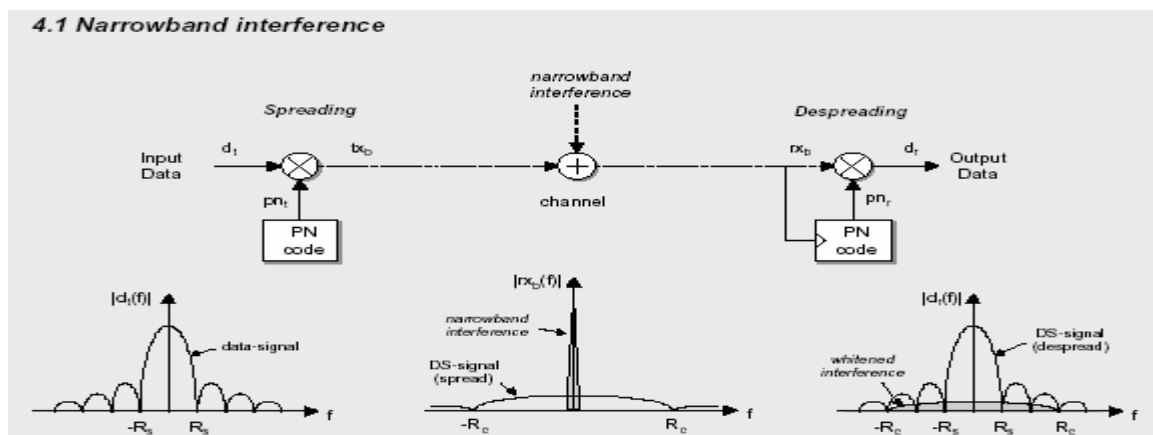


## 4.2 Wideband interference

Multiplication of the received signal with the PN sequence of the receiver gives a selective despread of the data signal (smaller bandwidth, higher power density).
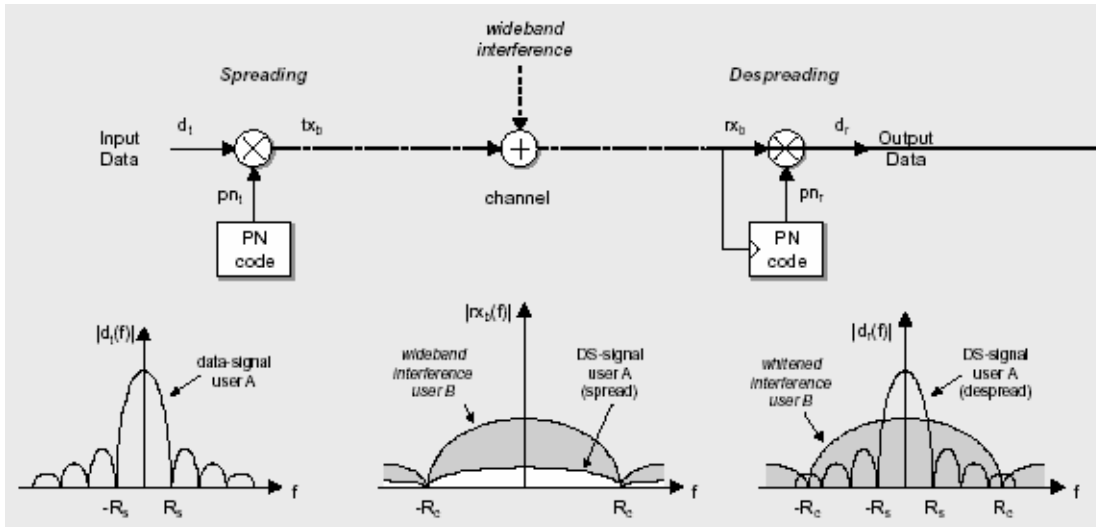
The interference signal is uncorrelated with the PN sequence and is spread.

Origin of wideband noise:

- Multiple spread spectrum users: multiple access mechanism.
- Gaussian Noise: There is no increase in SNR with spread spectrum. The larger channel bandwidth ($R_c$ instead of $R_s$) increases the received noise power with $G_p$:

$$N_{info} = N_{0.}\ BW_{info} \rightarrow N_{ss} = N_0.BW_{ss} = N_{info}.G_p$$

The spread spectrum has a lower power density than the directly transmitted signa

# 5. Spreading Codes

## 5.1 Pseudo-Noise Sequences PN

A pseudo-noise code (PN sequence) acts as a noiselike (but deterministic) carrier used for bandwidth spreading of the signal energy. The selection of a good code is important, because type and length of the code sets bounds on the system capability.

The PN code sequence is a Pseudo-Noise or Pseudo-Random sequence of 1's and 0's, but not a real random sequence (because periodic). Random signals can not be predicted.

The autocorrelation of PN code has properties similar to those of white noise.

*Pseudo-Random*:

- Not random, but it looks randomly for the user who doesn't know the code
- Deterministic, periodical signal that is known to both the transmitter and the receiver. The longer the period of the PN spreading code the closer will the transmitted signal be truly random binary wave and the harder it is to detect.
- Statistical properties of sampled white-noise.

*Length*:

- Short code : The same PN sequence for each data symbol ($N_c, T_c = T_s$)
- Long code : The Pn sequence period is much longer than the data symbol, so that a different chip pattern is associated with each symbol ($N_c, T_c \gg T_s$)

## 5.1.1 Properties of PN sequences

Balance Property

In each period of the sequence the number of binary ones differs from the number of binary zeros by at most one digit (for Nc odd).

$$pn = +1 \ +1 \ +1 \ -1 \ +1 \ -1 \ -1 \quad \rightarrow \Sigma = +1$$

When modulating a carrier with a PN coding sequence, one zero balance (DC component) can limit the degree of carrier suppression obtainable, because carrier suppression is dependent on the symmetry of the modulating signal.

Run-length Distribution

A run is a sequence of a single type of binary digits. Among the runs of ones and zeros in each period it is desirable that about one half the runs of each type are length 1, about one-fourth are of length 2, one eight are of length 3, and so on.

Autocorrelation

The Origin of the name pseudo-noise is that the digital signal has an

autocorrelation function which is very similar to that of a white noise signal (impulse like).

The autocorrelation function for the periodic sequence pn is defined as the number of agreements less the number of disagreements in a term by term comparison of one full period of the sequence with a cyclic shift (position τ) of the sequence itself:

$$\text{Ra}(\tau) = \int\limits_{-NcTc/2}^{NcTc/2} pn(t).pn(t+\tau)dt$$

```
pn(0) =    +1  +1  +1  -1  +1  -1  -1
pn(0) =    +1  +1  +1  -1  +1  -1  -1
           +1  +1  +1  +1  +1  +1  +1  = Σ = 7 = Ra(τ=0)


pn(0) =    +1  +1  +1  -1  +1  -1  -1
pn(1) =    +1  +1  -1  +1  -1  -1  +1
           +1  +1  -1  -1  -1  +1  -1  = Σ = -1= Ra(τ=1)
```
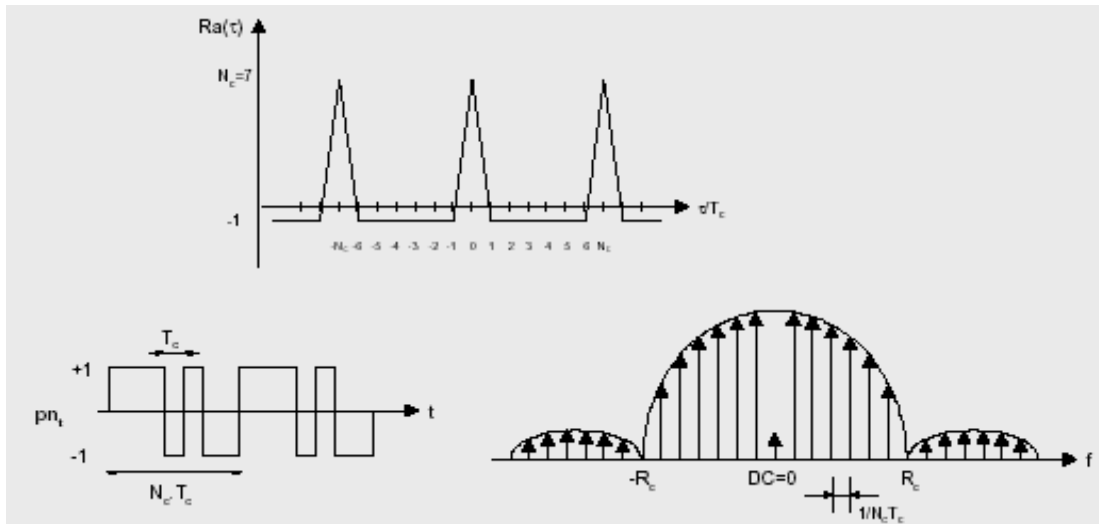


For PN sequences the autocorrelation has a large peaked maximum (only) for perfect synchronization of two identical sequences (like white noise). The synchronization of the receiver is based on this property.

Frequency spectrum

 Due to the periodic nature of the PN sequence the frequency spectrum has spectral lines which become closer to each other with increasing sequence length

Nc. Each line is further smeared by data scrambling, which spreads each spectral line and further fills in between the lines to make the spectrum more nearly continuous. The DC component is determined by the zero-one balance of the PN sequence.

Cross-correlation

Cross-correlation describes the interference between codes $pn_i$ and $pn_j$:

$$Rc(\tau) = \int_{-NcTc/2}^{NcTc/2} pn_i(t).pn_j(t+\tau)dt$$

Cross-correlation is the measure of agreement between two different codes $pn_i$ and $pn_j$. When the cross-correlation $Rc(\tau)$ is zero for all $\tau$, the codes are called orthogonal. In CDMA multiple users occupy the same RF bandwidth and transmit simultaneous. When the user codes are orthogonal, there is no interference between the users after dispreading and the privacy of the communication of each user is protected.

In practice, the codes are not perfectly orthogonal hence the cross-correlation between user codes introduces performance degradation (increased noise power after dispreading), which limits the maximum number of simultaneous users.

## 5.1.2 M – Sequence

A simple shift register Generator (SSRG) has all the feedback signals returned to a single input of a shift register (delay line). The SSRG is linear if the feedback function can be expressed as a modulo -2 sum (xor).

SSRG [5,3]

..., $a_i$, $a_{i+1}$, $a_{i+2}$, ...



reverse SSRG [5,2]

..., $a_{i+2}$, $a_{i+1}$, $a_i$, ...

[5,2] = -1 -1 -1 -1  1 -1  1 -1  1  1  1 -1  1  1 -1 -1 -1  1  1  1  1  1 -1 -1  1  1 -1  1 -1 -1  1

[5,3] = -1 -1 -1 -1  1 -1 -1  1 -1  1  1  1 -1 -1  1  1  1  1  1 -1 -1 -1 -1  1  1 -1  1  1  1 -1  1 -1  1  1
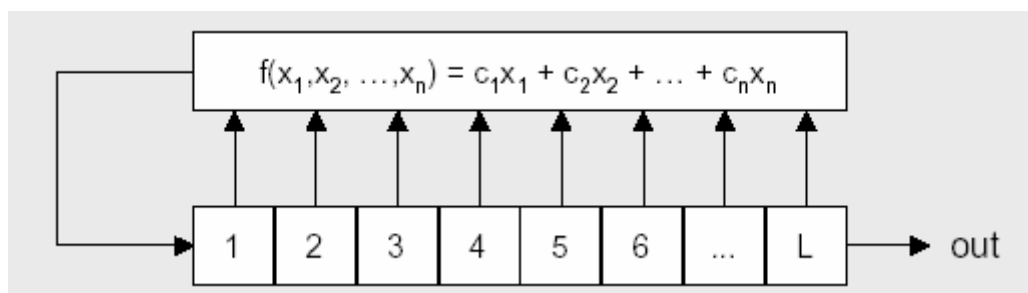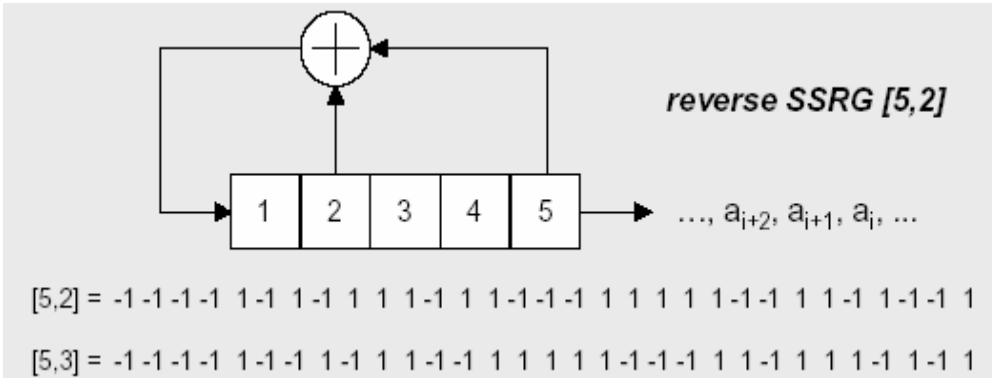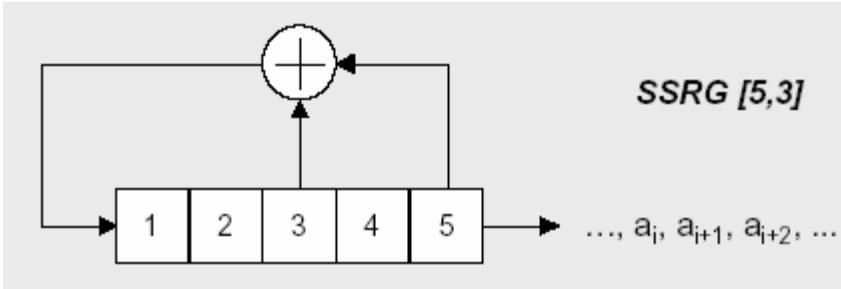
In the following table the feedback connections (even number) are tabulated for m sequences generated with a linear SSRG (without image set).

| L | $N_c = 2^L - 1$ | Feedback taps for m – sequences | # m- sequences |
|---|---|---|---|
| 2 | 3 | [2,1] | 2 |
| 3 | 7 | [3,1] | 2 |
| 4 | 15 | [4,1] | 2 |
| 5 | 31 | [5,3][5,4,3,2][5,4,2,1] | 6 |
| 6 | 63 | [6,1][6,5,2,1][6,5,3,2] | 6 |
| 7 | 127 | [7.1][7.3][7,3,2,1][7,4,3,2] [7,6,4,2][7,6,3,1][7,6,5,2] | 18 |

| | | | |
|---|---|---|---|
| | | [7,6,5,4,2,1][7,5,4,3,2,1] | |
| 8 | 255 | [8,4,3,2][8,6,5,3][8,6,5,2]<br>[8,5,3,1][8,6,5,1][8,7,6,1]<br>[8,7,6,5,2,1][8,6,4,3,2,1] | 16 |
| 9 | 511 | [9,4][9,6,4,3][9,85,4][9,8,4,1]<br>[9,5,3,2][9,8,6,5][9,8,7,2]<br>[9,6,5,4,2,1][9,7,6,4,3,1]<br>[9,8,7,6,5,3] | 48 |
| 10 | 1023 | [10,3][10,8,3,2][10,4,3,1][10,8,5,1]<br>[10,8,5,4][10,9,4,1][10,8,4,3]<br>[10,5,3,2][10,5,2,1][10,9,4,2]<br>[10,6,5,3,2,1][10,9,8,6,3,2]<br>[10,9,7,6,4,1][10,7,6,4,2,1]<br>[10,9,8,7,6,5,4,2][10,8,7,6,5,4,3,1] | 60 |

For every set [L, k… p] feedback taps listed in the table, there exists an image set (reverse set) of feedback taps[L,L-k ….. L-p] that generates an identical sequence reversed in time.

## 5.1.3 Properties of M - Sequence

For an m-sequence there is one "one" more than "zero" in a full period of the sequence. Since all states but the 'all-zero' state are reached in an m-sequence, there must be $2^{L-1}$ "once" and $2^{L-1}-1$ "zeros".

Run-length distribution

For every m-sequence period, half the runs (of all 1's or all 0's) have length 1, one fourth have length 2, one eight have length 3, etc. For each of the runs there are equally many runs of 1's and 0's.

<u>Autocorrelation</u>

The autocorrelation function of the m-sequence is $-1$ for all values of the chip phase shift $\tau$, except for the $[-1,+1]$ chip phase shift area, in which correlation varies linearly from the $-1$ value to $2^L-1=Nc$ (the sequence length)

The autocorrelation peak increases with increasing length Nc of the m-sequence and approximates the autocorrelation function of white noise.

Other codes can do no better than equal this performance of m-sequences.

<u>Crosscorrelation</u>

Crosscorrelation is the measure of agreement between two different codes. Unfortunately crosscorrelation is not as well behaved as autocorrelation. When large numbers of transmitters, using different codes, are to share a common frequency band (multi-user environment), the code sequences must be carefully chosen to avoid interference between users.

<u>Security</u>

The m-sequence codes are linear and thus not usable to secure a transmission system. The linear codes are easily decipherable once a short sequential set of chips (2L+1) from the sequence is known.(The overall system could still be secure if the information itself where encoded by a cryptographically secure technique).

## 5.2 Gold Codes

The autocorrelation properties of the m-sequence can not be bettered. But a multi-user environment (Code Division Multiple Access) needs a set of codes with the same length and with good cross correlation properties.

Gold code sequences are use full because a large number of codes (with the same length and with controlled crosscorrelation) can be generated, although they require only one pair of feedback tap sets.
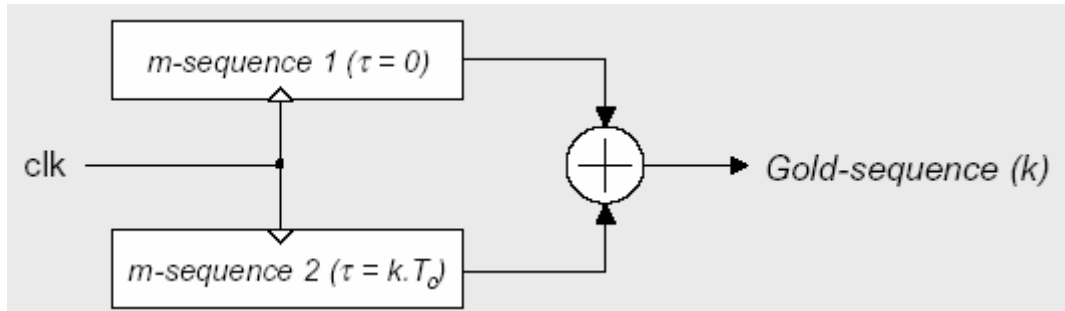
Gold codes are product codes achieved by the exclusive or-ing(modulo-2 adding) of two maximum-length sequences with the same length (factor codes). The code sequences are added chip by chip by synchronous clocking. Because the m-sequences are of the same length, the two code generators maintain the same phase relationship, and the codes generated are of the same length as the two base codes which are added together, but are nonmaximal (so the autocorrelation function will be worse than that of m-sequences). Every change in phase position between the two generated m-sequences causes a new sequence to be generated.

Any 2-register Gold code generator of length L can generate $2^L-1$ sequence (length $2^L-1$) plus the two base m-sequences, giving a total of $2^L+1$ sequences. In addition to their advantage in generating large number of codes, the gold codes may be chosen so that over a set of codes available from a given generator the autocorrelation and the crosscorrelation between the codes are uniform and bounded. When specially selected m-sequences, called preferred m-sequences, are used the generated Gold codes have a three valued crosscorrelation.

| L | $N_c$ | Normalized crosscorrelation 3-valu | Frequency of occurrence |
|---|---|---|---|
| odd | 2-1 | $-1/N_c$ | ~0.50 |
| | | $-(2^{L+1/2}+1)/N_c$ | ~0.25 |
| | | $(2^{(L+1)/2}-1)/N_c$ | ~0.25 |

| Even (not k.4) | 2-1 | $-1/N_c$ | ~0.75 |
|---|---|---|---|
| | | $-(2^{L+2/2}+1)/N_c$ | ~0.125 |
| | | $(2^{(L+2)/2}-1)/N_c$ | ~0.125 |

This important subset of gold codes is the preferred pair gold codes.

Predictable cross-correlation properties are necessary in an environment where one code must be picked from several codes which exist in the spectrum.

Only part of the generated Gold codes is balanced.

[5, 4, 3, 2] = - 1 - 1 - 1 - 1 1 - 1 1 1 - 1 1 - 1 1 - 1 - 1 1 1 1 - 1 1 1 1 1 1 1 1 - 1 - 1 - 1 - 1 1 1

## 5.3 Hadamard-Walsh Codes

The Hadamard-Walsh codes are generated in a set of $N=2^n$. The generating algorithm is simple:

$$H_N = \begin{bmatrix} H_{N/2} & H_{N/2} \\ H_{N/2} & -H_{N/2} \end{bmatrix} \text{ with } H_0 = [1]$$

The rows (or columns) of the matrix $H_N$ are the Hadamard-Walsh codes. In each case the first row (row 0) of the matrix consists entirely of 1s and each of the other rows contains N/2 0s and N/2 1s. Row N/2 starts with N/2 1s and ends with N/2 0s.

$$H_2 = \begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix} \quad H_4 = \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & -1 & 1 & -1 \\ 1 & 1 & -1 & -1 \\ 1 & -1 & -1 & 1 \end{bmatrix} \quad H_8 = \begin{bmatrix} 1 & 1 & 1 & 1 & 1 & 1 & 1 & 1 \\ 1 & -1 & 1 & -1 & 1 & -1 & 1 & -1 \\ 1 & 1 & -1 & -1 & 1 & 1 & -1 & -1 \\ 1 & -1 & -1 & 1 & 1 & -1 & -1 & 1 \\ 1 & 1 & 1 & 1 & -1 & -1 & -1 & -1 \\ 1 & -1 & 1 & -1 & -1 & 1 & -1 & 1 \\ 1 & 1 & -1 & -1 & -1 & -1 & 1 & 1 \\ 1 & -1 & -1 & 1 & -1 & 1 & 1 & -1 \end{bmatrix}$$

The distance (number of different elements) between any pairs of rows is exactly N/2.

For $H_8$ the distance between any two rows is 4, so the Hamming distance of the Hadamard code is 4. The Hadamard–Walsh code can be used as a block code in a channel encoder: each sequence of n bits identifies one row of the matrix (there are $N = 2^n$ possible rows)

All rows are mutually orthogonal:

$$\sum_{k=0}^{N-1} h_{ik}.h_{jk} = 0 \quad \text{for all rows I and j}$$

The cross-correlation between any two Hadamard-Walsh codes of the same matrix is zero, when perfectly synchronized. In a synchronous CDMA this ensures that there is no interference among signals transmitted by the same station.

Only when synchronized, these codes have good orthogonal properties. The codes are periodic, which results in less spreading efficiency and problems with synchronization based on autocorrelation.

# 6. Channel

The physical medium between the transmitter and receiver is known as channel. This channel results in random delay (random phase shift) with total a factor

## 6.1 Additive White Gaussian Noise

Zero-mean white Gaussian Noise (WGN) has the same power spectral density $G_{WGN}(f)$ for all frequencies. The adjective 'white' is used in the sense that white light contains equal amounts of all frequencies within the visible band of electromagnetic radiation.

The autocorrelation function of WGN is given by the inverse Fourier transform of the noise power spectral density $G_{WGN}(f)$:

$$Ra_{WGN}(\tau) \int_{-\infty}^{+\infty} gwn(T).gwn(t+\tau)dt = F^{-1}\{G_{WGN}(f)\} = \frac{N_0}{2}\delta(\tau)$$

The autocorrelation function RaWGN ($\tau$) is zero for $\tau \neq 0$. This means that any two different samples of WGN, no matter how close together in time they are taken, are uncorrelated. The noise signal WGN (t) is totally decorrelated from its time

shifted version for any τ≠0.
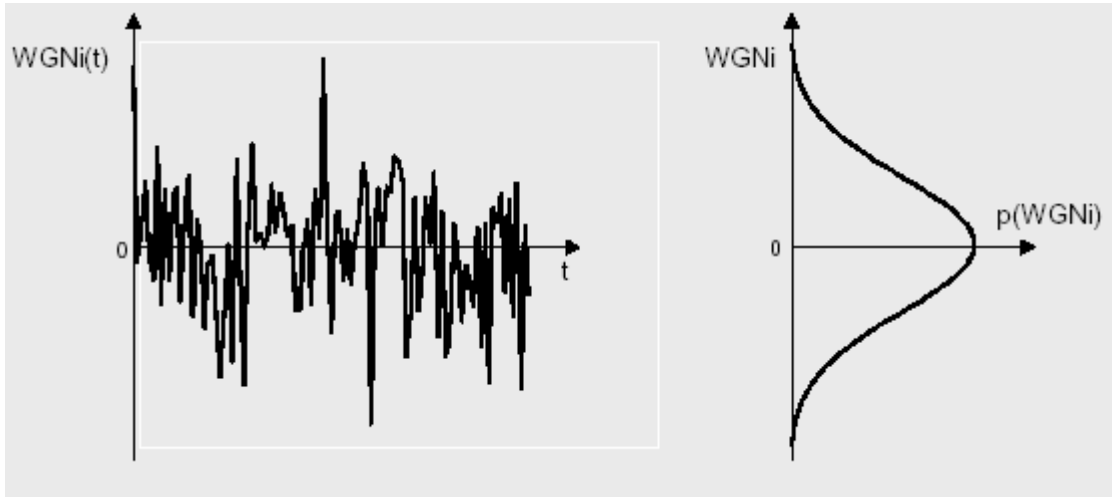


The amplitude of 'integrated' (bandwidth) WGN has a gaussian probability density distribution P(WGNi):

$$P(\text{WGNi}) = \frac{1}{\sigma\sqrt{2\pi}} e^{\left[-\frac{1}{2}\left(\frac{n}{\sigma}\right)^2\right]}$$
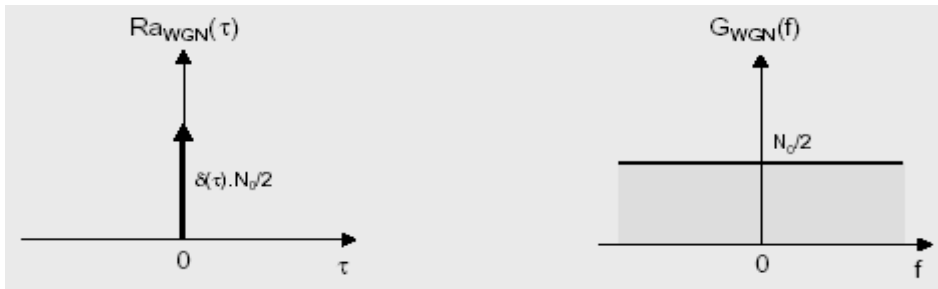


## 6.2 Fading in Communication Channels

Fading is the term used to describe the rapid fluctuations in the amplitude of the received radio signal over a short period of time. Fading is a common phenomenon in Mobile Communication Channels, where it is caused due to the interference between two or more versions of the transmitted signals which arrive at the

receiver at slightly different times. The resultant received signal can vary widely in amplitude and phase, depending on various factors such as the intensity, relative propagation time of the waves, bandwidth of the transmitted signal etc.

## A Mathematical Model

Consider a transmitted signal $s(t) = A \cos(2*pi*fct)$ through a fading channel. The received signal can be expressed as (ignoring the effects of noise):

(1)
$$y(t) = A \sum_{i=1}^{N} a_i \cos(2\pi f_c t + \theta_i)$$

Where,

- $a_i$ is the attenuation of the $i^{th}$ multipath component.
- Theta_i is the phase-shift of the $i^{th}$ multipath component.

It must be noted that ai and Theta_i are random variables. The above expression can be re-written as:

(2)

$$y(t) = A \left\{ \left( \sum_{i=1}^{N} a_i \cos(\theta_i) \right) \cos(2\pi f_c t) - \left( \sum_{i=1}^{N} a_i \sin(\theta_i) \right) \sin(2\pi f_c t) \right\}$$

We introduce two random processes X1(t) and X2(t), such that the above equation becomes:

(3)

$$y(t) = A \{ X_1(t) \cos(2\pi f_c t) - X_2(t) \sin(2\pi f_c t) \}$$

If the value of N is large (i.e., a large number of scattered waves are present), invoking the Central-Limit Theorem, we get approximate X1(t) and X2(t) to be gaussian random variables with zero mean and variance sigma square. The expression 3 can be rewritten as:

(4)

$$y(t) = A R(t) \cos(2\pi f_c t + \theta(t))$$

Where,

- The amplitude of the received waveform R(t) is given by:

(5)

$$R(t) = \sqrt{X_1(t)^2 + X_2(t)^2}$$

Since the processes X1(t) and X2(t) are gaussian, it can be shown that R(t) has a Rayleigh

Distribution with a probability density function (pdf) given by:

(6)

$$f_R(r) = \frac{r}{2\sigma^2} e^{\frac{-r^2}{2\sigma^2}}, \qquad r > 0$$

The phase of the received waveform Theta(t) is given by:

(7)

$$\theta(t) = \tan^{-1}\left(\frac{X_2(t)}{X_1(t)}\right)$$

Since the processes X1(t) and X2(t) are gaussian, it can be shown that theta(t) has a Uniform Distribution with a probability density function (pdf) given by:

(8)

$$f_\theta(\theta) = \frac{1}{2\pi}, \qquad -\pi \leq \theta \leq \pi$$

The distortion in the phase can be easily overcome if differential modulation in employed. It is the amplitude distortion R(t) that severely degrades performance of digital communication systems over fading channels. It is usually reasonable to assume that the fading stays essentially constant for at least one signaling interval.

## Diversity Combining

One of the most efficient and simple techniques to overcome the destructive effects of fading is Diversity. Diversity is an efficient technique to exploit the random nature of radio propagation by finding methods to generate and extract independent signal paths for communication. The concept behind diversity is

relatively simple: *if one signal path undergoes a deep fade at a particular point of time, another independent path may have a strong signal*. By having more than one path to select from, both the instantaneous and average SNR can be improved in the receiver by a large amount. There are various types of diversity used in communication systems operating over fading channels. They are:

· Space Diversity.
· Frequency Diversity.
· Time Diversity.
· Polarization Diversity.
· Multipath Diversity.

Whatever be the diversity technique employed, the receiver has to process the diversity signals obtained in a fashion that maximizes the power efficiency of the system. There are several possible diversity reception methods employed in communication receivers. The most common techniques are:

· Selection Diversity.
· Equal Gain Combining (EGC).
· Maximal Ratio Combining (MRC).

These methods are discussed in the following sections.

### Selection  Diversity.

Selection Diversity is one of the most simple diversity techniques. The receiver simply picks the signal with the largest SNR as given in fig. below. In the case of a two-fold

diversity, the diversity combining strategy for selective combining is given by:

$$Z_k = \begin{cases} Z_{1k}, & \text{if } |Z_{1k}| > |Z_{2k}| \\ Z_{2k}, & \text{if } |Z_{2k}| > |Z_{1k}| \end{cases}$$

Where, Z1k and Z2k are decision variables at the first and second diversity paths and Zk is the decision variable at the output of the diversity combiner.
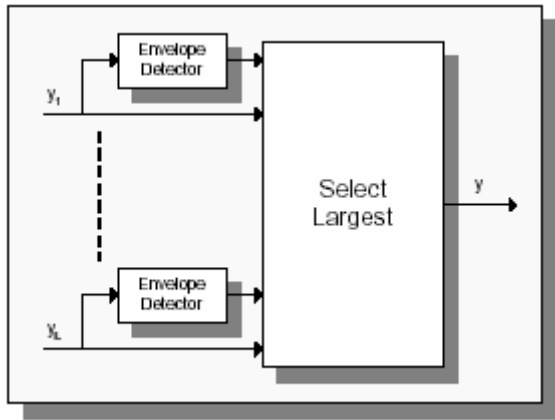


Figure. Selection Diversity

Let Roh be the threshold SNR that must be achieved for proper demodulation and detection of the received signal. Let there be L diversity branches and let k be the instantaneous SNR for the kth branch. Using the expression for the pdf for the instantaneous SNR ,we can get an expression for the outage probability (probability that the SNR falls below the threshold) for all the branches as:

$$P(\gamma_1, \ldots, \gamma_L \leq \rho) = (1 - e^{-\rho/\bar{\gamma}_c})^L$$

Where, gamma(c) is the average SNR at every diversity branch, assuming all the branches have the same average SNR. It is given by:

$$\bar{\gamma}_c = \frac{E_b}{N_o} E\{R^2\}$$

The improvement in the SNR due to selective combining is evident from the above equation.

## Equal Gain Combining (EGC)

In Equal Gain Combining (EGC), all the received signals are co-phased at the receiver and added together without any weighting. The performance of EGC is only marginally inferior to the optimal maximal ratio combiner. In case of a two-fold diversity scheme, the combining equation is given by:

$$Z_k = Z_{1k} + Z_{2k}$$

## Maximal Ratio Combining (MRC).

In Maximal Ratio Combining (MRC), the signal all the branches are co-phased and individually weighed to provide the optimal SNR at the output.It can be shown that the output SNR is maximized when the signals in each of the diversity branches are weighed by their own envelopes. In case of a two-fold diversity scheme, the combining equation is given by:

$$Z_k = r_{1k}Z_{1k} + r_{2k}Z_{2k}$$

Where, r1k and r2k represent the instantaneous envelopes of the signals received at each of the diversity branches. The SNR per bit at the output of the maximal ratio combiner (b) can be written as:

$$\gamma_b = \sum_{k=1}^{L} \gamma_k = \frac{E_b}{N_o} \sum_{k=1}^{L} R_k^2$$

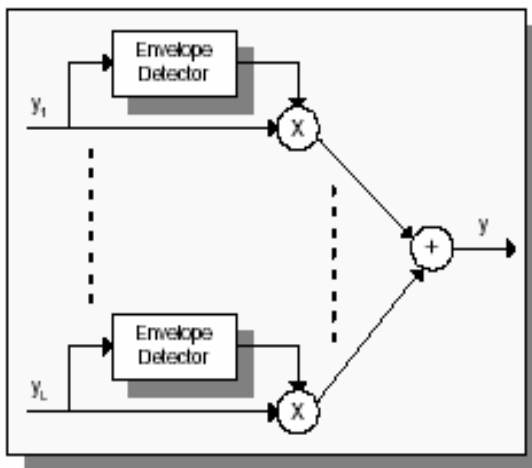where k = R2(Eb/No) is the instantaneous SNR in the kth diversity branch.
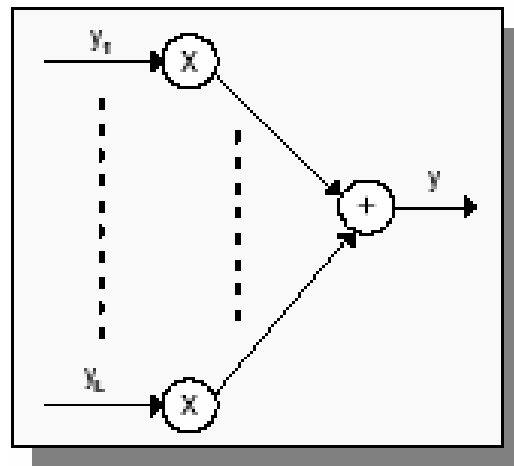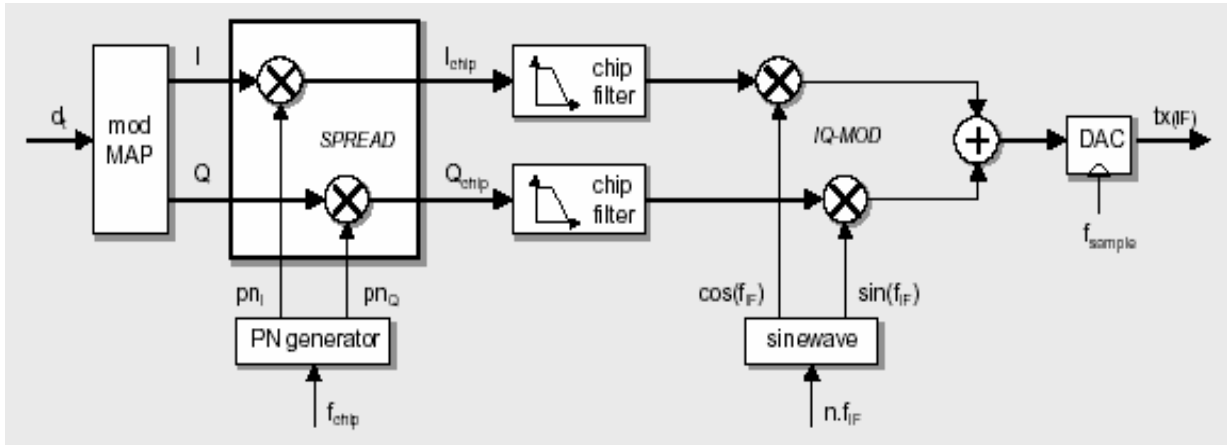


Figure. Maximal Ratio Combining (MRC)

Figure. Equal Gain Combining (EGC)

# 7. Transmitter Architecture

A typical architecture of a direct sequence spread spectrum (DS-SS) transmitter.



# 8. Receiver Architecture

A typical architecture of a direct sequence Spread Spectrum (DS-SS) receiver:
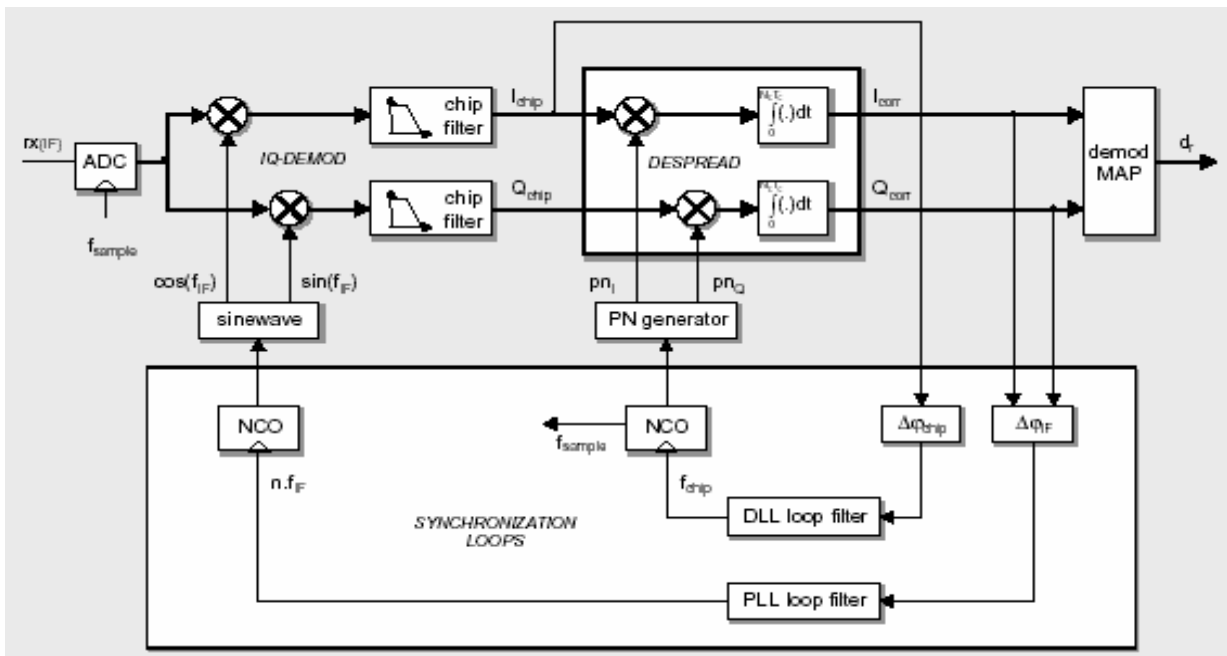
The basic building blocks of a DS-SS (digital) receiver are:

- Coherent IQ vector-demodulator with waveform synthesizer (Direct Digital Synthesis) at the IF-Carrier frequency ($f_{if}$) and chip matched filters (usually square root raised cosine)

- Dispreading (correlation of the received symbols with the locally generated PN sequence(s) $pn_i$ and $pn_Q$)

- Decorrelated 'IQ to data' demodulator mapping

- Synchronization loops for the IF carrier (fif phase error, $\Delta\varphi_{iF}$ measured after dispreading to reduce the influence of noise) and chip frequency ($f_{chip}$)

# 9. PN decorrelators

Two PN decorrelators architecture can be used for dispreading spread spectrum signal: the matched filter and the active correlator. They are optimum from a SNR point of view.

## 9.1 PN Matched filter

A typical matched filter implements convolution using a finite impulse response filter (FIR) whose coefficients are the time reverse of the expected PN sequence, to decode the transmitted data.
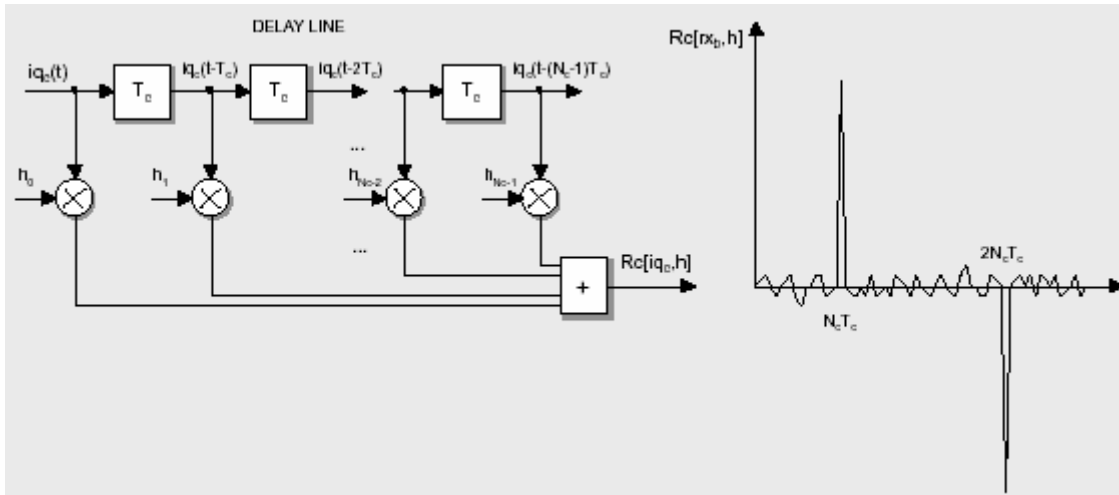
For the given example:

$Pn_t = [pn0 \ pn_1 \ pn_2 \ pn_3 \ pn_4 \ pn_5 \ pn_6] = [+1 \ +1 \ +1 \ -1+1\text{-}1\text{-}1]$

$h = [h0 \ h_1 h_2 h_3 h_4 h_5 h_6] = [\text{-}1 \ -1+1\text{-}1+1+1+1]$

The output of the FIR filter is the convolution of the received IQ-demodulated and filter signal $iq_c$ ($I_{chip}$ or $Q_{chip}$ on the receiver architecture block diagram) with the FIR impulse response $h=[h_0 \ h_1 \ .....h_{nc\text{-}1}]$. Due to the time reversion, the output of the filter is the correlation of $rx_b$ with the local PN sequence.

$$Rc(\tau)= \sum_{i=0}^{N_C-1} iq_c \,(t - i.Tc).h_i = \sum_{i=0}^{N_C-1} iq_c \,(t - i.T_c).pN_{c-1-i}$$

In the shown example, 1 sample of the received signal per chip is taken ($f_{sample}=f_{chip}$). To increase the accuracy of synchronization oversampling with a factor s can be used. In the case there are s sample per chip ($f_{sample}=s.f_{chip}$). The dimension of the matched filter are also increased with a factor s (each filter coefficient $h_i$ is used  time.)

If the receiver is not synchronized, then the received signal will propagate through the matched filter, which outputs the complete correlation function. The large peak confirms that the correct code is indeed being received and provides accurate timing information for the synchronization of the received signal. The output $R_c$ of

the FIR PN matched filter is immediately the decorrelated data: the polarity of the large correlation peaks indicates the data value.

## 9.2 PN active Correlator (Integrate and Dump)

When timing information is already available, then the simpler active correlator receiver can be used. This receiver only operates correctly when the local PN sequence $pn_r$ is accurately matched and correctly timed, with respect to the spreading code within the received signal $rx_b$. Synchronization can be obtained by sliding the reference signal through the received signal. This can be extremely slow process, however, for large spreading waveforms (long codes)
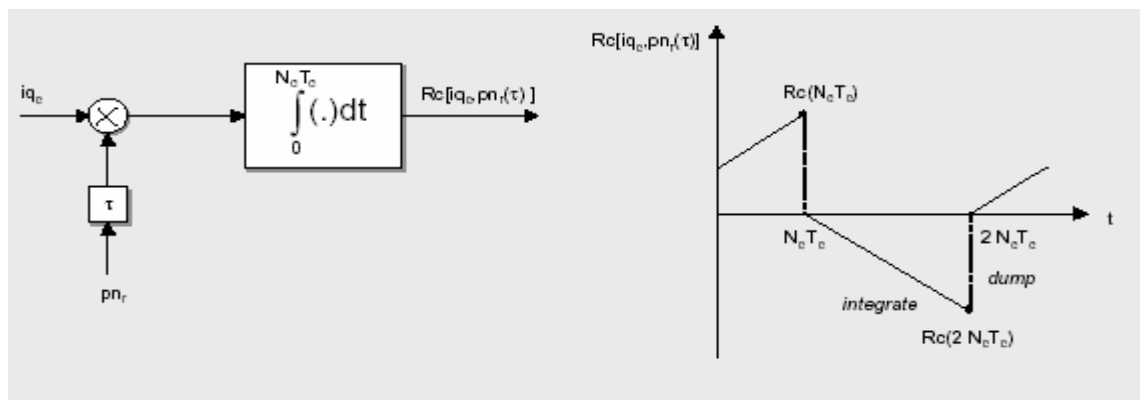


# 10. Evaluation of SS

Positive

1. Signal hiding (lower power density, noise-like), non-interference with conventional systems and other SS systems
2. Secure Communication (privacy)
3. Code Division Multiple Access CDMA (multi-user)
4. Mitigation (rejection) of multipath, hold only the direct path
5. Protection to intentional interference (Jamming)
6. Rejection of unintentional interference (narrowband)
7. Low probability of detection and interception (LP1)
8. Availability of license-free ISM (Industrial, Scientific and Medical)

frequency-bands

## Negative

9. No improve in performance in the presence of Gaussian noise
10. Increased bandwidth (frequency usage, wideband receiver)
11. Increased Complexity and computational load

# 1.Introduction to CDMA

Code Division Multiple access (CDMA) is a method of multiplexing (wireless users by district (orthogonal) codes. All users can transit at the same time and each is allocated the entire available frequency spectrum for transmission. CDMA is also know as spread-spectrum multiple access SSMA.

CDMA does not require the bandwidth allocation of FDMA, nor the time synchronization of the individual users needed in TDMA. A CDMA user has full time and full bandwidth available, but the quality of the communication decreases with an increasing number of users (BER).

In CDMA each user:

- Has its own PN code
- Uses the same RF bandwidth
- Transmit simultaneously (asynchronous or synchronous)
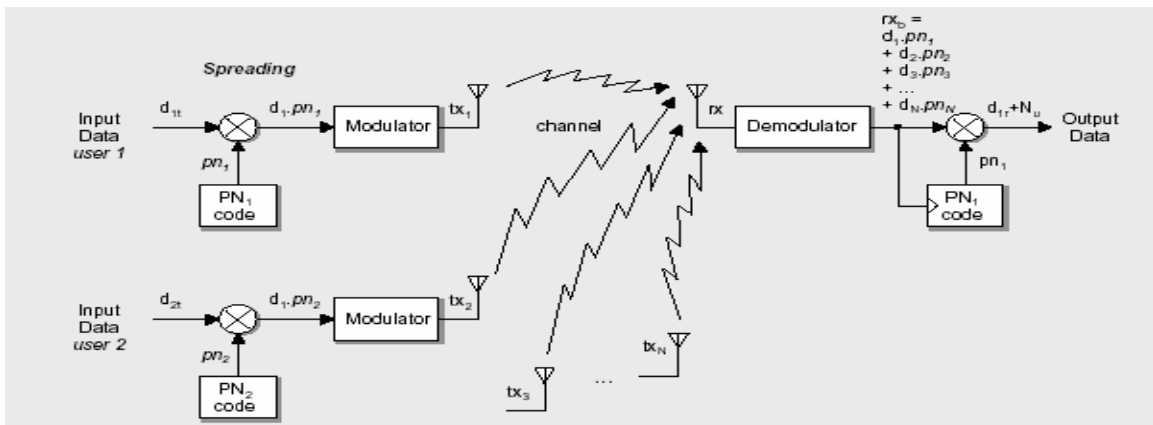


Correction of the received baseband spread spectrum signal $rx_b$ with the PN sequence of user 1 only despreads the signal of user 1.
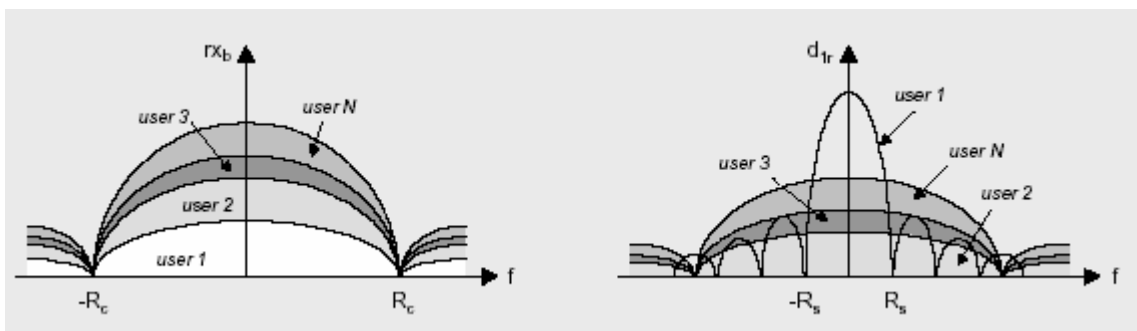
The other users produce noise $N_u$ for user 1.

Only that portion of the noise produced by the other users falling in the information bandwidth [$R_s$, $_R$s] of the receiver will cause interference with the desired signal.

The set of PN codes must have the following properties:

- o Autocorrelation for good synchronization
- o Low crosscorrelation (orthogonal codes) for low MAI

Useful codes are:

- Gold codes, Kasami codes (asynchronous CDMA)
- Hadamard-Walsh codes(synchronous CDMA)

# 2. Multiple access comparison

It is easier to understand CDMA if it is compared with other multiple access technologies. The following sections describe the fundamental difference between a Frequency Division Multiple Access Analog Technology (FDMA), a Time Division access digital technology (TDMA) and Code Division Access Digital Technology (CDMA).

## FDMA- Frequency Division Multiple Access

FDMA is used for standard analog cellular. Each user is assigned a discrete slice of the RF spectrum. FDMA permits only one user per channel since it allows the user to use the channel 100% of the time. Therefore, only the frequency "dimension" is used to define channels.

## TDMA – Time Division Multiple Access

The key point to make about TDMA is that users are still assigned a discrete slice of RD spectrum, but multiple users now share that RF carrier on a time slot basis. Each of the users alternates their use of the RF channel. Frequency division is

still employed, but these carriers are now further sub-divided into some number of time slots per carrier.

A user is assigner a particular time slot in a carrier and can only send or receive information at those times. This is true whether or not the other time slots are being used. Information flow is not continuous for any user, but rather is sent and received in bursts" The bursts are reassembled at the receiving end, and appear to provide continuous sound because the process is very fast.

## CDMA – Code Division Multiple Access

Each user is assigned a binary, direct sequence code during a call. The DS code is a signal generated by liner modulation with wideband Pseudorandom Noise (PN) sequences. As result, DC CDMA uses much wider signals than those used in other technologies. Wideband signal reduce interference and allow one-cell frequency reuse. There is no time division, and all users use the entire carrier, all of the time.

## An Illustration ::

To illustrate the conceptual differences among the multiple access technologies,. Picture a large room and a number of people, in pairs, who would like to hold conversations. The people in each pair only want to talk and listen to each other, and have no interest in what is being said by the other pairs. In order for these conversations to take place, however it is necessary to define the environment for each conversation.

First, let us apply this analogy to an FDMA system. An FDMA environment would be simulated by building walls in the single large room, creating a larger number of small rooms. A single pair of people would enter each small room and hold their conversation. When that conversation is complete, the pair of people would leave and another pair would be able to enter that small room.

In a TDMA environment, each of these small rooms would be able to accommodate multiple conversations "simultaneously." For Example, with a 3- slots TDMA system such as IS-54, each "room" would contain up to 3 pairs of people, with each

pair taking turns talking. Think of each pair having the right to speak for 20 seconds during each minute, with pair A able to use 0:01 second through 0:20 second, pair using 0.21 second through 0:40 second, and pair C using 0:41 second through 0:60 second. Even if there are fewer than three pairs in the small room, each pair is still limited to its 20 seconds per minute.

Now, for CDMA, get rid of all of the little room. Pairs of people will enter the single large room. However, if every pair uses a different language, they can all use the air in the room as carrier for their voices and experience little interference from the other pairs. The analogy here is that the air in the room is wideband "carrier" and the languages are represented by the "codes" assigned by the CDMA system. In addition language "filters" are incorporated, people speaking German will hear virtually nothing from those speaking Spanish etc.

We can continue to add pairs, each speaking unique language (as defined by the unique code) until the overall "background noise" (interference from other users) makes it too difficult for some of the people to understand the other their pair (frame erasure rates get too high). By controlling the voice volume (signal strength) of all users to no more than necessary, we maximize the number of conversation which can take place in the room (maximize the number of user per carrier.)

Therefore, the maximum number of users or effective traffic channels, per carrier depends on the amount of activity that is going on in each channel, and is therefore not precise. It is a "soft overload" concept where an additional user (or conversation in our analogy) can usually be accommodated if necessary, at the "cost" of a bit more interference to the other users.

# 3. Problems

## Multiple Access interference (MAI)

The detector receives a signal composed of the sum of all users' signal, which overlap in time and frequency. Multiple access interference (MAI) refers to the interference between direct sequence users and is a factor which limits the capacity and performance of DS-CDMA systems.

In a conventional DS-CDMA system, a particular user's signal is detected by correlating the entire received signal with that user's signal is detected by correlating the entire received signal with that user's code waveform. The conventional detector does not take into account the existence of MAI. Because of the interference among users, however a better detection strategy is one of multi-user detection. Information about multiple users is used jointly to better detect each individual user.

## Near- far problem

Suppose

- Wireless channel
- Multi-users(transmitters)using the same channel
- One receiver

Each user is a source of interference for the other user, and if one is received with more power, than that user generates more interference for the other user. It is important that the receiver gets the same power from each transmitter. The use of power control ensures that all user arrive at about the same power $P_{rx}$ at the receiver, and therefore no user is unfairly disadvantaged relative to the other. The signal- to – noise interference power ratio at the receiver input for $N_u$ simultaneous users is:

$$SNR = \frac{P_{rx}}{(N_u - 1).P_{rx}} = \frac{1}{(N_u - 1)}$$
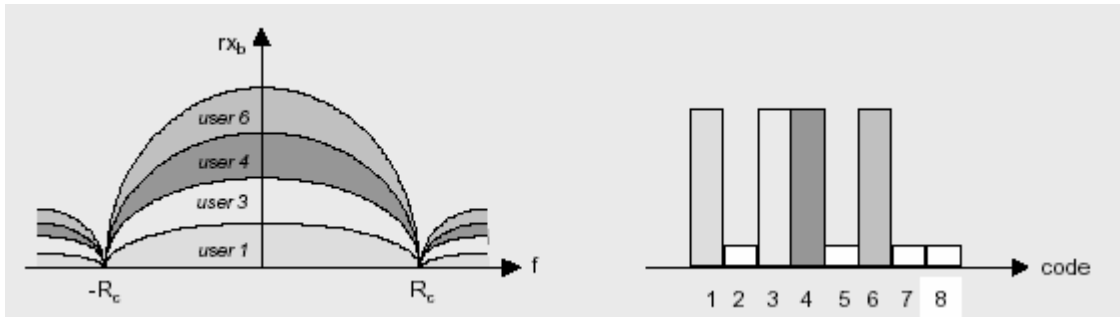
## Multipath Channels (interference)

In wireless channels, there exists often multiple path propagation: there is more than one path from the transmitter to the receiver. Such multipath may be due to:

- Atmospheric reflection or refraction
- Reflection from ground, buildings or other objects

Multipath may result in fluctuations in the received signal level (fading). Each path has its own attenuation and time delay. It is important to keep the direct path and reject the others.

Assume that the receiver is synchronized to the time delay and RF phase of the direct path. The signals at the receiver can be from: the direct path, other paths, white noise, interference.

Suppose two discrete paths: a direct path and only one non-direct path (delayed by a time $\tau$ compared to the direct path).
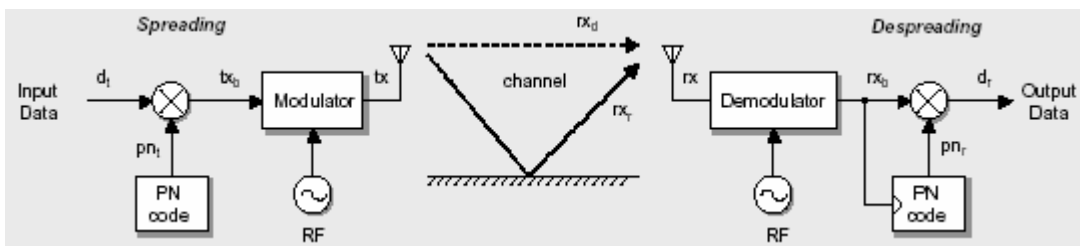


If:

$\tau$ = differential time delay between the two paths $0<\tau<T$

$\theta$ = random angle phase between the carrier of the direct and the non-direct path

$\alpha$ = attenuation of the secondary path

The signal at the receiver can be expressed as:

$$rx = rx_d + rx_r + n = Ad_t(t)\, pn(t)\cos(\omega_0 t) + \alpha Ad_t(t\text{-}\tau)pn(t\text{-}\tau)\cos(\omega_0 t + \theta)+n(t)$$

for the receiver, synchronized to the direct path signal, the output of the correlator, can be written as :

$$d_r(t = N_c T_c) = \int_0^{N_c T_c} pn(t).rx(t)\, dt = \int_0^{N_c T_c} \left[B.pn(t).pn(t) + C.pn(t).pn(t-\tau) + n(t)\right] dt$$

The PN sequence has an autocorrelation function with the property

$$pn(t)\, pn(t) = 1$$
$$pn(t)\, pn(t\text{-}\tau) \neq 1$$

Multipath signals that are delayed by a chip period or longer relative to the desired signal (outdoor reflections) are essentially uncorrelated and do not contribute to multipath fading.

The SS System effectively rejects (mitigation) the multipath interference like in the case of CDMA.

$$d_r(t = N_c T_c) = d_1 + n_0$$

with $n_0$ = noise and multipath interference

The PN code that arrives from the non direct channel(s) is not synchronized to the PN Code of the direct path and is rejected.

## 4. Natural Attributes of CDMA

There are many attributes of CDMA which are of great benefits to the cellular system.

1) **Voice activity cycle:** The real advantage of CDMA is the mature of human conversation. The human voice activity cycle is 35%. The rest of the time we are listening. In CDMA all the users are sharing one radio channel when users assigned to the channel are not talking, all others on the channel benefit with less interference in a single CDMA radio channel. Thus the voice activity cycle reduces mutual interference by 65% increasing the true

channel capacity by three times. CDMA is the only technology that takes the advantage of this phenomenon.

2) **No equalizer needed:** When the transmission rate is much higher than 10kb/s in both FDMA and TDMA, an equalizer is needed for reducing intersymbol interference caused by the time delay spread. However, in CDMA, only a correlator is needed instead of an equalizer at the receiver to despread the SS signal. The correlator is simple than the equalizer.

3) **One radio per site:** Only one radio is needed at each side or at the each sector. It saves equipment space and is easy to install.

4) **No hard handoff:** Since every cell uses the same CDMA radio, the only difference is the code sequences. Therefore no handoff from one frequency while moving from one cell to another cell. It is called the soft handoff.

5) **No guard time in CDMA:** The guard time is required in TDMA between time slots. The guard time does occupy the time period certain bits. Those waste bits could be used to improve quality performance in TDMA. In CDMA, the guard time does not exist.

6) **Sectorization for capacity:** In FDMA and TDMA, the utilization of sectorization in each cell is for reducing the interference. The trunking efficiency of dividing channels in each sector also decreases. In CDMA, the sectorization is used to increase capacity by introducing three radios in three sectors and therefore, three times the capacity is obtained as compared with one radio in a cell in theory.

7) **Less fading:** Less fading is observed in the wide- band signal while propagating in a mobile radio environment. More advantage of using a wide- band signal in urban areas than in suburban areas for fading reduction.

8) **Easy transition:** In a situation where two systems, analog and CDMA, have to share the same allocated spectrum, 10% of bandwidth (1.25MHz) will increase two times (=0.1* 20) of the full bandwidth of FM radio capacity as shown below. Since only 5%( heavy users) of the total users take more than

30% of the total traffic, the system providers can let the heavy user exchange their analog units for dual mode (analog / CDMA) units and converts 30% of capacity to CDMA on the first day of CDMA operations.

9) **Capacity advantage** : Given that

$$B_t = 1.25 \text{ Mhz, the total bandwidth}$$
$$B_{ss} = 1.25 \text{ MHz, the CDMA radio channel}$$
$$B_c = 30 \text{ khz for FM}$$
$$B_c = 30 \text{ khz and three times slots for TDMA}$$

Capacity of FM 1.25/30 = 41.6

Total numbers of channel = $1.25 * 10^6/ 30* 10^3 = 41.67$ channels

The cell reuse pattern K= 7

The radio capacity $m_{FH} = 41.67 / 7 = 6$ channel/ cell

Capacity of TDMA

Total number of channels = $1.25 * 10^6 / 10 * 10^3 = 125$ channels

The cell reuse pattern K= 4 (assumed)

The radio capacity $m_{TDMA} = 125 / 4 = 31.25$ CHANNELS/ CELL

Capacity of CDMA

Total number of channels per cell, m = 13

The cell reuse pattern K = 1.33

The radio capacity, take at $E_b / I = 7$ Db

Add voice activity cycle and sectorization,

$m_{CDMA} = 13 * 3*3 \approx 120$ channels / cell

therefore ,

$$m_{CDMA} = 20 * m_{FM} = 4* m_{TDMA}$$

10) **No frequency management or assignment needed:** In FDMA and TDMA, the frequency is always a critical task to carry out. Since there is only one common radio channel in CDMA, no frequency management is needed. Also, the dynamic frequency would implement in TDMA and FDMA to reduce real time interference, but needs a linear broadband power

amplifier, which is hard to develop. CDMA does not need the dynamic frequency assignment.

11) **Soft capacity:** In CDMA, the entire traffic channels shares one CDMA radio channel. Therefore, we can add one additional user so the voice quality is just slightly degraded as compared to that of the normal 40-channel cell.

12) **Coexistence:** Both systems, analog and CDMA can operate in two different spectras, and CDMA only needs 10% of bandwidth to general 200% of capacity. No interference would be considered between two systems.

13) **For microcell and in-building systems:** CDMA is a natural waveform suitable for microcell and in building because of being susceptible to the noise and the interference.

# 5. Performance & Evaluation

## Advantages

- Selective addressing is accomplished by transmitting the proper signature sequence as modulation.
- Code modulation generates a wideband signal spectrum with low power density. in any narrow region. This helps in signal hiding.
- CDMA enables a large number of asynchronous users to transmit information.
- The thumbtack shape of the auto correlation of the signature sequence and the low cross correlation between different signature sequences reduce the interference from unwanted signals in the network.
- It is possible to implement both the encoder and the decoder using optical fibers thereby enabling the use of much higher frequency.
- Capacity increases of 8 to 10 times of an AMPS analog system.
- Improved call quality, with better and more consistent sound as compared to AMPS system.

- Simplified system planning through the use of the same frequency in every sector of ever cell.
- Enhanced privacy.
- Improved coverage characteristics, allowing for the possibility of fewer cell sites.
- Increased talk time for portables.
- Bandwidth on demand.

## Disadvantages

- number of the codes are small
- Spread frequency increases significantly with increases significantly with increase in number of users (e.g. 25 users @ 1Mbps/user → 1024 Mbps)
- High speed digital circuits consume large amount of current and power of current and power dissipation is high.]

## CDMA Channels

Just when one grasp on understanding of the CDMA carrier which is 1.25 MHz wide, someone talks about "traffic channels" and confuses the issue. The fact is that with CDMA, the path by which voice or data passes is the entire carrier, as described previously.

CDMA traffic channels are different they are dependent on the equipment platform, such as Motorola's SC™ products, on which the CDMA is implemented. Motorola designates channels in three ways: Effective traffic channels, actual traffic channels and physical traffic channels.

- The number of "Effective" traffic channels includes the traffic carrying channels less the soft handoff channels. The capacity of an effective traffic channel is equivalent to the traffic carrying capacity of an analog traffic channel.
- The number of "Actual" traffic channels includes the effective traffic channels, plus channels allocated for soft handoff.

- The number of "Physical" traffic channels includes the Pilot channels, the Sync channels, the Paging channels, the soft handoff Overhead channels and the effective (voice and data) traffic channels.

CDMA uses the terms "forward" and "reverse" channels just like they are used in analog systems. Base transmit equates to the forward direction and base receive is the reverse direction. ("Forward" is what the subscriber hears and "reverse" is what the subscriber speaks.)

## CDMA forward Channels

### Pilot Channel

The pilot channel is used by the mobile unit to obtain initial system synchronization and to provide time, frequency and phase tracking of signals from the cell site.

### Syn Channel

This channel provides cell site identification, pilot transmit power, and the cell site pilot pseudo-random (PN) phase offset information. With this information the mobile units can establish the system time as well as transmit proper power level to use to initiate a call.

### Paging Channel

The mobile unit will begin monitoring the paging channel after it has set its timing to the System Time provided by the sync channel. Once a mobile unit has been paged and acknowledges that page, call setup and traffic channel assignment information is then passed on this channel to the mobile unit.

### Forward traffic Channel

This channel carries the actual phone call and carries the voice and mobile power control information from the base station to the mobile unit.

## CDMA reverse Channels

### Access Channel

When the mobile unit is not active on a traffic channel, it will communicate to the base station over the access channel. This communication includes registration requests, responses to pages, and call originations. The access channels are paired with a corresponding paging channel.

### Reverse Traffic Channel

This channel carries the other half of the actual phone call and carried the voice and mobile power control information from the mobile unit to the base station.

### CDMA Modulation

Both the forward and reverse traffic channels use a similar control structure consisting of 20 millisecond frames. For the system, frames can be sent at either 14400, 9600, 7200, 4800, 3600, 2400, 1800 or 1200 bps.

For example, with a traffic Channel operating at 9600 bps, the rate can vary from frame to frame, and can be 9600, 4800, 2400 or 1200 bps. The receiver detects the rate of the frame and processes it at the correct rate. This technique allows the channel rate to dynamically adapt to the speech or data activity. For speech, when talker pauses, the transmission rate is reduced to low rate. When the talker speaks, the system instantaneously shifts to using a higher transmission rate. This technique decreases the interference to other CDMA signals and thus allows an increase in system capacity.

CDMA start with a basic data rate of 9600 bits per second. This is then spread to a transmitted bit rate, or chip rate (the transmitted bits are called chips), of 1.2288 MHz. The spreading process applies digital codes to the data bits, which increases the data rate while adding redundancy to the system.

The chips are transmitted using a form of QPSK (quadrature phase shift keying) modulation which has been filtered to limit the bandwidth of the signal. This is added to the signal of all the other users in that cell when the signal is received, the coding is removed from the desired signal, returning it to rate of 9600 bps. When decoding is applied to the other user's code, there is no dispreading; the signals maintain the 1.2288 MHz bandwidth. The ratio of transmitted bits or chip to data bits is the coding gain. The coding gain for the IS-95 CDMA system is 128 or 21 dB.

## CDMA advanced Features

### Advanced Features Overview

As was mentioned earlier, the CDMA development Group continues to design feature and capabilities for CDMA cellular systems in order to provide better service to subscriber and new revenue generating services for the operators. The current set of advanced feature which have been defined by the CDG and TIA and are being incorporated into systems include:

- Multiple, High Quality Vocoders
- CDMA short Message Services (SMS)
- Over the air activation
- Sleep mode
- CDMA Data and Fax

Apart from the CDG, Motorola is implementing features that ease subscriber migration from analog to CDMA.

- Simultaneous Subscriber Unit Ringing ("Extension Phone")

- Subscriber Access Control (SAC) (SAC allows an operator to selectively control mobile access to the CDMA system 011, a per cell/sector basis.) CDMA for Cellular

When implemented in a cellular telephone system, CDMA technology offers numerous benefits to the cellular operator and their Subscribers. These can be summarized as follows:

- Capacity increases: 8 to 10 times that of an AMPS analog system, and 4 to 5 times that of a GSM system.

- Improved call quality: CDMA will provide better and more consistent sound as compared to AMPS. Cellular telephone systems using CDMA should be able to provide higher quality sound and phone calls than systems based on other technologies.

- Simplified system planning: Engineers, will no longer have to perform the detailed frequency planning which is necessary in analog and TDMA systems.

- Enhanced Privacy: increased privacy over other cellular systems, both analog and digital, is inherent in CDMA technology.

- Increased talk time and standby time for portables: Because of precise power control and other system characteristics, CDMA subscriber units normally transmit at only a fraction of the power of analog and TDMA phones.

- Advanced Features: These illc1ude Multiple/High Quality Vocoders, Short Messaging Services, Over-the-Air-Activation, Sleep Mode, and Data/Fax,


## CDMA for Personal Communications Services (PCS)

Personal Communications Services (PCS) is anticipated to be the "omnipresent" wireless communication system of the future - a vehicle through which people can communicate to whomever they want, whenever they want, wherever they want. CDMA is a strong solution for PCS because it offers higher capacity and increased

range of coverage. By requiring fewer cell sites than traditional analog and other digital systems, CDMA is viewed as the most cost effective technology for PCS. From enhanced spectral efficiency to improved call quality and feature flexibility, CDMA offers a host of key benefits *to* wireless PCS operators. Specifically, benefits which are especially relevant to PCS operators include:

- Increased Capacity: In the long run, carriers who provide greater capacity from the same spectrum will experience lower operating expenses than operators who choose less spectrum-efficient technologies.

- Audio Quality: CDMA has a number of unique features which provide a superior, robust RF communications path to digital communications. The qualities that allow CDMA to provide high quality voice performance also ensure superior dropped call performance over existing cellular systems.

- Vocoder Flexibility: Different vocoders can be used for product and service differentiation.

- Decreased Blocking: System capacity in CDMA is flexible and has a "soft" limit. As a result, blocking is decreased since capacity can grow in response to user demand.

- Enhanced Coverage: CDMA reduces the number of cells needed for coverage compared to other air interfaces, while preserving robust audio quality.

- Low Deployment Cost: With a large advantage in link margin (approximately 5 dB), a CDMA system requires approximately half the number of cell sites for the same coverage area and system loading. This efficiency lowers equipment and site acquisition costs, and is especially cost-effective for new system deployment.

### CDMA for Wireless Local Loop (WiLL)

In many developing countries, there is tremendous demand for new business and residential telephone service. More and more operators are looking to wireless Technologies to rapidly provide thousands of new Subscribers with high quality telephone service at a reasonable price. Motorola has introduced a set of wireless

Local Loop (WiLL) systems which can meet the telephony requirements of many communities.

- Existing landline operators can extend their network with WiLL.
- Cellular operators can capitalize on their current network to deliver residential service with WiLL.
- New service providers can quickly deploy non-traditional WiLL solutions to rapidly meet a community's telephony needs.

The unique features and benefits of COMA make it an excellent technology choice for fixed wireless telephone systems.

In a fixed telephony environment, COMA is estimated to provide up to 15-20 times the capacity of an AMPS cellular system, resulting in the highest capacity cellular based offering for wireless local loop applications. The key to this increased WiLL capacity is that, in a fixed environment, CDMA power control is able to very accurately track required power, thereby resulting in reduced overall transmission power and increased capacity. Subscribers in a fixed environment require less RF power to achieve quality communications, therefore more subscribers can be placed on a CDMA channel. In addition, CDMA optimizes use of the radio spectrum, which is an increasingly scarce resource worldwide. CDMA's single cell frequency re-use capability and non-contiguous bandwidth requirement, along with its extended coverage range, simplifies RF planning and implementation. This allows an operator to invest in fewer cell sites with faster deployment, ultimately giving the service provider increased and quicker access to revenues.

The CDMA digital interface also allows for a number of complex and unique features such as variable rate vocoders; robust error correction; frequency, space and time diversity; and multipath immunity. These features all contribute to improved system and call quality.

# What Is MATLAB?

MATLAB® is a high-performance language for technical computing. It integrates computation, visualization, and programming in an easy-to-use environment where problems and solutions are expressed in familiar mathematical notation. Typical uses include:

- Math and computation
- Algorithm development
- Modeling, simulation, and prototyping
- Data analysis, exploration, and visualization
- Scientific and engineering graphics
- Application development, including graphical user interface building

MATLAB is an interactive system whose basic data element is an array that does not require dimensioning. This allows solving many technical computing problems, especially those with matrix and vector formulations, in a fraction of the time it would take to write a program in a scalar noninteractive language such as C or FORTRAN.

The name MATLAB stands for *matrix laboratory*. MATLAB was originally written to provide easy access to matrix software developed by the LINPACK and EISPACK projects. In university environments, it is the standard instructional tool for introductory and advanced courses in mathematics, engineering, and science. In industry, MATLAB is the tool of choice for high-productivity research, development, and  analysis.

MATLAB features a family of application-specific solutions called *toolboxes*. Very important to most users of MATLAB, toolboxes allow you to *learn* and *apply* specialized technology. Toolboxes are comprehensive collections of MATLAB functions (M-files) that extend the MATLAB environment to solve particular classes of problems. Areas in which toolboxes are available include signal processing, control systems, neural networks, fuzzy logic, wavelets, simulation, and many others.

## The MATLAB System

The MATLAB system consists of five main parts:

**Development Environment:** This is the set of tools and facilities that help you use MATLAB functions and files. Many of these tools are graphical user interfaces. It includes the MATLAB desktop and Command Window, a command history, and browsers for viewing help, the workspace, files, and the search path.

**The MATLAB Mathematical Function Library.** This is a vast collection of computational algorithms ranging from elementary functions like sum, sine, cosine, and complex arithmetic, to more sophisticated functions like matrix inverse, matrix eigenvalues, Bessel functions, and fast fourier transforms.

**The MATLAB Language.** This is a high-level matrix/array language with control flow statements, functions, data structures, input/output, and object-oriented programming features. It allows both "programming in the small" to rapidly create quick and dirty throw-away programs, and "programming in the large" to create complete large and complex application programs.
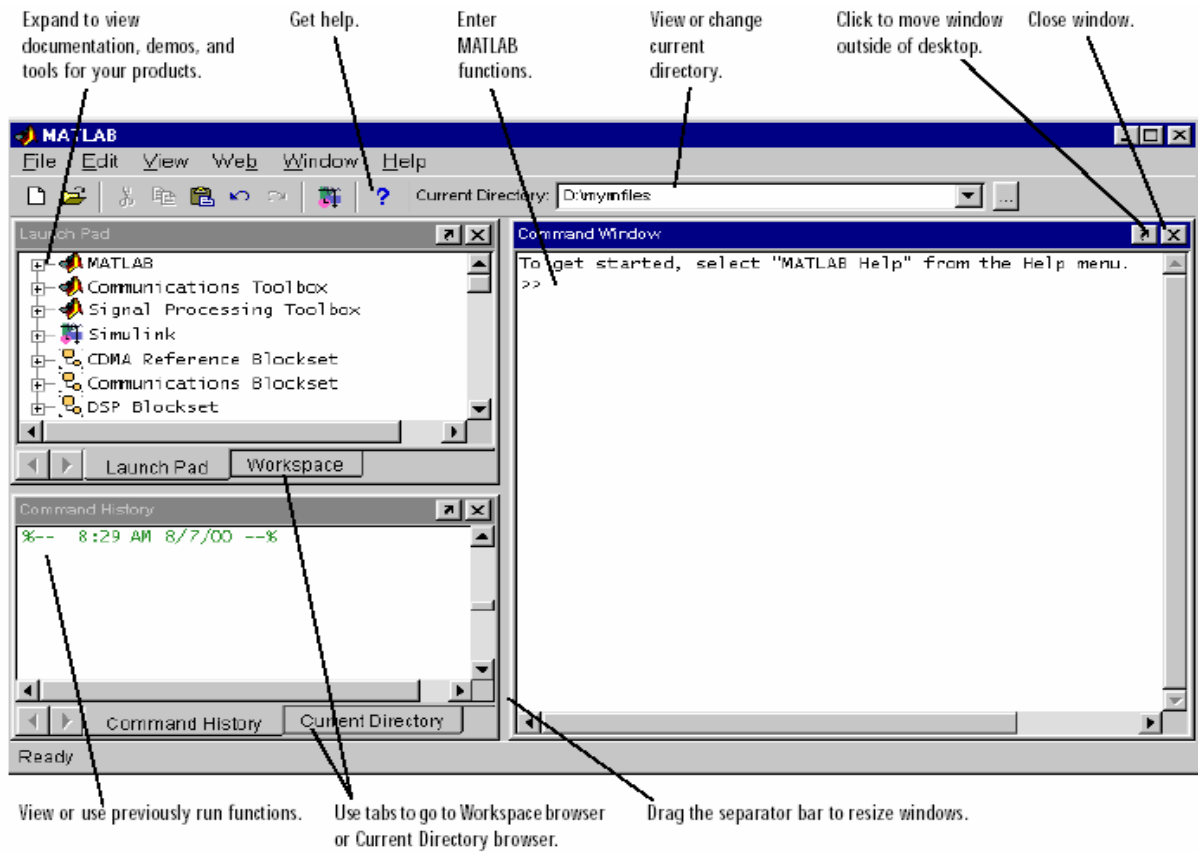
**Handle Graphics®.** This is the MATLAB graphics system. It includes high-level commands for two-dimensional and three-dimensional data visualization, image processing, animation, and presentation graphics. It also includes low-level commands that allow to fully customize the appearance of graphics as well as to build complete graphical user interfaces on MATLAB applications.

**The MATLAB Application Program Interface (API).** This is a library that allows you to write C and FORTRAN programs that interact with MATLAB. It include facilities for calling routines from MATLAB (dynamic linking), calling MATLAB as a computational engine, and for reading and writing MAT-files.
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# MATLAB Desktop

The MATLAB desktop appears, containing tools (graphical user interfaces) for managing files, variables, and applications associated with MATLAB. The first time MATLAB starts, the desktop appears as shown in the following illustration.

Expand to view documentation, demos, and tools for your products.    Get help.    Enter MATLAB functions.    View or change current directory.    Click to move window outside of desktop.    Close window.

MATLAB
File   Edit   View   Web   Window   Help
Current Directory: D:\mymfiles

Launch Pad
- MATLAB
- Communications Toolbox
- Signal Processing Toolbox
- Simulink
- CDMA Reference Blockset
- Communications Blockset
- DSP Blockset

Launch Pad    Workspace

Command Window
To get started, select "MATLAB Help" from the Help menu.
>>

Command History
%-- 8:29 AM 8/7/00 --%

Command History    Current Directory

Ready

View or use previously run functions.    Use tabs to go to Workspace browser or Current Directory browser.    Drag the separator bar to resize windows.

**Command Window:** The Command Window is the place to enter variables and run functions and M-files.

**Command History:** Lines you enter in the Command Window are logged in the Command History window. In the Command History, you can view previously used functions, and copy and execute selected lines.

**Launch Pad:** MATLAB's Launch Pad provides easy access to tools, demos, and documentation.

**Current Directory Browser:** MATLAB file operations use the current directory and the search path as reference points. Any file you want to run must either be in the current directory or on the search path.

**Workspace Browser:** The MATLAB workspace consists of the set of variables (named arrays) built up during a MATLAB session and stored in memory. You add variables to the workspace by using functions, running M-files, and loading saved workspaces.

**Editor/Debugger:** Use the Editor/Debugger to create and debug M-files, which are programs you write to run MATLAB functions. The Editor/Debugger provides a graphical user interface for basic text editing, as well as for M-file debugging.
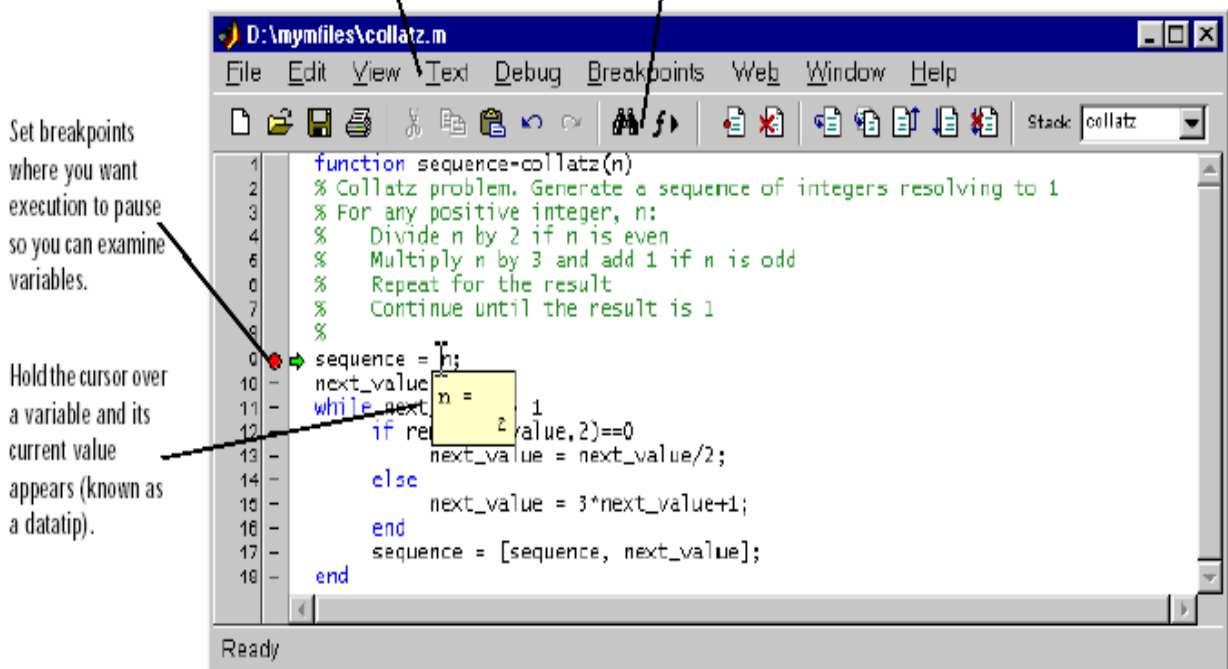


# Working with MATLAB

### Manipulating Matrices

In MATLAB, a matrix is a rectangular array of numbers. Special meaning is sometimes attached to 1-by-1matrices, which are scalars, and to matrices with only one row or column, which are vectors. MATLAB has other ways of storing

both numeric and nonnumeric data, but in the beginning, it is usually best to think of everything as a matrix. The operations in MATLAB are designed to be as natural as possible.

Matrices may be entered into MATLAB in several different ways:

- Enter an explicit list of elements.
- Load matrices from external data files.
- Generate matrices using built-in functions.
- Create matrices with your own functions in M-files.

### Expressions

Like most other programming languages, MATLAB provides mathematical *expressions*, but unlike most programming languages, these expressions involve entire matrices. The building blocks of expressions are:

- Variables
- Numbers
- Operators
- Functions

### Functions

MATLAB provides a large number of standard elementary mathematical functions, including abs, sqrt, exp, and sin. Taking the square root or logarithm of a negative number is not an error; the appropriate complex result is produced automatically. MATLAB also provides many more advanced mathematical functions, including Bessel and gamma functions. Most of these functions accept complex arguments.

### M-Files

We can create your own matrices using *M-files*, which are text files containing MATLAB code. Use the MATLAB Editor or another text editor to create a file containing the same statements you would type at the MATLAB command line. Save the file under a name that ends in .m. There are two kinds of M-files:

- Scripts, which do not accept input arguments or return output arguments. They operate on data in the workspace.
- Functions, which can accept input arguments and return output arguments. Internal variables are local to the function.

## Flow Control

MATLAB has several flow control constructs:

- if statements
- switch statements
- for loops
- while loops
- continue statements
- break statements

## Reference Blockset (toolbox)

The Reference Block sets are collection of Simulink® blocks designed to help to develop and simulate various systems, incorporating the prominent international standards. About 40 such block sets (toolbox) are available, like

- **Communication blockset:** Simulink block libraries for modeling the physical layer of communications systems
- **Communication toolbox:** MATLAB functions for modeling the physical layer of communications systems
- **DSP blockset:** Simulink block libraries for the design, simulation, and prototyping of digital signal processing systems
- **Signal processing toolbox:** Tool for algorithm development, signal and linear system analysis, and time-series data modeling
- **Simulink:** Interactive, graphical environment for modeling, simulating, and prototyping dynamic systems
- **CDMA reference blockset:** With the CDMA Reference Blockset block diagram models of wireless systems quickly and easily constructed using

click-and-drag mouse operations. And the simulations on those models could be run along with change of parameters as needed. The blocks in the CDMA Reference Blockset encompass the complete functionality required by the IS-95A standard. The CDMA Reference Blockset to develop end-to-end (transmitter-to-receiver) simulation models for forward and reverse communication links. Hence, the simulation stage could be easily achieved with focus on the specific components of greatest interest in the design.

# Chapter 4. Simulation

## 4.1 Simulation Parameters

This dissertation presents the performance evaluation of a multi-user CDMA system in terms of Probability of error (Pe) v/s SNR for wide variety of vital system parameters & components.

As an alternative to traditional approach a simulation has been envisaged in Matlab which is a powerful technical computing tool. The variable parameters are:

- Number of chips(Length of the signature sequence)
- Number of active users(extent of MAI)
- Synchronized/ non-synchronized transmitter & receiver.

The system has been tested for an optimal combination of the following components:

- Error detecting/correcting codes-Cyclic codes, convolution codes, hamming & linear block codes.
- Spreading codes- PN Sequences, Gold Codes, and Hadamard-Walsh codes (fully or partially orthogonal).
- QPSK modulation.
- Channel- AWGN, Rayleigh(multipath) fading & both.
- Diversity Combination- Selection Diversity, EGC, MGC.
- Receiver – integrate & dump followed by a correlation detector.
- Pulse shaping- rectangular.

Graphs have been plotted between Pe & SNR as a measure of relative performance appraisal. The results for the simulation as included in this report are derived for the following cases:

### Case A: Variable Chip- Length (1, 3, 7, 15, 31 chips)
- Synchronized Transmitter & Receiver.
- Multiple users- equal to 5.
- Hamming Codes.
- PN Sequences.
- AWGN with Rayleigh Fading Channel.
- Diversity Combination – EGC.

For this case all the suggestive graphs of the signals after each section of the system have been drawn to showcase the processing.

## Case B: Variable Chip- Length (1, 3, 7, 15, 31 chips).

- Asynchronized Transmitter & Receiver.
- Single user.
- Convolution Codes
- Hadamard-Walsh Sequences.
- Rayleigh Fading Channel.
- Diversity Combination – Space Diversity.

## Case C: Synchronous / Asynchronous System.

- Multiple users- equal to 4.
- Cyclic Codes.
- Gold Sequences.
- Rayleigh Fading Channel.
- Diversity Combination – MGC.
- Number of Chips – equal to 15.

## Case D: Single/ Multiple Active users.

- Asynchronized Transmitter & Receiver.
- Convolution Codes.
- Hadamard-Walsh Sequences.
- AWGN with Rayleigh Fading Channel.
- Diversity Combination – Space Diversity.
- Number of Chips – equal to 16.

## Case E: Error correcting/Detecting Codes- Hamming, Linear Block Codes, Cyclic & Convolution Codes.

- Synchronized Transmitter & Receiver.
- PN Sequences.
- AWGN with Rayleigh Fading Channel.
- Diversity Combination – MGC.
- Number of Chips – equal to 31.
- Multiple users – equal to 4.

## Case F: Signature Sequence – PN, Gold, Hadamard-Walsh Sequence.

- Synchronized Transmitter & Receiver.
- Cyclic Codes.
- AWGN with Rayleigh Fading Channel.
- Diversity Combination – MGC.
- Number of Chips – equal to 31.
- Multiple users – equal to 4.

**Case G: Channel – AWGN Channel, Rayleigh Fading Channel & AWGN with Rayleigh Fading.**

- Synchronized Transmitter & Receiver.
- Linear Block Codes.
- Gold Sequences.
- Diversity Combination – EGC.
- Number of Chips – equal to 7.
- Single user.

**Case H: Diversity Combination – Space Diversity, Equal Gain Combination & Marginal Gain Combination.**

- Asynchronized Transmitter & Receiver.
- Convolution Codes.
- Hadamard-Walsh Sequence.
- Number of Chips – equal to 16.
- Multiple users – equal to 5.
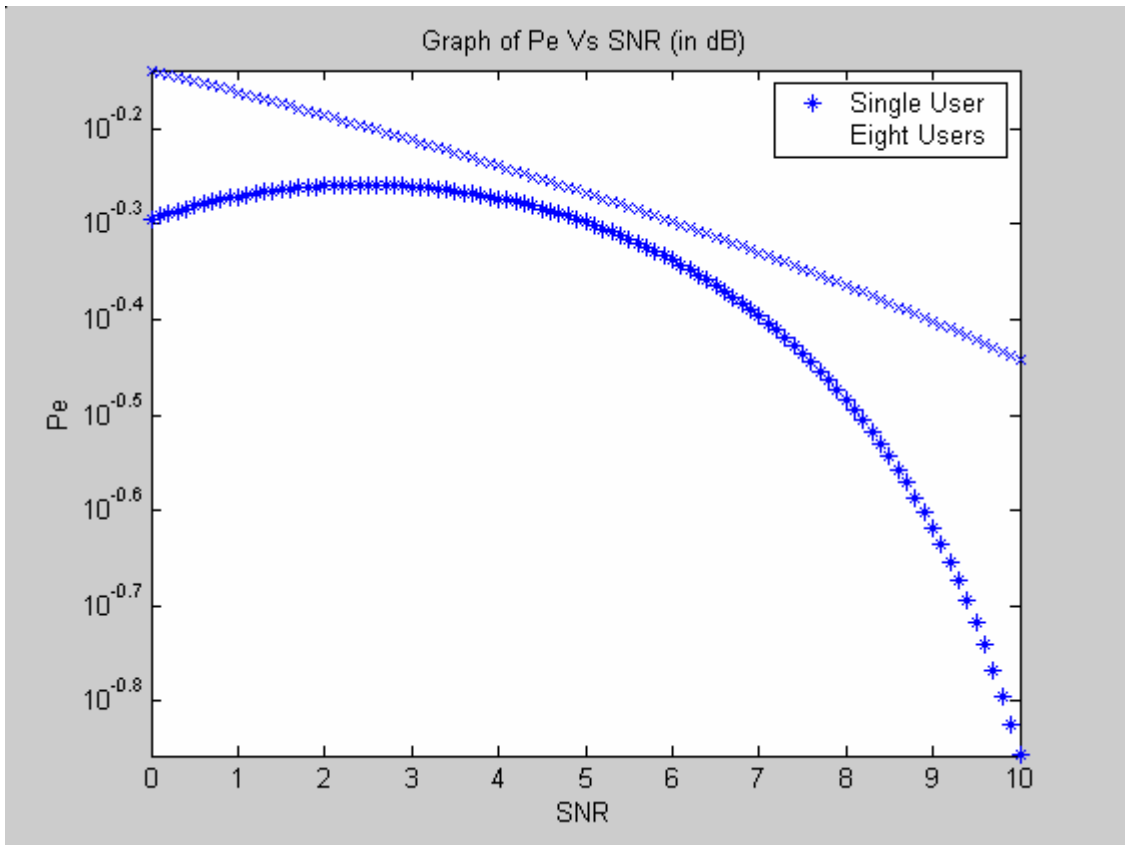
## 4.2 Simulation Graphs & Results

Assumptions
· Fading is amplitude only.
· Fading is constant over a bit-interval.
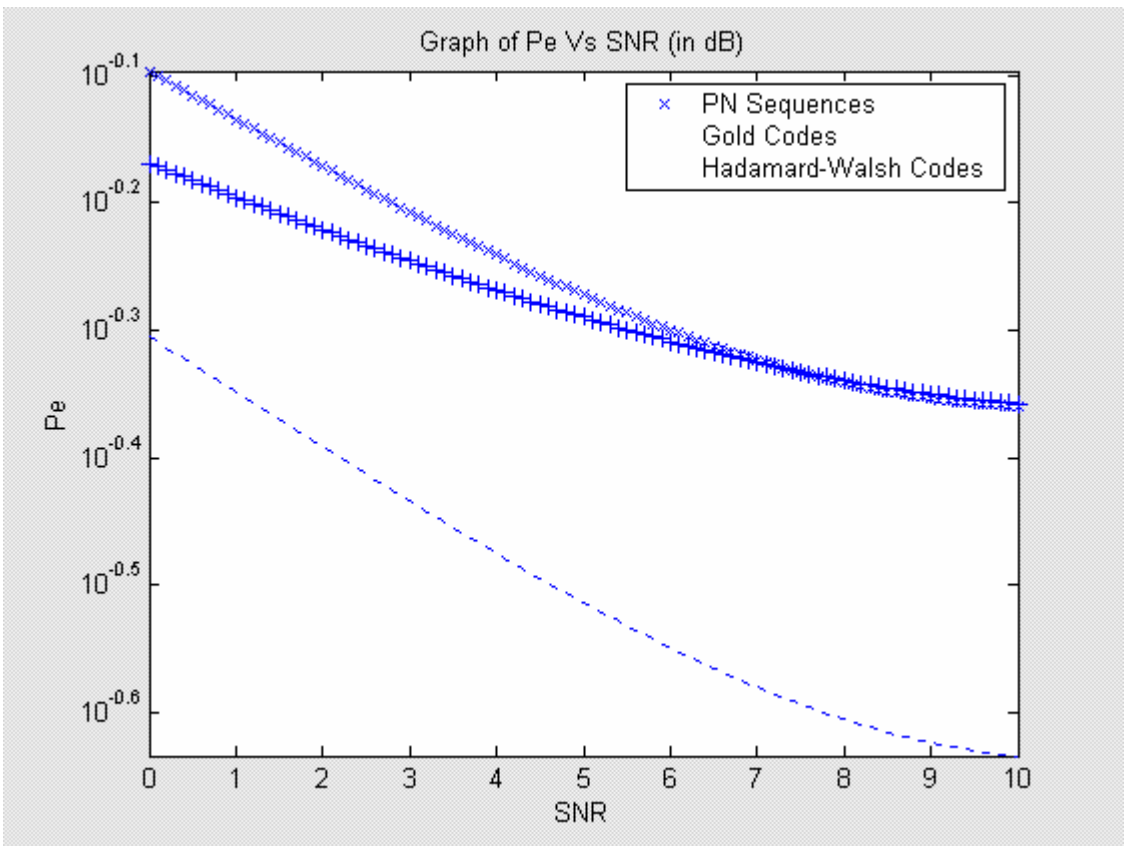· Fading is independent from one bit interval to the next bit interval.

Pe Results - Fading Channel. Using the above mentioned parameters and assumptions, Monte-Carlo simulations were performed in both an AWGN channel and on a fading channel. The results are plotted in fig (5). The results obtain conform with the theoritical values for the Pe given by equations, (12) and (14) and also illustrates through simulation, the destructive effect of fading on the BER performance. For example, even for a BER of 0.02, the increase in the power requirement shoots up to 6.5dB and much worse for lesser BER requirements.
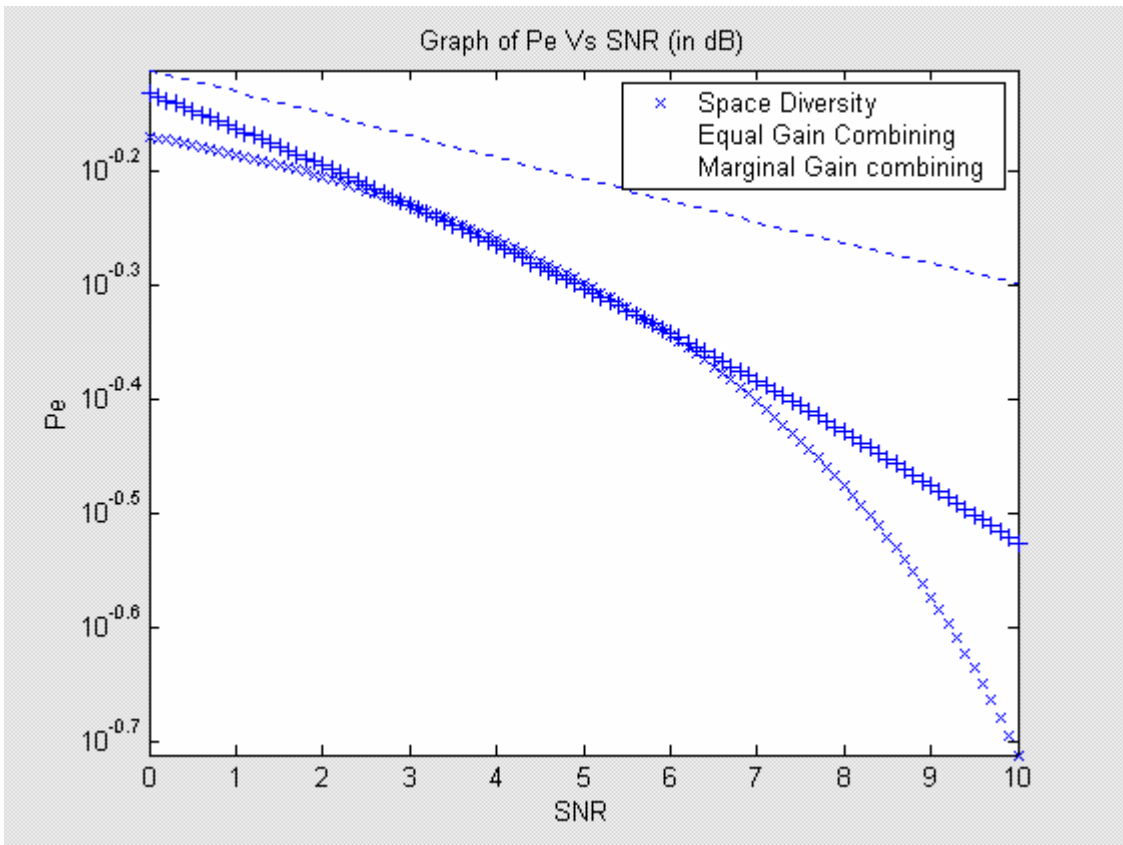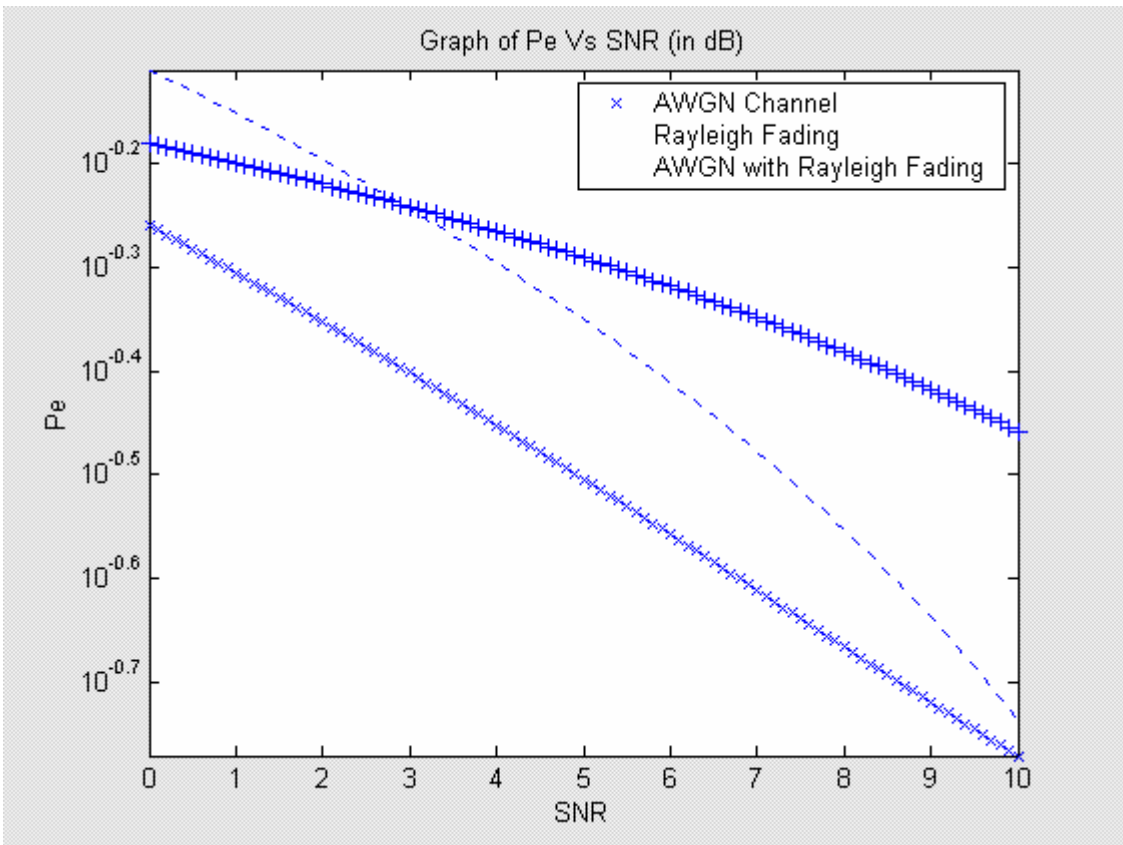
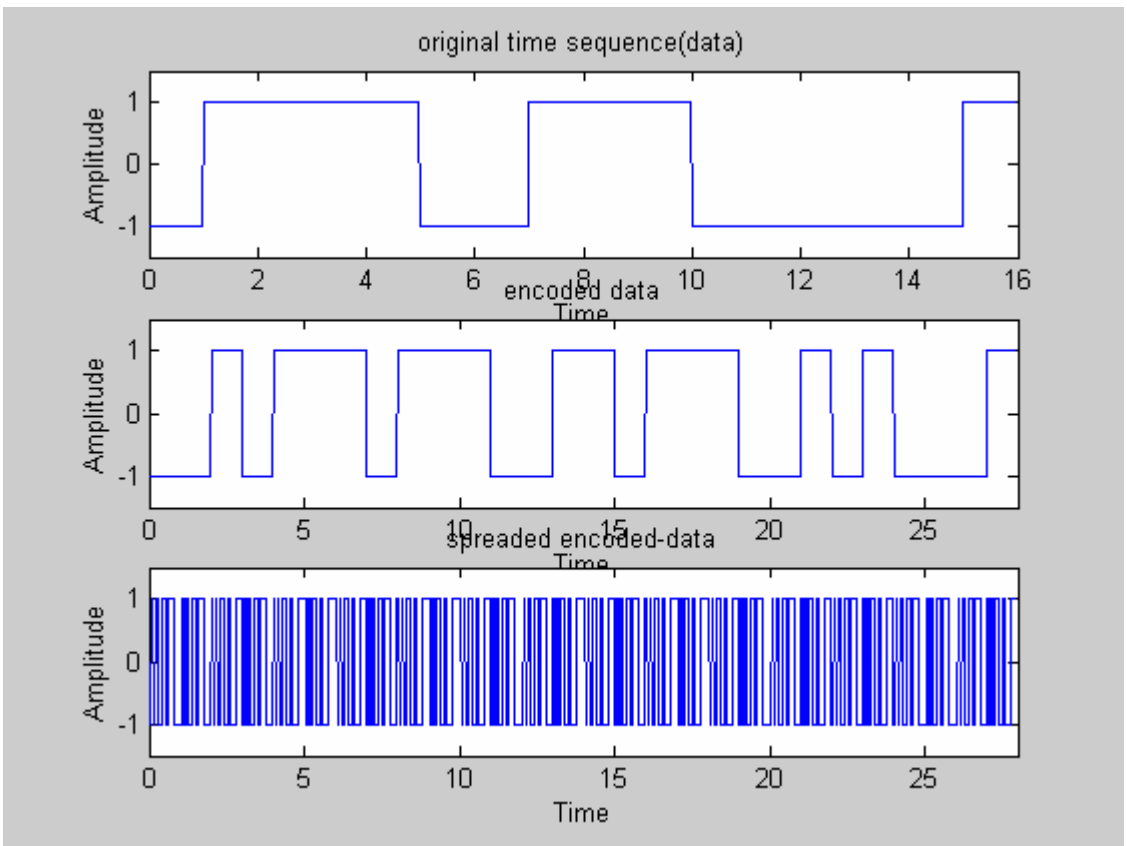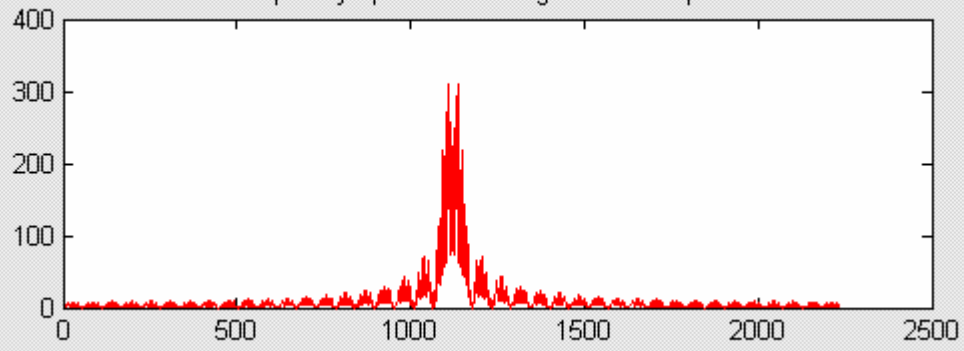· There is an excessive degradation in the BER performance due to fading.

Graph of Pe Vs SNR (in dB)



Graph of Pe Vs SNR (in dB)

Graph of Pe Vs SNR (in dB)

Graph of Pe Vs SNR (in dB)



Graph of Pe Vs SNR (in dB)

Graph of Pe Vs SNR (in dB)

Graph of Pe Vs SNR (in dB)



Graph of Pe Vs SNR (in dB)

original time sequence(data)

encoded data

spreaded encoded-data

Frequency spectrum of Orignal data sequence

Frequency spectrum of spreaded data sequence

I- channel (odd) clock

Q- channel (even) clock

I- channel data

Q- channel data

plot for I-channel modulated output

plot for Q-channel modulated output

QPSK modulated waveform of data

Frequency spectrum of QPSK modulated output

Data recieved at the reciever with AWGN noise

demodulated I- channel data

demodulated Q-channel data

demodulated but spreaded data

# Chapter 5.Conclusion & Future Scope

Monte-Carlo Simulation approach has been used in the present work to evaluate the probability of error of the desired user as per the SNR of the received signal. The performance optimization & relative merit of the usage of various system techniques has been broached.

The Pe versus SNR Graphs has been plotted for a multitude of the cases. The results have been analyzed in the preceding chapter.

The highly suggestive Matlab code utilizes the immense flexibility & computing ease the Matlab offers. It has been included in the appendix.

## Conclusions

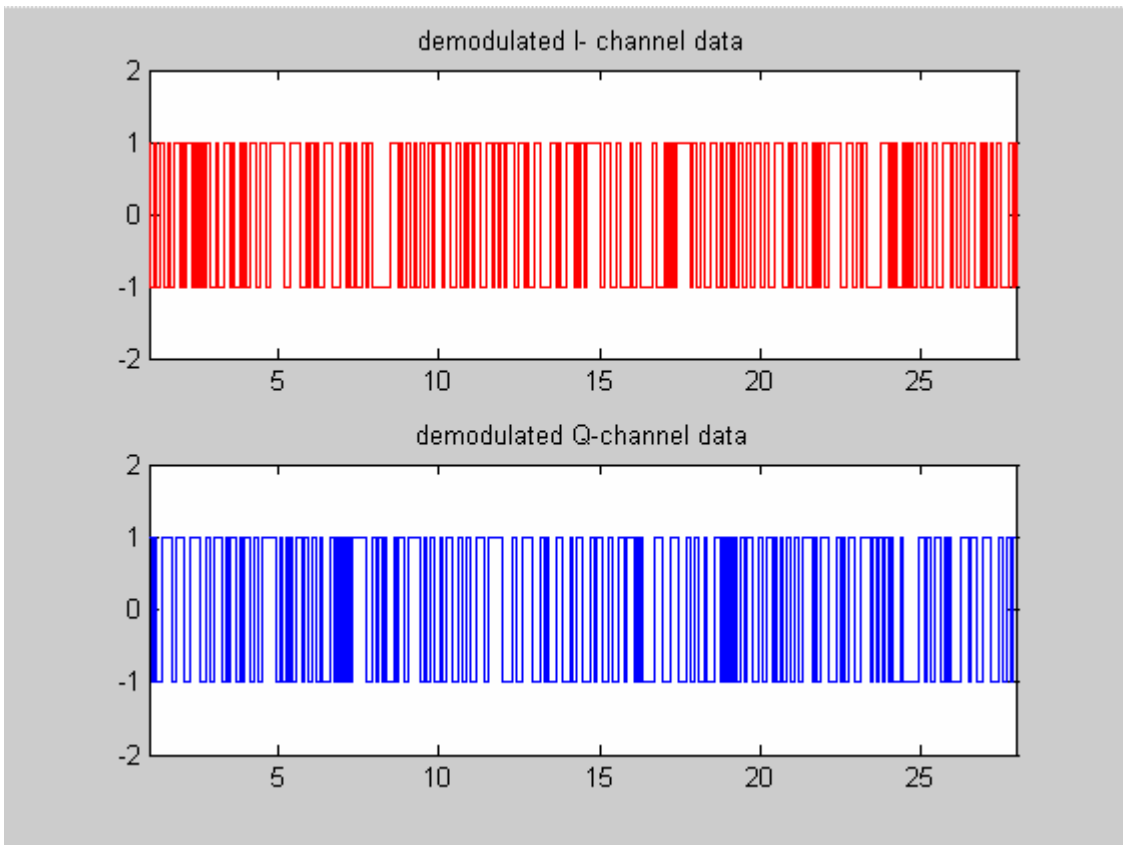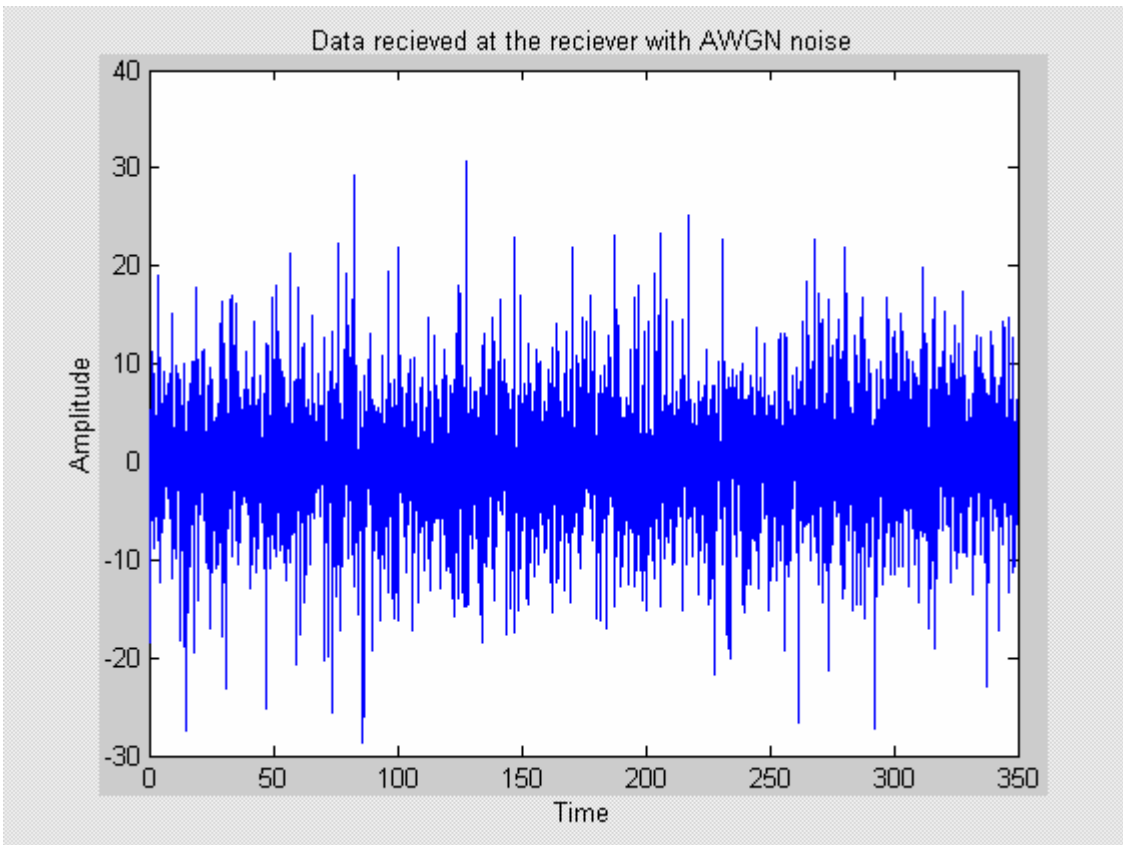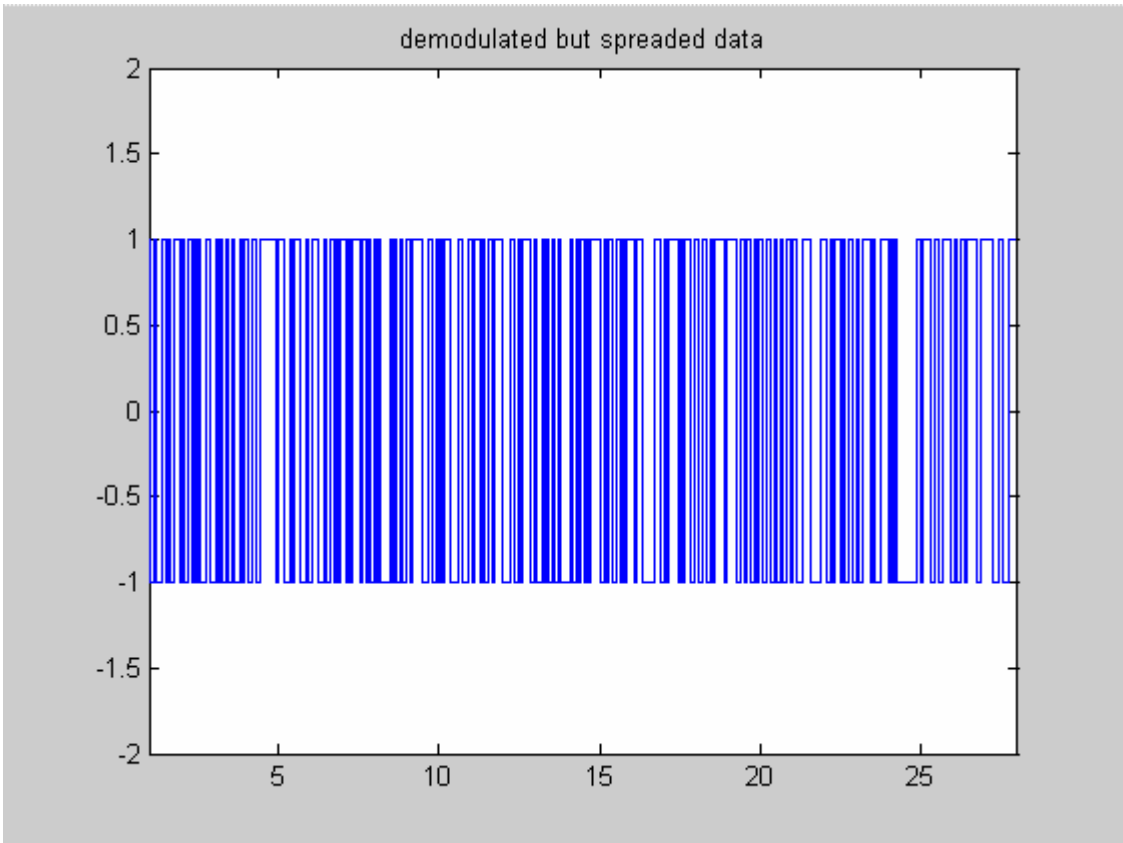♦ Monte-Carlo methods have provided reliable & consistent results obtained for the number of variations considered.

♦ The simulation of Multi-User CDMA System model Pe versus SNR has been done successfully.

♦ The number of chips has considerable effect on the system probability of error. As number of chips increases, probability of error decreases.

♦ As the level of Multiple-Access Interference increases the performance of the system decreases.

♦ Synchronization of the desired user also leaves significant impact on the quality of reception of desired user. In this case system returns better performance than the non-synchronized system.

♦ Rayleigh or multipath fading of a desired user's signal seriously degrades the quality of communication. The desired user with just single path in the channel shows better performance.

♦ Orthogonality of the signature sequence also leaves reasonable effect on BER. Best case is of Hadamard-Walsh codes in this regard. The system performance decreases as the orthogonality of the spreading code decreases.
♦ Error detecting/correcting codes are of immense help in improving the quality of transmission, the best being cyclic codes.
♦ Diversity techniques can effectively combat fading. The performance improvement in the BER is maximum for Maximal Ratio Combining (MRC), while Equal Gain Combining (EGC) and Selective Combining provide marginally inferior performances.

## Scope for future work

After completion of the work presented in this dissertation, the following areas have been identified and recommended for future work.

♦ In this simulation, rectangular pulses have been taken for signature sequence. Performance can be optimized testing other pulse shapes.
♦ Performance of CDMA system can be analyzed in terms of the signal-to-interference ratio versus signal-to-noise ratio by varying the signal power (Power Control) of the interfering users.
♦ The CDMA channel capacity bound optimization can be achieved via simulation.
♦ Adaptive CDMA can be analyzed for relative difference between Adaptive/non-adaptive CDMA.
♦ M-Ary PSK modulation scheme can be utilized for the data transmission.
♦ Bit interleavers (as used in IS 95) can be incorporated.

]

## Appendix:

### 1. Future of CDMA technology

Though CDMA's application in cellular telephony is relatively new, it is not a new technology. CDMA has been used in many military applications, such as anti-jamming (because of the spread signal, it is difficult to jam or interfere with a CDMA signal), ranging (measuring the distance of the transmission to know when it will be received), and secure communication (the spread spectrum signal is very hard to detect).

The standard data rate of a CDMA call is 9600 bits per second (9.6 kilobits per second) This initial data is spread, including the application of digital codes to the data bits, up to the transmitted rate of about 1.23 megabits per second. The data bits of each call are then transmitted in combination with the date bits of all of the calls in the cell. At the receiving end, the digital codes are separated out, leaving only the original information which was to be communicated. At that point, each call is once again is unique data stream with a rate of 9600 bits per second.

Traditional uses of spread spectrum are in military operations. Because of the wide bandwidth of spread spectrum signal, it is very difficult to jam, difficult to interfere with, and difficult to identify. This is in contrast to technologies using a

narrower bandwidth of frequencies. Since a wideband spread spectrum signal is very hard to detect, it appears as nothing more than slight rise in the "noise floor" or interference level. With other technologies, the power of the signal is concentrated in a narrower band which makes it easier to detect.

Increased privacy is inherent in CDMA technology. CDMA phone calls will be secure from the casual eavesdropper since, unlike an analog conversation, a simple radio receiver will not be able to pick individual digital conversation out of the overall RF radiation in frequency band.

## Synchronization

In the final stages of the encoding of the radio link from the base station to the mobile, CDMA adds a special "pseudo-random code" to the signal that repeats itself after a finite amount of time. Base Station in the system distinguishes themselves from each other by transmitting different portions of the code at given time. In other words, the base stations transmit time offset versions of the same pseudo-random code. In order to assure that time offsets used remain unique from each other, CDMA stations must remain synchronized to a common time reference.

The primary source of the very precise synchronization signals required by CDMA systems is the global positioning system (GPS). GPS is radio navigation system based on a constellation of orbiting satellites. Since the GPS system covers the entire surface of the earth, it provides a readily available method for determining position and time to as many receivers as are required.

## The balancing act

CDMA cell coverage is dependent upon the way the system is designed. In fact, three primary system characteristics – coverage, quality and capacity – must be balanced off each other to arrive at the desired level of system performance.

In a CDMA system these three characteristics are tightly inter-related. Even higher capacity might be achieved through some degree of degradation in coverage and or quality. Since these parameters are all intertwined operators can not have the best of all worlds: three times wider coverage, 40 times capacity, and "CD" quality sound. For example, 13 Kbps vocoder provides better sound quality, but reduces system capacity as compared to an 8 kbps vocoder.

## 2. MATLAB Code

### Mother file

```
% the mother file
%initialization signal
clc;
fb=1;                   %Data in Hz
tb=1/fb;                %bit interval of data
snr=[0 0.5 1 1.5 2 2.5 3 3.5 4 4.5 5 5.5 6 6.5 7 7.5 8 8.5 9 9.5 10];  %SNR after noise is added
sigPower=10;            %Signal Power
BER=[];                 %for BER result
fc = 3*fb;              %Carrier frequency in KHz
fs = 10*fc;             %Sampling frequency

% specify type of transmitter
sync=menu('Select','Synchronised Transmitter & Receiver', 'Asynchronised Transmitter and Receiver');

% specify type of data encoding
coding=menu('Select the coding technique','none', 'Hamming coding','Linear Block coding','Cyclic coding','Convolution code');

% specify type of spreading code
spread_code=menu('Select Spreading Code','Hadamard-Walsh','PN Sequences','Gold sequences');

% specify type of channel
channel=menu('Select kind of channel', 'AWGN Channel','Fading channel','Both AWGN & Fading');
if channel>1

% specify type of diversity technique
diverse=menu('Select type of diversity technique', 'Space Diversity','Equal Gain combining','Marginal Gain combining');
```

```
end;

% specify type of Environment
env=menu('select the type of ennvironment','Single user', 'Multilple user');

flag=1;        %to control the no. of times of output graph

if env==1        %single user
   users=1;
   a_user=1;
else           % multiple users
   users=input('Enter the number of users : ');
   a_user=input('Enter the desired user : ');
end;
chips=input('Enter the number of chips : ');
tc=tb/chips;           %chip interval

%generation of data stream randomly
Bitstreamlength=16;
Bitstream=infobits(users,Bitstreamlength);
data=[];
for d=1:users
data(d,:)=Bitstream(d,:);
end
data1=data;

%coding input data for error correction and detection
% coverting data to RZ form
dataz=(data+1)./2;
nn=7;
kk=4;
switch coding
case 1
   dt_encode=dataz;
case 2
   dt_encode=encoder(dataz,2,nn,kk);
case 3
   dt_encode=encoder(dataz,3,nn,kk);
case 4
   dt_encode=encoder(dataz,4,nn,kk);
case 5
   dt_encode=encoder(dataz,5,nn,kk);
end;

%reconverting encoded data into NRZ form
dt_encode=dt_encode*2-1;

if flag==1
%Converting input data array and encoded data into continuous function of time
nos=40;         %no. of samples
ts=tb/nos;       %sampling interval
%for data
n=0;            %initialize start of time
```

```matlab
dt=[ ];          % to store data as continuous function of time
l=1;
t=[];            %time index
for T=1:length(data(a_user,:))
  if data(a_user,T)==1
    t(1,l:nos+l-1)=n:ts:n+tb-ts;
    dt(1,l:nos+l-1)=1;
  elseif data(a_user,T)==-1
    t(1,l:nos+l-1)=n:ts:n+tb-ts;
    dt(1,l:nos+l-1)=-1;
  end;
  n=n+tb;
  l=l+nos;
end;

%for encoded data
n=0;             %initialize start of time
dt_en=[ ];       % to store encoded data as continuous function of time
l=1;
t1=[];           %time index
for T=1:length(dt_encode(a_user,:))
  if dt_encode(a_user,T)==1
    t1(1,l:nos+l-1)=n:ts:n+tb-ts;
    dt_en(1,l:nos+l-1)=1;
  elseif dt_encode(a_user,T)==-1
    t1(1,l:nos+l-1)=n:ts:n+tb-ts;
    dt_en(1,l:nos+l-1)=-1;
  end;
  n=n+tb;
  l=l+nos;
end;

%plotting data & encoded data sequence
subplot(3,1,1);
plot(t,dt);
axis([0 length(data(a_user,:))*tb -1.5 1.5]);
title 'original time sequence(data)'
xlabel 'Time';
ylabel 'Amplitude';
subplot(3,1,2);
plot(t1,dt_en);
axis([0 length(dt_encode(a_user,:))*tb -1.5 1.5]);
title 'encoded data'
xlabel 'Time';
ylabel 'Amplitude';
pause;
end;
%Generation of spreading codes
switch spread_code
case 1
   codes=hadamardwalsh(users,chips);
case 2
   codes=pn_seq(users,chips,a_user);
```

```
case 3
    codes=gold_seq(users,chips,a_user);
end;

%Assigning spreading code to desired user
code=codes;

%Spreading encoded data stream
spread_bits=spread(dt_encode,code,a_user);   % Information bits after spreading
dt1=spread_bits;
a=length(dt1(a_user,:));

%Converting spreaded data array into continuous function of time
nos=10;               %no. of samples
tc=tb/length(code);      %chip duration
ts=tc/nos;            %sampling interval
n=0;                %initialize start of time
dt_s=[ ];             % to store spreaded data as continuous function of time
l=1;
t=[];               %time index
for T=1:length(dt1(a_user,:))
  if dt1(a_user,T)==1
    t(1,l:nos+l-1)=n:ts:n+tc-ts;
    dt_s(1,l:nos+l-1)=1;
  elseif dt1(a_user,T)==-1
    t(1,l:nos+l-1)=n:ts:n+tc-ts;
    dt_s(1,l:nos+l-1)=-1;
  end;
  n=n+tc;
  l=l+nos;
end;

if flag==1
%Graph of  after spreading the encoded data
subplot(3,1,3);
plot(t,dt_s);
title 'spreaded encoded-data'
xlabel 'Time';
ylabel 'Amplitude';
axis([0 length(dt1(a_user,:))*tc -1.5 1.5]);
pause;

% Analysis of orignal encoded data sequence frequency spectrum
dt_en_f=fftshift(abs(fft(dt_en,2*length(dt_en))));
subplot(2,1,1);
n=1:length(dt_en_f);
plot(n,dt_en_f,'r');
title 'Frequency spectrum of Orignal data sequence'

% Analysis of spreaded data sequence frequency spectrum
dt_s_f=fftshift(abs(fft(dt_s,2*length(dt_s))));
subplot(2,1,2);
n=1:length(dt_s_f);
```

```
plot(n,dt_s_f,'b');
title 'Frequency spectrum of spreaded data sequence';
pause;
end;


%Considering whether synchronous or, asynchronous transmitter
switch sync
case 1
   % QPSK modulation of spreaded data spread_bits
   data_tx=fqpk1(spread_bits,fc,tc,fs,a_user,tb,flag);      %bits to ready for transmission
 case 2
    % QPSK modulation of spreaded data spread_bits
   data_tx=fqpk2(spread_bits,fc,tc,fs,a_user,tb,flag);      %bits to ready for transmission
end;
% low pass filter coefficients
[bb,aa]=ellip(2,.1,45,0.001);
% number of Iterations
loop=length(snr);
for iter=1:loop

% Stimulating channel and adding effect of channel(noise or fading) to transmitted signal
switch channel
case 1
   % Stimulating AWGN channel and adding noise to transmitted signal
   data_tx_noisy=ch_awgn(data_tx,snr(iter),sigPower,a_user);      %noise effect of channel
   dt_rx=data_tx_noisy;
   if flag==1
   fs=6*fc;
   t=(1:length(dt_rx(a_user,:)))/fs;
   plot(t,dt_rx(a_user,:));
   title 'Data recieved at the reciever with AWGN noise'
   xlabel 'Time';
   ylabel 'Amplitude';
   pause;
end;
case 2
   % Stimulating Fading channel
   data_tx_fade=fadingchannel(data_tx,a_user);      %fading effect of channel
   dt_rx=data_tx_fade;
   fs=6*fc;
   if flag==1
      t=(1:length(dt_rx(a_user,:)))/fs;
      plot(t,dt_rx(a_user,:));
      title 'Data recieved at the reciever with AWGN noise'
      xlabel 'Time';
      ylabel 'Amplitude';
      pause;
   end;
case 3
   % Stimulating AWGN & Fading channel and adding noise to transmitted signal
   data_tx_noisy=ch_awgn(data_tx,snr(iter),sigPower,a_user);      %noise effect of channel
   data_tx_fade=fadingchannel(data_tx_noisy,a_user);                %fading effect of channel
   dt_rx=data_tx_fade;
```

```
   fs=6*fc;
   if flag==1
      t=(1:length(dt_rx(a_user,:)))/fs;
      plot(t,dt_rx(a_user,:));
      title 'Data recieved at the reciever with AWGN noise'
      xlabel 'Time';
      ylabel 'Amplitude';
      pause;
   end;
end;
%effect of multiple users "MULTIPLE ACCESS INTERFERENCE"
a=size(dt_rx,1);          %no. of users
dtr=[];
dtr=sum(dt_rx,1);          %recived signal is sum of signal of all user
dt_rx=dtr./a;              %effective power of desired signal is reduced

%Applying diversity combination techniques by stimulation of rake reciever
%creating 'ten' time delayed version of recieved signals assuming 'ten' fingers of rake reciever
a=length(dt_rx);
n=2;              %assumed maximum delay as 2*tc
de_dt=[];          %to store time delayed versions
for i=1:10
b=ceil(rand(1)*n*50);
dt_r=dt_rx(1,b:a);
dt_r=[dt_r dt_rx(1,1:b-1)];     %adding delay
de_dt(i,:)=dt_r(1,:);
end;
if channel>1
   switch diverse
case 1
  % applying Space Diversity technique
  j=1;             % to track the delayed sequence while sorting
  for i=1:10
    dedt(i,:)=[sum(de_dt(i,:).*de_dt(i,:))/a , j];
    j=j+1;
  end
  dt_a=sortrows(dedt,[1]);    % arranging in ascending order
  dt_h=de_dt((dt_a(10,2)),:); % signal with maximum power
  dt_rx=dt_h;
case 2
  % applying Equal Gain combining Diversity technique
  dt_rx=sum(de_dt,1);
case 3
  % applying Marginal Gain combining Diversity technique
[a,b]=size(de_dt);
k=rand(a,b);
de_dta=de_dt.*k;
dt_rx=sum(de_dta,1);
end;
end;

%demodulating recieved noisy data bits
demod_dt_rx=demod(dt_rx,fc,tc,flag,aa,bb);
```

```
% despreading demodulated data
despreaded_dt=despread(demod_dt_rx, codes, users, a_user,flag);
data_rx=despreaded_dt;

% decoding despreaded data
% coverting despreaded data to RZ form
data_rx=(data_rx+1)./2;
if  coding==1
    decoded_dt=data_rx;
else
    decoded_dt=decoder(data_rx,nn,kk,coding,Bitstreamlength);
end;

%reconverting decoded data into NRZ form
decoded_dt=decoded_dt*2-1;

% final required data i.e what was transmitted
data_rx=decoded_dt;

%calculation of Bit Error rate
Errors = sum(data1(a_user,:)~=data_rx);
BER(iter) = Errors;
flag=2;
end;

%Plotting BER Vs SNR
if flag==1semilogy(snr,BER);
ylabel 'BER';
xlabel 'SNR';
legend('BER','SNR');
title 'Graph of BER & SNR in dB'
end;

%Plotting Pe Vs SNR in dB
Pe=BER./length(data1(a_user,:));
semilogy(snr,Pe,'+');
ylabel 'Pe';
xlabel 'SNR';
%legend('Space Diversity','Equal Gain Combining','Marginal Gain combining');
title 'Graph of Pe Vs SNR (in dB)'

%------___end of programme____-------%
```

## Other function files

### 1.Genration of  information bit-stream

```
function BITS = infobits(K,BIT)
% infobits.m
% this function creates random transmitted bits for the active users
% bit value can be 1 or -1
% bits stored in BITS(K,BIT)
```

```
for k1=1:K
    randn('state',sum(100*clock));
    for i=1:BIT
        U= rand(1)*2-1;
        if U<0
            x=floor(U);
        else
            x=ceil(U);
        end
        BITS(k1,i)=x;
    end
end
```

## 2. Data encoding

### function [dt,g]=encoder(data,type,n,k)

```
% to encode the data bits to improve SNR,performane & minimize BER
% allows to select different coding techniques
[d c]=size(data);
dt2=[];
dt=[];
for h=1:d
if type==2
    dt1=encode(data(h,:),7,4,'hamming');

elseif type==3
    k=3;
    g=hammgen(k);
    dt1=encode(data(h,:),n,k,'linear',g);

elseif type==4
     dt1=encode(data(h,:),n,k,'cyclic');

elseif type==5
    g = poly2trellis([3 3],[4 5 7;7 4 2]);
    dt2(h,:)=[data(h,:) 0 0];
    dt1=convenc(dt2(h,:),g);
end;
if type==5
    dt(h,:)=dt1(1,:);
else
    dt1=dt1';
    dt(h,:)=dt1(1,:);
end;
end;
```

## 3. For generation of pn-sequence

### function [pn1]=pn_seq(n,ch,x);

```
%generates pn sequence of chip length 'ch'
%avoid the just next line for total arbitarary pn-sequence
%rand('seed',931316785)
ch=log2(ch+1)                    % no. of shift registers
```

```matlab
pn2=[ ];
k=[];
for m=1:n                %m users
   k=rand(1,ch);      % initializing shift registers by random sequence k
      for i=1:length(k)
         g=k(i)*2-1;
   if g>0
      k(i)=ceil(g);
   elseif g<0
      k(i)=ceil(g);
    end
  end
if ch>3
t=ch;                % No. of shift registers
N=2;
a=rem(t,N);
if a==0              % Even no. of shift registers
 um=[];
 a=[];
 for f=1:t
   a(1,f)=k(:,f);
 end
for f=1:t
   um=[um a(1,f)];
end
for sc=0:((t-2)/2)-1
   z=2*sc;
   for u=1:t-z-2
      lm=um;
      zm=lm;
      pn=[];
      for q=1:2^(t)-1
         lm=[lm(:,t) lm];         % shifting bits
         lm(:,t+1)=[];
         tagselect=lm(:,1:t-z-2);
         s=length(tagselect);
         for inc=1:s-1
            v=xor(tagselect(:,inc),tagselect(:,inc+1));  %xor the taged bits as per the system
            tagselect(:,inc+1)=v;
         end
         zm=[v zm];
         pn=[pn zm(:,t+1)];        % pn seq being formed
         zm(:,t+1)=[];
         lm=zm;
      end
   end
end

else  % odd no. of shift registers
 um=[];
 a=[];
 for f=1:t
   a(1,f)=k(:,f);
```

```matlab
  end
  z=0;
  for f=1:t
    um=[um a(1,f)];
  end
  for sc=0:((t-3)/2)-1
    z=2*sc;
    for u=1:t-z-1
      lm=um;
      zm=lm;
      pn=[ ];
      for q=1:2^(t)-1
        lm=[lm(:,t) lm];          % shifting bits
        lm(:,t+1)=[];
        tagselect=lm(:,1:t-z-1);
        s=length(tagselect);
        for inc=1:s-1
          v=xor(tagselect(:,inc),tagselect(:,inc+1));   %xor the taged bits as per the system
          tagselect(:,inc+1)=v;
        end
        zm=[v zm];
        pn=[pn zm(:,t+1)];        % pn seq being formed
        zm(:,t+1)=[];
        lm=zm;
      end
    end
  end
end
pn2(m,:)=pn;
else
pn2(m,:)=k;
end;
end;                  % to continue generation of pn seq for other users
pn1=pn2;

%Converting to NRZ form
pn1=pn1*2-1;

%displaying generated pn-sequences
disp('PN-sequences of all users in matrix form with Nth row for Nth user : ')
disp(pn1)

%extracting desired codes
disp('PN-sequence of desired user is : ')
disp(pn1(x,:))
codes=pn1(x,:);
```

## 4. Generation of Hadamard-Walsh codes

**function codes=hadamardwalsh(users,chips)**
```matlab
% generates the hadamard-walsh codes for a given set of active users
% input is the number of users
if users>chips
   C=hadamard(128);
```

```
else
    C=hadamard(chips);
end;
codes=C(1:users,1:chips);
```

## 5. Spreading

### function dt1=spread(data,codes,a_user);

```
%File to spread the data stream
% It inputs the encoded data & spreading codes and spreads the encoded data stream

a=length(data(a_user,:));
spread_dt=[ ];
%generation of codes and its compliment
ncodes=codes.*-1;
d=size(data,1);          % no. of users
for t=1:d
%spreading data sequences using returned codes
for k=length(codes(t,:))-1;
l=1;

for m=1:a
   if data(t,m)==1
      spread_dt(t,l:l+k)=codes(t,:);
   elseif data(t,m)==-1
      spread_dt(t,l:l+k)=ncodes(t,:);
   end;
   l=1+l+k;
end;
end;
end;
dt1=spread_dt;
%disp(' spreaded data of desired user is:  ')
%disp(dt1(a_user,:))
```

## 6. QPSK modulation

### function [k]=fqpk2(dt,fc,tc,fs,a_user,tb,flag)

```
%qmod: QPSK modulator; generates I & Q channels bte & bto from spreaded data and hence modulates the
%spreaded sequence bt.
%it accepts spreaded data, carrier frequency & chip duration as input and returns the modulated bit-
%stream ready for transmission
a=length(dt);
w=2*pi*fc;
ds=size(dt,1);     % for no. of users

if flag==1
%defining odd & even clocks clko & clke respectively.
%even clock
for i=0:2:a
   clke(i+1)=-1;
   clke(i+2)=1;
```

```
    end;
%odd clock
 for i=0:2:a
    clko(i+1)=1;
    clko(i+2)=-1;
 end;

T=(1:length(clke))*tc;
subplot(2,1,1);
stairs(T,clke,'r');
axis([0 15 -2 2]);
title 'I- channel (odd) clock';
subplot(2,1,2);
T=(1:length(clko))*tc;
stairs(T,clko,'b');
axis([0 15 -2 2]);
title 'Q- channel (even) clock';
end;
%Extracting I & Q channels
%I channel
for j=1:ds
for i=0:2:a-1
    if dt(j,i+1)==1
        dto(j,i+1)=1;
        dto(j,i+2)=1;
    elseif dt(j,i+1)==-1
        dto(j,i+1)=-1;
        dto(j,i+2)=-1;
    end;
end;
%Q channel
for i=0:2:a-1
    if dt(j,i+1)==-1 %initial val prob i to i-1
        dte(j,i+2)=-1;
        dte(j,i+3)=-1;
    elseif dt(j,i+1)==1
        dte(j,i+2)=1;
        dte(j,i+3)=1;
    end;
end;
dte(j,1)=-1;
end;
if flag==1
pause
T=(1:length(dte(a_user,:)))*tc;
subplot(2,1,1)
stairs(T,dte(a_user,:),'r')
axis([0 a*tc -2 2])
title 'I- channel data'
T=(1:length(dto(a_user,:)))*tc;
subplot(2,1,2)
stairs(T,dto(a_user,:),'b');
axis([0 a*tc -2 2])
```

```
title 'Q- channel data'
pause
end;
%Converting input data array and encoded data into continuous function of time
%modulating I & Q channels
%I-channel
k1=[];
%sampling rate=fs=6*f so signals are cos(pi*n/3)and sin(pi*n/3)
fs=15/tc;
for j=1:ds
l=1;
c=1;
b=0;
nos=15;

for T=1:length(dte(j,:))
  if dte(j,T)==1
    n=b*tc:1/fs:(b+1)*tc-1/fs;
    k1(j,l:c*nos)=cos(2*pi*n*fc);
  else
    n=b*tc:1/fs:(b+1)*tc-1/fs;
    k1(j,l:c*nos)=-cos(2*pi*n*fc);
  end;
  l=l+nos;
  c=c+1;
  b=b+1;
end;
end;
if flag==1
n1=0:1/fs:length(dte(a_user,:))*tc-1/fs;
subplot(2,1,1)
plot(n1,k1(a_user,:));
axis([0 length(dte(a_user,:))*tc/fc -1.5 1.5]);
title 'plot for I-channel modulated output ';
end;

%Q-channel
k2=[ ];

for j=1:ds
l=1;
c=1;
b=0;

for T=1:length(dto(j,:))
  if dto(j,T)==1
    n=b*tc:1/fs:(b+1)*tc-1/fs;
    k2(j,l:c*nos)=sin(2*pi*n*fc);
  else
    n=b*tc:1/fs:(b+1)*tc-1/fs;
    k2(j,l:c*nos)=-sin(2*pi*n*fc);
  end;
  l=l+nos;
```

```
      c=c+1;
      b=b+1;
   end;
end;
if flag==1
n2=0:1/fs:length(dto(a_user,:))*tc-1/fs;
subplot(2,1,2);
title 'plot for Q-channel';
plot(n2,k2(a_user,:));
axis([0 length(dto(a_user,:))*tc/fc -1.5 1.5]);
title 'plot for Q-channel modulated output ';
end;

%final QPSK output
k=[];
c=length(k1(a_user,:));
d=length(k2(a_user,:));
if c>d
   c=d ;
   n1=n2;
end;

for j=1:ds
   for t=1:c
      k(j,t)=k1(j,t)+k2(j,t);
   end;
end;
if flag==1
pause
subplot
plot(n1,k)
axis([0 length(dto(a_user,:))*tc/fc -2 2]);
title 'QPSK modulated waveform of data'
pause

%frequency spectrum analysis
subplot
kmag=abs(fftshift(fft(k(a_user,:),length(k(a_user,:)))));          %kmag  is  magnitude  of  frequency
spectrum
w=1:length(kmag);
plot(w,kmag);
title 'Frequency spectrum of QPSK modulated output'
pause
end;
```

## 7. QPSK Demodulation

### function demod_dt_rx=demod(dt_rx,fc,tc,flag,aa,bb)

```
%--QPSK DEMODULATOR--%
% To demodulate the recieved data dt_rx and return demod_dt_rx

fs=15/tc;            % sampling frequency fs>fc to avoid aliasing

%Extracting I-channel data
```

```
%coherent detection
dt_i=[]; %For demodulated I channel data
dt_rx_i=[];
b1=0;
l=1;
c=1;
a1=ceil(length(dt_rx)/(fs*tc));
a2=ceil(fs*tc);
for T=1:a1
    n=b1*tc:1/fs:(b1+1)*tc-1/fs;
    dt_rx_i=dt_rx(1,l:c*a2).*cos(2*pi*n*fc);   % multipying by same carrier for I channel
    % low pass filter to separate out the dc term
     dt_f_i=filter(bb,aa,dt_rx_i);

    zi=sum(dt_f_i);
    if zi>0
        dt_i(T)=1;
    elseif zi<0
        dt_i(T)=-1;
    end
  l=l+a2;
  c=c+1;
  b1=b1+1;
end

%Extracting Q-channel data
%coherent detection
dt_q=[];        %For demodulated Q channel data
b1=0;
l=1;
c=1;
for T=1:a1
    n=b1*tc:1/fs:(b1+1)*tc-1/fs;
    dt_rx_q=dt_rx(1,l:c*a2).*sin(2*pi*n*fc);   % multipying by same carrier for Q channel
    % low pass filter to separate out the dc term
    dt_f_q=filter(bb,aa,dt_rx_q);
    zq=sum(dt_f_q);
    if zq>0
        dt_q(T)=1;
    elseif zq<0
        dt_q(T)=-1;
    end
  l=l+a2;
  c=c+1;
  b1=b1+1;
end
if flag==1
a=length(dt_i);
T=(1:a)*tc;
subplot(2,1,1)
stairs(T,dt_i,'r')
axis([1 a*tc -2 2])
title 'demodulated I- channel data'
```

```
a=length(dt_q);
T=(1:a)*tc;
subplot(2,1,2)
stairs(T,dt_q,'b');
axis([1 a*tc -2 2])
title 'demodulated Q-channel data'
pause
end;
% obtaining orignal data i.e. combining I & Q channel
demod_dt_rx=[];            % to store the final demodulated data
for t=1:2:length(dt_i)
    demod_dt_rx(t)=dt_i(t);
end

for t=2:2:length(dt_q)
    demod_dt_rx(t)=dt_q(t);
end
if flag==1
a=length(demod_dt_rx);
T=(1:a)*tc;
subplot
stairs(T,demod_dt_rx,'b');
axis([1 a*tc -2 2])
title 'demodulated but spreaded data'
pause
end;
```

# References

♦ John. G. Proakis, "Digital Communications", McGraw-Hill Series in Electrical Computer Engineering, Third Ed.

♦ Theodore. S. Rappaport, "Wireless Communications", Prentice-Hall Communications Engineering and Emerging Technologies Series.

♦ A. J. Viterbi, *CDMA Principles of Spread Spectrum Communication.* Reading, MA: Addison-Wesley, 1995.

♦ M. Nakagami, "The m-distribution—A general formula of intensity distribution of rapid fading," in *Statistical Methods in Radio Wave Propagation.* New York: Pergamon, 1960, pp. 3–36.

♦ W. Feller, *An Introduction to Probability Theory and Its Applications,* Vol. II. New York: Wiley, 1970.

♦ William C Y Lee, \Overview of Cellular CDMA", IEEE Transactions on Vehicular Technology, v 40, n 2, May 1991, pp 291-302.

♦ Klien S Gilhousen, Irwin M Jacobs, Roberto Padovani, Andrew J Viterbi, Lindsay A Weaver Jr, Charles E Wheatley III, \On the Capacity of a Cellular CDMA System", IEEE Transactions on Vehicular Technology, v 40, n 2, May 1991, pp 303-312.

♦ Audrey M Viterbi, Andrew J Viterbi, \Erlang capacity of a power controlled CDMA system", IEEE Journal on Selected Areas in Communications, v 11, n 6, Aug 1993, pp 892-900.

♦ Andrew J Viterbi, Audrey M Viterbi, Klein S Gilhousen, Ephraim Zehavi, \Soft handoff extends CDMA cell coverage and increases reverse link capacity", IEEE Journal on Selected Areas in Communications, v 12, n 8, Oct 1994, pp 1281-1288.

♦ Klein S Gilhousen, Irwin M Jacobs, Roberto Padovani, Lindsay A Weaver, Increased Capacity Using CDMA for Mobile Satellite Communication", IEEE Journal on Selected Areas in Communications, vol 8, n 4, May 1990, pp 503-514.

♦ Linda M Zeger, Mark E Newbury, \CDMA Capacity with Added Carriers in a Cellular Network", Bell Labs Technical Journal, July-Sep 1999, p 104-119.

♦ Theodore S Rappaport, \Wireless Communications", Prentice Hall PTR, Upper Saddle River, New Jersey.