THE COMPARATIVE STUDY OF MULTIPLE ACCESS TECHNIQUES

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DECLARATION

This is to certify that this work has been done by us and it has not been submitted elsewhere for the award of any degree or diploma.

CERTIFICATE

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ABSTRACT

In this Thesis, the system performance of multiple access schemes in Wide-band (WB) communications is evaluated in a multi path and multi user fading environment. Three multiple access schemes, namely Time Hopping (TH), Direct Sequence (DS) and hybrid Direct Sequence-Time Hopping (DS-TH) are investigated. The TH multiple access has been well studied in Radar communication systems. This research extends the previous studies by applying three pulse modulations techniques, including pulse position modulation, pulse shift keying and pulse amplitude modulation. The idea of the DS multiple access schemes is also generalized from the well-known Direct Sequence-Code Division Multiple Access (DS-CDMA) cellular systems to WB radios. It is shown that the DS multiple access has the potential to reach higher data transmission rates and that TH techniques are more resistant to fading. The DS-TH multiple access schemes are proposed by combining the advantages of both TH and DS multiple access schemes. Results show that the DS-TH ultra wideband achieves better system performance while maintaining the required data transmission rate and multiple access capacity. The system performance is illustrated and examined in terms of the signal to noise plus interference ratio, bit error rate and outage probability.

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Content

Page No

CHAPTER 1

1.1 INTRODUCTION

As local, state, and federal governments plan and install radio networks supporting communications requirements, the success of these networks may be driven by the availability of the radio frequency spectrum. The radio frequency spectrum, a finite natural resource, has greater demands placed on it every day. In an effort to make the most efficient use of this resource, various technologies have been developed so that multiple, simultaneous users can be supported in a finite amount of spectrum. This concept is called "multiple access." The three most commonly used access methods are frequency division multiple access (FDMA), time division multiple access (TDMA), and code division multiple access (CDMA).

To plan, design, procure, and use any kind of radio communications, a basic understanding of the technologies involved is essential. In the last 20 years, great strides have been made in this area of technology. Wireless communications technologies, which were virtually unheard of as recently as the late 1970s, are now prevalent throughout today's society and growing in demand. However, limited by the finite frequency spectrum, spectrally efficient technologies have not kept pace with today's high traffic demands. To ensure profit grows parallel with the demand for wireless technologies, manufacturers have had to develop methods of putting more users in the same spectrum space.

This report will discuss the functionality of each access method (FDMA, TDMA and CDMA), the advantages and disadvantages of each technology, and various forms of implementation for each technology. FDMA and TDMA are currently being used to support conventional and trunked radio systems, as well as commercial cellular systems. CDMA is being used primarily in cellular systems at this time. (See PSWN Report: Comparisons of Conventional and Trunked Systems, May 1999.) An easy to understand example is given to provide the reader a general overview of how each technology differs. Next, a comparison is made between the technologies followed by a discussion of the primary benefits each technology offers to wireless communication system. Finally, the report discusses how the TIA-102 (Project 25) and TETRA (Terrestrial Trunked Radio) standards can support public safety communications systems. This document is not intended to cover the tremendous amount of detailed technical information available on this subject.

General concepts to keep in mind: Each radio frequency or "channel" must consist of enough space to carry the intended signal to its destination. Therefore, when the terms "spectrum space," "radio frequencies," or "channels" are used, it should be understood that actual physical space is involved, albeit invisible to the human eye. (It should also be noted the terms "frequency" and "channel" can be used interchangeably, and "channel' may refer to a set of two frequencies used to support one link, such as between a base station and a vehicle.) Radio signals vary in size depending on the type of "message" being carried. The size of the signal is referred to as "bandwidth."

MULTIPLE-ACCESS TECHNIQUES

A limited amount of bandwidth is allocated for wireless services. A wireless system is required to accommodate as many users as possible by effectively sharing the limited bandwidth. Therefore, in the field of communications, the term multiple access could be defined as a means of allowing multiple users to simultaneously share the finite bandwidth with least possible degradation in the performance of the system. There are several techniques how multiple accessing can be achieved. The are four basic schemes

Frequency Division Multiple-Access (FDMA)
Time Division Multiple-Access (TDMA)
Code Division Multiple-Access (CDMA)
Space Division Multiple-Access (SDMA)

Here we will discuss the functionality of each access method (FDMA, TDMA, CDMA and SDMA), the advantages and disadvantages of each technology, and various forms of implementation for each technology. FDMA and TDMA are currently being used to support conventional and trunked radio systems, as well as commercial cellular systems. CDMA is being used primarily in cellular systems at this time. An easy to understand example is given to provide the reader a general overview of how each technology differs. Next, a comparison is made between the technologies followed by a discussion of the primary benefits each technology offers to wireless communication system.

1.2UNDERSTANDING FDMA, TDMA, AND CDMA: A TECHNICAL EXAMPLE

The best way to describe the differences between FDMA, TDMA, and CDMA technologies is with an example of how they work. The following example is one of the best.

Picture a large room with a group of people divided up into pairs. Each pair would like to hold their own conversation with no interest in what is being said by the other pairs. For these conversations to take place without interruption from other conversations, it is necessary to define an isolated environment for each conversation. In this example, the room should be considered as a slice of the radio spectrum specifically allocated to be used by this group of people. Imagine each pair communicating through cellular telephones or radios.

Applying an FDMA system to this analogy, the single large room (slice of spectrum) would be partitioned with many dividing walls and creating a large number of smaller rooms. A single pair of people would enter each small room and hold their conversation. Each room is like a single frequency/channel. No one else could use the room (or frequency) until the conversation was complete, whether or not the parties were actually talking. When the conversation is completed, the first pair of people would leave and another pair would then be able to enter that small room.

In a TDMA environment, each of the small rooms would be able to accommodate multiple conversations "simultaneously." For example, with a three-slot TDMA system, each "room" would contain up to three pairs of people, with the different pairs taking turns talking. According to this system, each pair can speak for 20 seconds during each minute. Pair A would use 0:01 second through 0:20 second, pair B would use 0:21 second through 0:40 second, and pair C would use 0:41 second through 0:60 second. However, even if there were fewer than three pairs in the small room, each pair would still be limited to 20 seconds per minute.

Using the CDMA technology, all the little rooms would be eliminated. All pairs of people would enter the single large room (our spectrum space). Each pair would be holding their conversations in a different language and therefore they could use the air in the whole room to carry their voices while experiencing little interference from the other pairs. The air in the room is analogous to a wideband "carrier" and the languages represent the "codes" assigned by the CDMA system. In addition, language "filters" would be

incorporated so that, for example, people speaking German would hear virtually nothing from those speaking another language.

Additional pairs could be added, each speaking a unique language (as defined by the unique code) until the overall "background noise" (interference from other users) made it too difficult to hold a clear conversation. By controlling the voice volume (signal strength) of all users to a minimum, the number of conversations that could take place in the room could be maximized (i.e., maximize the number of users per carrier). Additional pairs can be easily added to the room without much interference to the other pairs.

1.3. FDMA—FREQUENCY DIVISION MULTIPLE ACCESS

A. Overview

Frequency division is the original multiple access technique. Currently, most legacy public safety wireless networks use FDMA to improve spectrum efficiency. FDMA is used throughout the commercial wireless industry. Legacy commercial telecommunication networks (analog networks based on Advanced Mobile Phone Service [AMPS] and Total Access Communications System [TACS] standards) are built on a backbone of cellular base stations, using the FDMA technology. However, due to increased spectrum efficiency of CDMA and TDMA systems, very few, if any, new cellular systems are using FDMA.

B. How it Works

FDMA systems separate a client's large frequency band into several smaller individual bands/channels. Each channel has the ability to support a user. Guardbands are used to separate channels to prevent interference. They are used to isolate channels from adjacent-channel interference.

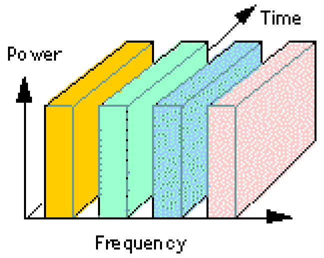

Figure 1: FDMA permits only one user per channel because it allows the user to use the channel 100 percent of the time. Therefore, only the frequency "dimension" is used to define channels. Each block represents a different user

When the FDMA technique is employed, each user is assigned a discrete slice of the radio frequency (RF) spectrum, a "channel" of spectrum space that will vary in size depending on the type of signal being transmitted. In a given amount of spectrum space, the user is granted access to a small sliver of the overall allocation. As long as the user is engaged in "conversation," no other user can access the same spectrum space. An example of this type of access is use of the spectrum by commercial radio broadcasters. In the commercial radio broadcast bands, 535–1705 kHz for amplitude modulation (AM) and 88–108 megahertz (MHz) for frequency modulation (FM), each local broadcast station (user) is assigned a specific slice of spectrum within the frequency band allocated for that purpose. As long as the station broadcasts, no other radio station in the same area can use that radio frequency bandwidth to send a signal. Another broadcast station can use that same bandwidth only when the distance between the stations is sufficient to reduce the risk of interference.

In a conventional two-frequency public safety radio system, one frequency is used to transmit and the other is used to receive. Each channel has its own center frequency and each channel has a bandwidth that is a fraction of the original allotted bandwidth. In this type of system, if an FDMA channel is in use, other users cannot use it until the “conversation” is complete. This is one of the inefficiencies of FDMA systems. Figure 2 graphically displays a two-frequency conventional system. The mobile and portable radio users transmit on frequency F1 to the repeater; the repeater then retransmits back to the users on frequency F2. In Figure 2, the F1 lightning symbol is an uplink to the repeater while the F2 lightning symbol is a downlink.

Base Station/ Repeater Site F1 (Talk-back) F2 “Hi Joe !” F2 (Talk-out) F1 “Hi Joe !” Mobile Radio User Portable Radio User

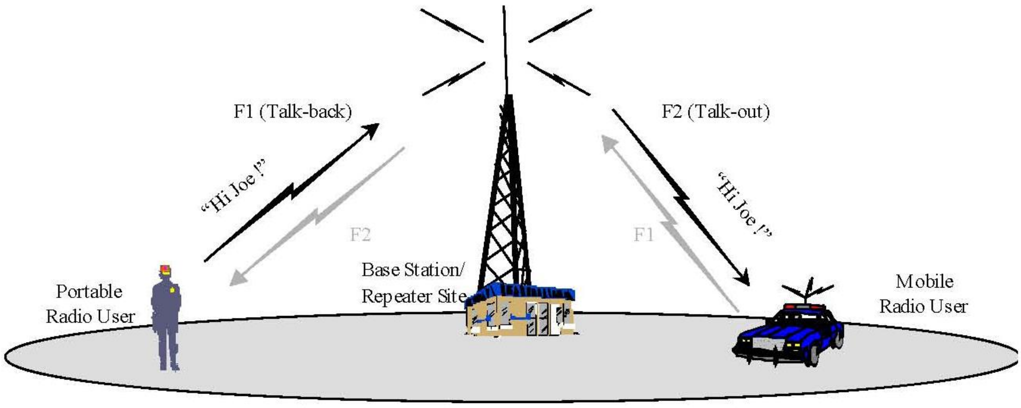

Figure 2: Single-Site Conventional System Configuration Operating in Half Duplex

Project 25's (P25) Phase I standard requires upgrades from standard analog technology with a 25 kHz bandwidth to digital technology with a narrower bandwidth of 12.5 kHz. Implementation of an FDMA system would give each user access to two separate frequency allotments, each with a 12.5 kHz bandwidth. Under P25, this newer equipment is also required to be “backward compatible” to the legacy 25 kHz analog equipment to allow a smooth transition.[3]

Because adjacent channel interference is an important factor in channel quality, frequency planning is a key consideration when selecting fixed or base station locations. Frequency planning is complicated and difficult. Available frequency bands must be researched and analyzed. Transceiver transmission strength affects fixed station range while antenna design affects its coverage patterns. These are also important factors in frequency planning. Figure 3 is a sample base station coverage scheme for a cellular system.

FDMA is one of the earliest multiple-access techniques for cellular systems when continuous transmission is required for analog services. In this technique the bandwidth is divided into a number of channels and distributed among users with a finite portion of bandwidth for permanent use as illustrated in figure. The channels are assigned only when demanded by the users. Therefore when a channel is not in use it becomes a wasted resource. FDMA channels have narrow bandwidth (30Khz) and therefore they are usually implemented in narrowband systems. Since the user has his portion of the bandwidth all the time, FDMA does not require synchronization or timing control, which makes it algorithmically simple. Even though no two users use the same frequency band at the same time, guard bands are introduced between frequency bands to minimize adjacent channel interference. Guard bands are unused frequency slots that separate neighboring channels. This leads to a waste of bandwidth. When continuous transmission is not required, bandwidth goes wasted since it is not being utilized for a portion of the time. In wireless communications, FDMA achieves simultaneous transmission and reception by using Frequency division duplexing (FDD). In order for both the transmitter and the receiver to operate at the same time, FDD requires duplexers. The requirement of duplexers in the FDMA system makes it expensive.

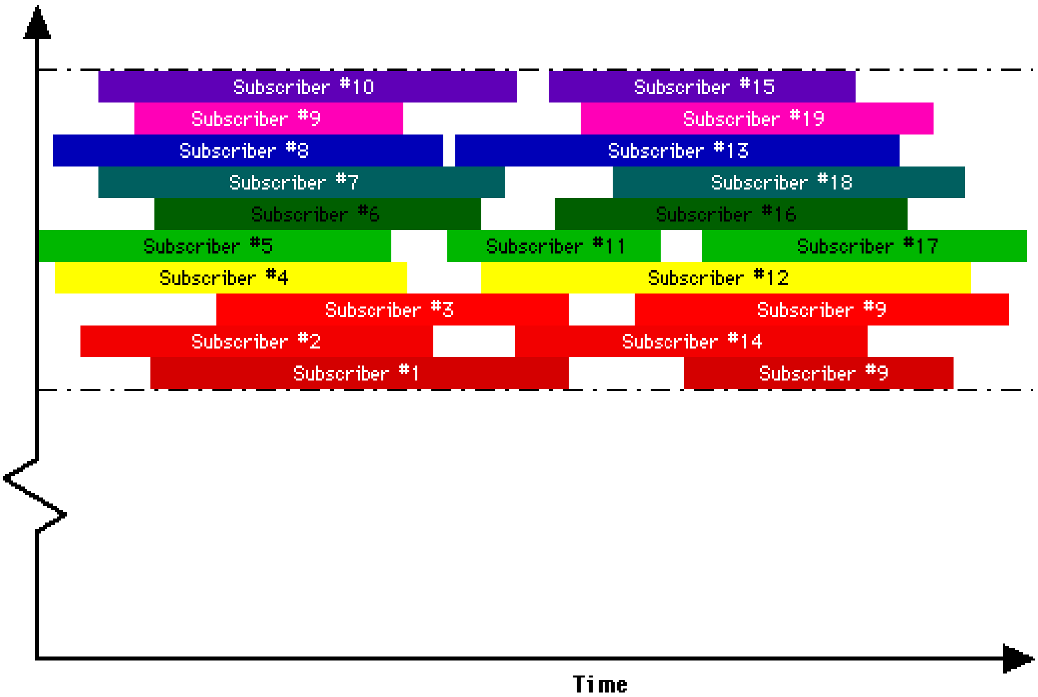

Fig: Schematic allocation of subscriber channels within an assigned frequency band (range).

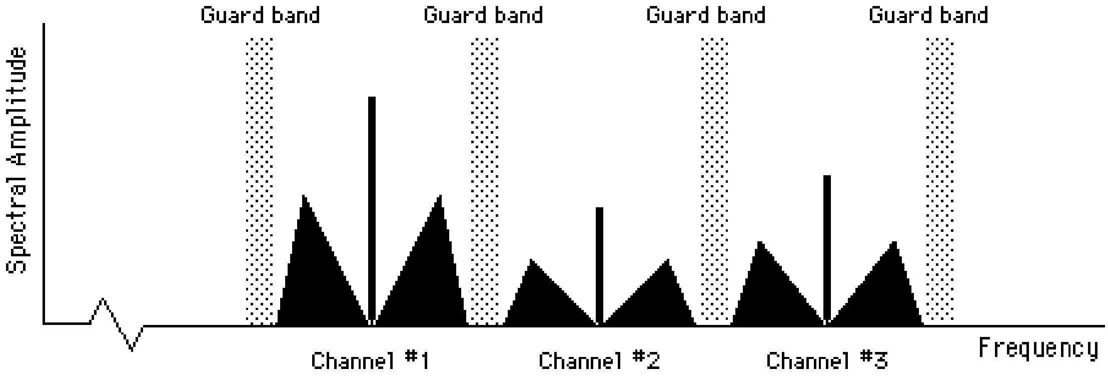

Fig: Schematic frequency spectrum of several subscriber channels.

Orthogonal Frequency Division Multiplexing

In an OFDM system the available bandwidth is divided into a large number of sub-carriers. Data symbols are modulated on these carriers using Inverse Discrete Fourier Transform (IDFT). In OFDMA, each user is assigned different subscribers. The sub-carrier of each user is interleaved with the subscribers assigned to other users. This process is termed carrier allocation. In uplink OFDMA each user has its own multi-path channel through which it communicates with the Base station.

OFDM is a block transmission technique. In the base-band, complex-valued data symbols modulate a large number of tightly grouped carrier waveforms. The transmitted OFDM signal multiplexes several low-rate data streams; each data stream is associated with a given sub-carrier. The main advantage of this concept in a radio environment is that each of the data streams experiences an almost flat fading channel. In slowly fading channels, the Inter Symbol Interference (ISI) and Inter Carrier Interference (ICI) within an OFDM symbol can be avoided with a small loss of transmission energy using the concept of a cyclic prefix.

The OFDM message is generated in the complex base-band. Each symbol is modulated onto the corresponding sub-carrier using variants of phase shift keying (PSK) or different forms of quadrature amplitude modulation (QAM). The data symbols are converted from serial to parallel before data transmission. The frequency spacing between adjacent sub-carriers is $^{2\pi}/_{N}$, where N is the number of sub-carriers. This can be achieved by using the inverse discrete Fourier transform (IDFT), easily implemented as the inverse fast Fourier transform (IFFT) operation. As a result, the

OFDM symbol generated for an N sub-carrier system translates into N samples.

An OFDM signal consists of N orthogonal sub-carriers modulated by N parallel data streams. Each base-band sub-carrier is of the form

$$\phi_k(t) = e^{j2\pi f_k t}$$

Where, f_k is the frequency of the k th sub-carrier. One base-band OFDM symbol (without a cyclic prefix) multiplexes N modulated sub-carriers:

$$s(t) = \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} x_k \phi_k(t), \qquad 0 < t < NT,$$

Where, x_k is the k th complex data symbol (typically taken from a PSK or QAM symbol constellation) and NT is the length of the OFDM symbol. The sub-carrier frequencies are equally spaced.

$$f_k = \frac{k}{NT},$$

which makes the sub-carriers $\varphi_k(t)$ on $0 < t < NT$ orthogonal. The signal separates data symbols in frequency by overlapping sub-carriers, thus using the available spectrum in an efficient way.

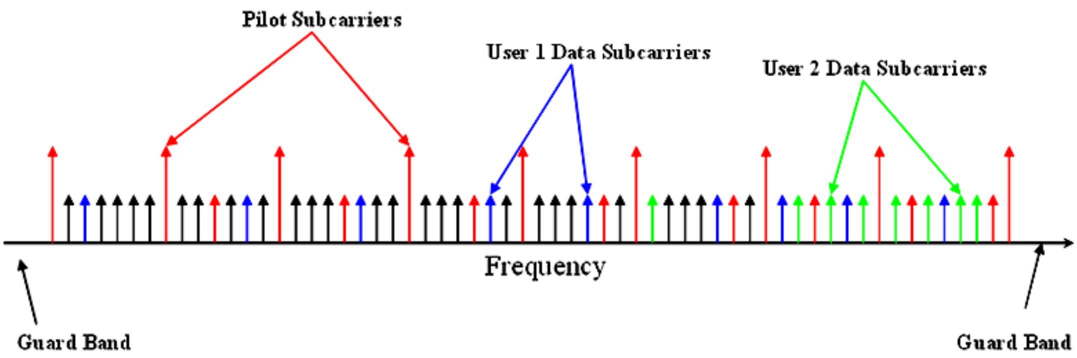

Fig: OFDMA Frequency Description

Figure shows time and frequency characteristics of an OFDM signal sub-carriers. As the OFDM signal is the sum of a large number of independent, identically distributed components its amplitude distribution becomes approximately Gaussian due to the central limit theorem. Therefore, it suffers from large peak-to-average power ratios.

The OFDM could typically be received using a bank of matched filters. However, an alternative demodulation is used in practice. T-spaced sampling of the in-phase and quadrature components of the OFDM symbol yields (ignoring channel impairments such as additive noise or dispersion)

$$s(nT) = \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} x_k e^{j2\pi\frac{nk}{N}}, \qquad 0 \le n \le N-1$$

which is the inverse discrete Fourier transform (IDFT) of the constellation symbols . Accordingly, the sampled data is demodulated with a DFT. This is one of the key properties of OFDM, typically implemented with an FFT.

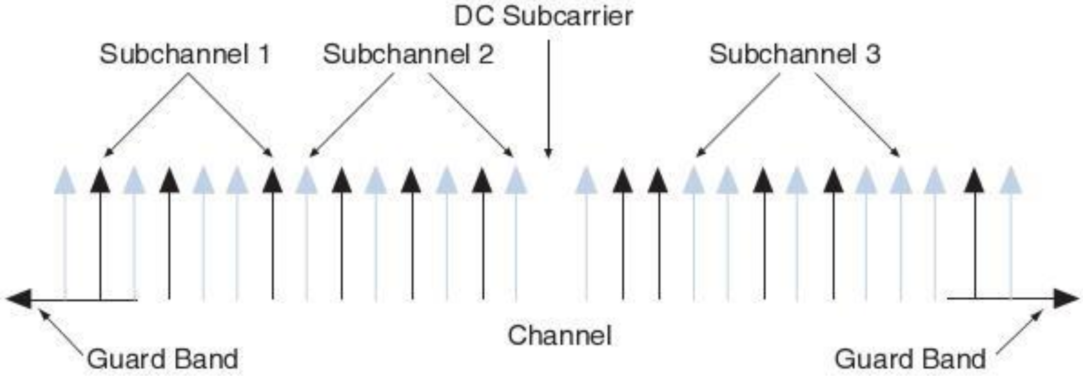

Fig: OFDM Transmitted Signal

C. Advantage and Disadvantages

Advantages

- Simple to implement, from a hardware standpoint
- Fairly efficient with a small base population and when traffic is constant
- P25 equipment is backward compatible to legacy 25 kHz analog radio equipment

Disadvantages

- Network and spectrum planning are intensive
- In a conventional system, because channels are allocated for one user, idle channels add to spectrum inefficiency
- Frequency planning is time-consuming

Single Carrier Frequency Division Multiple Access

SC-FDMA is a new technique that has drawn attention from the communications industry as an attractive alternative to Orthogonal Frequency Single Carrier Frequency Division Multiple Access (SC-FDMA) is a novel method of radio transmission under consideration for deployment in future cellular systems; specifically, in 3rd Generation Partnership Project Long Term Evolution (3GPP LTE) systems. SC-FDMA has drawn great attention from the communications industry as an attractive alternative to Orthogonal Frequency Division Multiple Access (OFDMA).

SC-FDMA uses a frequency domain equalizer to mitigate ISI. We assume that a MMSE equalizer is used, and from the SNR of data delivered in a chunk with MMSE equalization can be written as:

$$\gamma_k = \left(\frac{1}{\frac{1}{M}\sum_{i\in I}\frac{\gamma_{i,k}}{\gamma_{i,k}+1}} - 1 \right)^{-1}$$

If an arbitrary subset of sub-carrier, I (a chunk contains M sub-carriers) is assigned to user k. $\gamma_{i,k}$ is the SNR of sub-carrier i for user k. Using Shannon's formula, the achievable data rate of the chunk for user k has the upper bound.

$$C_k = (B/N)\log_2(1+\gamma_k)$$

Note that the effective bandwidth occupied by user k is B/N Hz, since one chunk is allocated to user k and there are N. chunks in bandwidth B Hz.

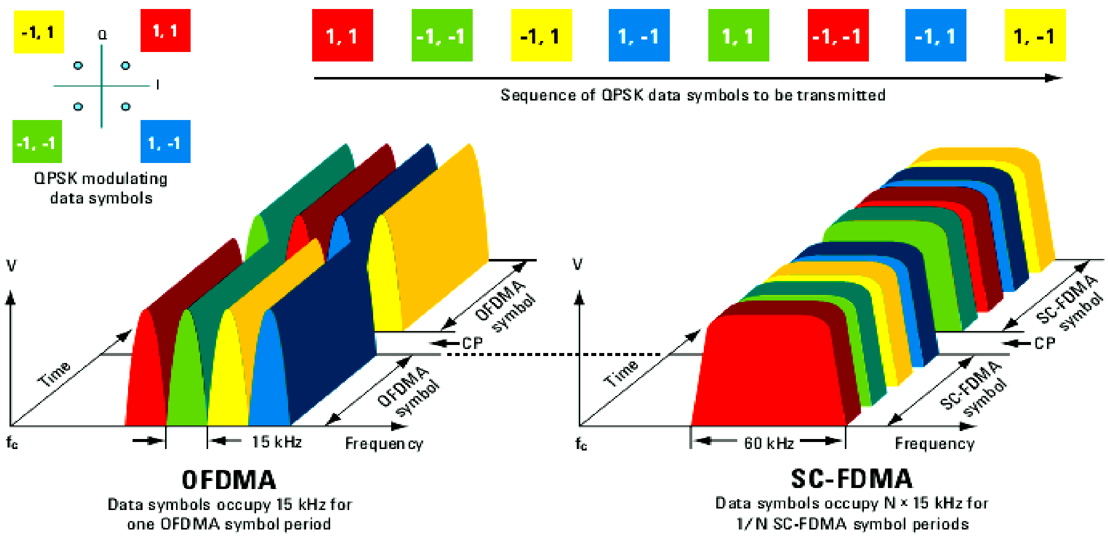

Fig: OFDMA and SC-FDMA transmit a sequence of QPSK data symbols

1.4. TDMA—TIME DIVISION MULTIPLE ACCESS

A. Overview

As the frequency spectrum experiences more traffic, spectrum efficiency becomes increasingly important. TDMA systems were developed as FDMA system spectrum efficiency became insufficient. Not only do TDMA systems split users into an available pair of channels, but they also assign each user an available time-slot/cell within that channel. TDMA systems have the capability to split users into time slots because they transfer digital data, instead of analog data commonly used in legacy FDMA systems. Each of the users takes turns transmitting and receiving in a round-robin fashion. Frequency division is still employed, but these frequencies are now further subdivided into a defined number of time slots per frequency. In reality, only one user is (actually) using the channel at any given moment. Each user is transmitting and receiving in short "bursts." Because TDMA systems do not transmit all of the time, their mobile phones have an extended battery life and talk time.

B. How it works

Similar to an FDMA trunked system, when a user depresses the Push-To-Talk (PTT) switch in a TDMA system, a control channel registers the radio to the closest base station. During registration, the base station assigns the user an available pair of channels, one to transmit and the other to receive. But, unlike an FDMA system registration, a TDMA system registration also assigns an available time-slot within the channel. The user can only send or receive information at that time, regardless of the availability of other time-slots. Information flow is not continuous for any user, but rather is sent and received in bursts. The bursts are re-assembled at the receiving end and appear to provide continuous sound because the process is very fast.

In Figure 4, each row of blocks represents a single channel divided into three time-slots. Calls in a TDMA system start in analog format and are sampled, transforming the call into a digital format. After the call is converted into digital format, the TDMA system places the call into an assigned time slot.

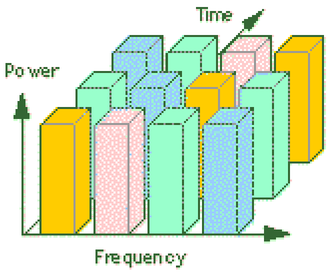

Figure 4: TDMA increases the number of users who have access to particular channel by dividing that channel into time-slots.

Figure 4 is also a graphical display of the efficiency of a TDMA system. The improved efficiency of TDMA over FDMA can be realized through a quick glance at Figures 1 and 4. In Figure 1, the FDMA system supports 4 users while in Figure 4, the TDMA system supports 12 users within the same bandwidth as the FDMA system. There are systems in place today that allow an increase of up to six times the capacity of FDMA alone.

Because TDMA systems also split an allotted portion of the frequency spectrum into smaller slots (channels), they require the same level of frequency planning as FDMA systems. The same careful steps in frequency planning must be taken in both FDMA and TDMA systems.

Time division multiple access (TDMA) is digital transmission technology that allows a number of users to access a single radio- frequency (RF) channel without interference by allocating unique time slots to each user within each channel. The TDMA digital transmission scheme multiplexes three signals over a single channel. The current TDMA standard for cellular divides a single channel into six time slots, with each signal using two slots, providing a 3 to 1 gain in capacity over advanced mobile -phone service (AMPS). Each caller is assigned a specific time slot for transmission. TDMA frame structure showing a data stream divided into frames and those frames divided into time slots.

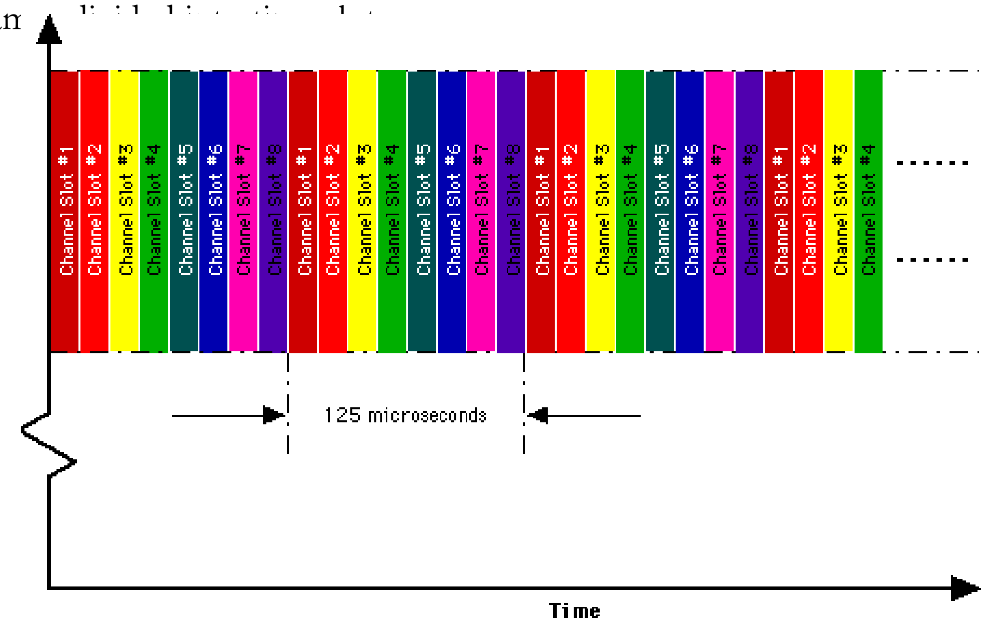

Fig: TDMA Scheme

TDMA relies upon the fact that the audio signal has been digitized; that is, divided into a number of milliseconds-long packets. It allocates a single frequency channel for a short time and then moves to another channel. The digital samples from a single transmitter occupy different time slots in several bands at the same time as shown in figure:

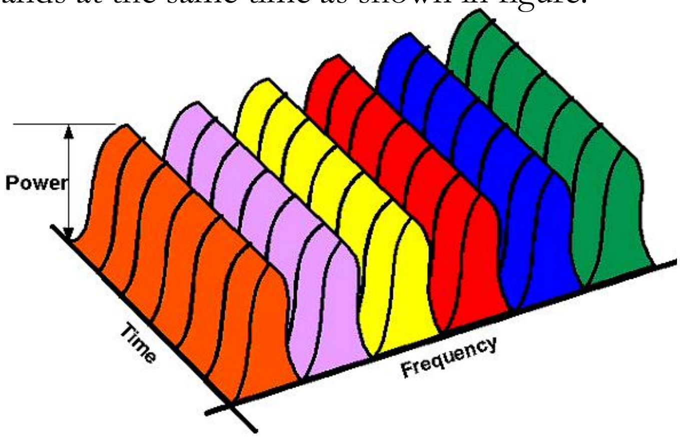

Fig: TDMA

TDMA accomplish talking the digitized information and parsing the bits into fixed size packets for each call. A second call is processed the same way and then the packets are alternated. The speed at which that occurs is transparent to the caller and listener. Figure below illustrates how the packet for each caller are alternated and placed into signal stream.

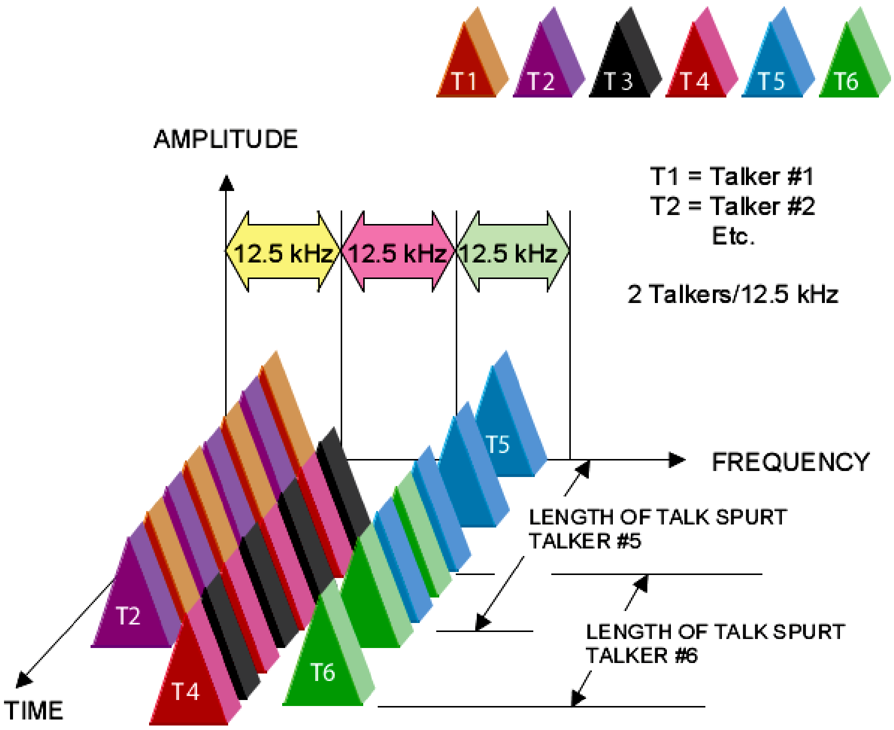

Fig: TDMA Transmission in Channel

As shown in the figure the numbers of talkers has doubled. It is shown that in the case of talkers 5 and 6 that they use the slot as necessary to support when the particular user initiates to talk.

In a TDMA system, each user transmits for $^{1}/_{k}$ of the time through the channel of bandwidth W, with average power KP. Therefore the capacity per user is:

$C_K = (^{1}/_{K})\ W \log_2 (1 + {}^{KP}/_{WN_0})$ which is identical to the capacity of a TDMA system.

C. Advantage and Disadvantages

Advantages

- Extended battery life and talk time
- More efficient use of spectrum, compared to FDMA
- Will accommodate more users in the same spectrum space than an FDMA system which improves capacity in high traffic areas, such as large metropolitan areas
- Efficient utilization of hierarchical cell structures – pico, micro, and macro cells
- Can handle video and audio data efficiently

Disadvantages

- Network and spectrum planning are intensive
- Multipath interference affects call quality
- Dropped calls are possible when users switch in and out of different cells
- Frequency planning is time consuming
- Frequency guard bands add to spectrum inefficiency
- Too few users result in idle channels (rural versus urban environment)
- Higher costs due to greater equipment sophistication

1.5 CDMA—CODE DIVISION MULTIPLE ACCESS

A. Overview

CDMA is a spread spectrum technique used to increase spectrum efficiency over current FDMA and TDMA systems. Although spread spectrum's application to cellular telephony is relatively new, it is not a new technology. Spread spectrum has been used in many military applications, such as anti-jamming (because of the spread signal, it is difficult to interfere with or jam), ranging (measuring the distance of the transmission to determine when it will be received), and secure communications (the spread spectrum signal is very hard to detect).

B. How it works

With CDMA, unique digital codes (Walsh Codes), rather than separate radio frequencies/ channels, are used to differentiate users. The Walsh codes are shared by the mobile phone and the base station, and are called "pseudo-Random Code Sequences." All users access the entire spectrum allocation all of the time. That is, every user uses the entire block of allocated spectrum space to carry his/her message. A user's unique Walsh Code separates the call from all other calls. Figure 5 graphically shows each user simultaneously accessing the fully allotted frequency spectrum.

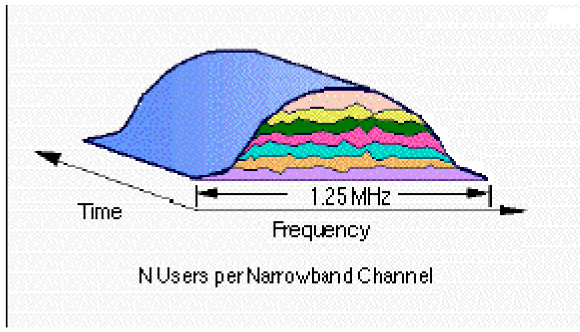

Figure 5*[1] *: *CDMA allows all users access to their entire allocated spectrum.*

CDMA, being a "spread-spectrum" technology, spreads the information contained in a signal over the entire available bandwidth and not simply through one frequency. Due to the wide bandwidth of a spread-spectrum signal, it is very difficult to cause jamming, difficult to interfere with, and difficult to identify. It appears as nothing more than a slight rise in the "noise floor" or interference level, unlike other technologies where the power of the signal is concentrated in a narrower band making it easier to detect. Therefore CDMA systems provide more privacy than FDMA or TDMA systems. These are great advantages over technologies using a narrower bandwidth.

CDMA channels can handle an unspecified number of users. There is not a fixed number. The capacity of the system depends on the quality of current calls. As more users are added, noise is added to the wideband frequency and therefore decreases the quality of current calls. Each user's transmission power increases the level of the frequency spectrum's "noise floor" and therefore decreases the overall call quality for all users. To help eliminate the "noise floor," CDMA mobile phones and base stations use the minimum amount of power required to communicate with each other. They use precise power control to decrease users' transmission power. By decreasing a user's transmission power, the mobile phone has added battery life, increased talk time, and smaller batteries.

Because CDMA is a spread spectrum technology, it requires less frequency planning. The full original spectrum is not divided into separate blocks/channels, like it is in FDMA and TDMA systems. Therefore, there is no need to plan for multiple frequency guard bands. Because all users have access to the entire spectrum at all times, frequency planning only needs to consider one frequency/channel. However, the channel requires relatively wide contiguous bandwidth.

C. Advantages and Disadvantages

Advantages

- Greatest spectrum efficiency: capacity increases of 8 to 10 times that of an analog system and 4 to 5 times that of other digital systems which makes it most useful in high traffic areas with a large number of users and limited spectrum

- CDMA improves call quality by filtering out background noise, cross-talk, and interference
- ”Soft handoffs”— Because of the multiple diversities in use, handoffs between cells are undetected by the user
- Simplified frequency planning - all users on a CDMA system use the same radio frequency spectrum.
  - Engineering detailed frequency plans are not necessary.
  - Frequency re-tunes for expansion are eliminated.
  - Fewer cells are required for quality coverage
- Random Walsh codes enhance user privacy; a spread-spectrum advantage
- Precise power control increases talk time and battery size for mobile phones

Disadvantages

- Backwards compatibility techniques are costly
- Currently, base station equipment is expensive
- Difficult to optimize to maximize performance
- Low traffic areas lead to inefficient use of spectrum and equipment resources

1.6. SYSTEM DESIGN FACTORS – TECHNOLOGY COMPARISON

Using five different performance metrics, the following table shows the differences between each of the technologies. Each technology is rated based on their performance with respect to the ideal performance level.

	FDMA	TDMA	CDMA
Capacity (Spectrum Efficiency)	1	2	3
Security	1	2	3
Ease of Network Planning	1	2	3
Ease of Implementation	4	4	2
Cost of Implementation	3	3	2
Backwards Compatibility	4	3	1

Definitions for each performance metric is described below:
Capacity (Spectrum Efficiency): Measures the ability to handle heavy traffic.
Security: Measures the ability to keep information from being intercepted by others.

Ease of Network Planning: Measures the ease of creating and planning network structures.

Ease of Implementation: Measures the ease of carrying out a network system and equipment.

Cost of Implementation: Measures the financial requirements of carrying out a network system and equipment.

Backward Compatibility: Measures the ability to comply with existing systems.

CHAPTER 2

2.1Motivation:

Among all the competing wireless technologies currently under development, Wideband (WB) wireless communication is undoubtedly the most interesting. A WB communication system can be broadly classified as any communication system whose instantaneous bandwidth is much greater than the minimum required bandwidth to transmit particular information. The early WB technique was widely used in radar and remote sensing applications . In 2002, the US Federal Communication Commission (FCC) approved the commercial use of UWB technology for faster and more secure wireless transmissions and the Institute of Electrical and Electronics Engineers (IEEE) is currently working on a wireless WB standard.

There are two common types of WB communications: one is based on sending very short duration pulses to convey information and the other is based on using multiple carriers. This dissertation focuses on the first type of UWB, in which the UWB technique does not use sinusoidal carriers, but instead generates many extremely narrow pulses per second, resulting in an ultra wideband signal. Therefore, WB is alternatively referred to as carrier less short pulses or Impulse Radio (IR).

Because of the use of extremely short pulse duration signals, WB systems have several potentially attractive advantages:

(1) Multipath immunity is a natural outcome of using short pulses . Thus, the fading that
is commonly experienced in mobile applications is significantly reduced or eliminated.

(2) Due to the potentially large processing gain, a WB system is capable of supporting wireless communications at a high data rate.

(3) WB operates within the electronic "noise" portion of the spectrum as shown in Figure 1.1. Asystemwith ultra wide bandwidth and lowpower will be extremely difficult to intercept. This means that WB will be particularly useful for communication security.In addition, the low power spectral density allows coexistence with existing communication services.

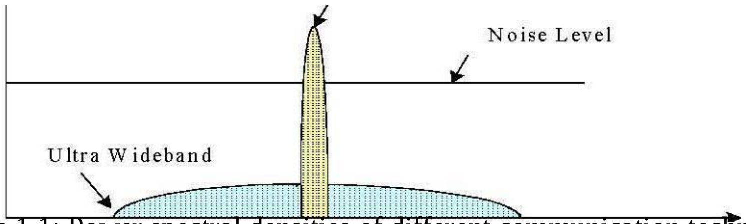

Figure 1.1: Power spectral densities of different communication techniques.

4) One ofthe most interesting benefits of WB technology is its ability to combine communications with radar applications. In a sense, a WB system uses radar like pulses for communications, and thus could be used for locating positions, detecting intrusions and remote sensing .
(5) Another benefit of UWB is its expected low cost. Unlike narrowband systems, the design of a UWB system requires relatively simple Radio Frequency (RF) electronics. This makes WB ideal for low cost production. WB is desirable for data transmission, especially for the distribution of music, video, and sensor information. Currently, target environments for using UWB techniques in communications are short-range systems, such as Wireless Local Area Networks (WLANs), Wireless Personal Area Networks (WPANs) and Home Area Networks (HANs). Also, the technology might be useful for small devices, like smart cellular phones, Personal Digital Assistants (PDAs), laptops, and much more . It is expected that the application of WB will continue to grow.

2.2 Problem Statement

Although WB technique provides new qualities and functionality, there are still several challenges in making this technology live up to its full potential. Issues of interest include:
(1) Propagation measurements and channel modeling . To perform system level engineering, the WB technique must consider its propagation characteristics. Multi path signals should be resolvable quickly to provide enough energy in order to recover the information signals. Thus, a valid statistical channel model is an important prerequisite for UWB system design and implementation. Proper analysis of the measured data will enable the development of a consistent channel model for system evaluation and simulation, in particular with respect to channel parameters such as the delay profiles, fading statistics and temporal and spatial correlations

(2) Multiple access modulation techniques for WB transmission . With high user densities, the performance of the network should be interference limited, and multi user interference cancellation is necessary. Thus achieving

better system performance necessitates the study of multiple access schemes and receiver designs.

(3) Signal shapes design. WB differs from conventional communications in that the signal shape might be changed due to the effect of channel fading and antenna radiation pattern. Therefore, the design of pulse shapes has received more and more attention.

(4) Timing acquisition and reception of ultra short pulses. The problem is mainly due to the low received signal power, which forces the acquisition system to process the signal over long periods of time before getting a reliable estimate of the timing of the signal. Hence there is a need to develop more efficient acquisition schemes by taking into account the signal and channel characteristics. In addition, due to the extremely short pulses, the need for highly accurate synchronization between transmitter and receiver must be considered.

(5) Coexistence of a WB system with other systems .WB radios, operating with extremely large bandwidths, must coexist with many other interfering narrowband signals. At the same time, these narrowband systems must not suffer intolerable interference from the WB radios. Thus, another concern is added to the design of the WB systems.

(6) The implementation issues of WB systems in network environments. Eventually, WB products must be applied to a network environment. The network should be self-organized and dynamic so that clients can join or leave at any time. Issues about both Physical (PHY) and Medium Access Control (MAC) layers must be addressed and related protocols should be designed .In this dissertation, UWB multiple access techniques are studied over a multi path indoor wireless channel, thus addressing item (2) for application purposes .Multi path propagation occurs when an RF signal takes different paths from a transmitter to a receiver due to reflection, refraction and scattering. A portion of the signal may go directly to the destination, and some of the signal will encounter delays and travel longer paths to the receiver. This can result in Inter-Symbol Interference (ISI), which is a critical phenomenon in short-range indoor environments. For the signal to survive in a fading environment, one solution is to increase the transmitted power. However, this is not a viable solution due to the limitation on power mandated by law. In cellular systems,
Another solution might be the use of RAKE receivers and diversity techniques, which provide a more practical and feasible solution for performance improvement. This work characterizes several kinds of diversity techniques for WB systems .The multiple access capacity is one of

the most important considerations in any multi user wireless communication system. Modulation techniques of IR systems have been well studied in radar systems for decades . However, for WB radios, modulation and
coding schemes need to be developed. To achieve better multiple access capability and highly resolved multi path, Spread Spectrum (SS) communication techniques are employed in cellular communications, including Time Hopping (TH), Frequency Hopping (FH) and
Direct Sequence (DS) . This dissertation investigates SS techniques for applications in WB systems. The various modulation schemes for currently existing WB systems are also presented along with some new systems proposed in this dissertation. In addition, the system performance in a multi path and multi user fading channel is evaluated.

2.3 Importance and Contribution of Research

The contribution of this research can be summarized as follows:

(1) Analysis of WB systems with a generalized channel model. In this dissertation, the propagation channel is modeled as a multipath channel, which is known to provide a close match to experimental data . Moreover, the Nakagami distribution is generalized since it can approximate several fading conditions, such as Gaussian, Rayleigh and Ricean distributions.

(2) Application of three different diversity combiners in WB systems. The diversity techniques considered include Selection Diversity (SD), Maximal Ratio Combining (MRC) and Equal Gain Combining (EGC).

(3) Presentation of a new multiple access scheme for WB communication systems. Beside Pulse Position Modulation (PPM), two more pulse modulation techniques for Time Hopping (TH) multiple access WB systems, Pulse Amplitude Modulation (PAM) and Pulse Shift Keying (PSK), are applied. The Direct Sequence-Time Hopping (DS-TH) multiple access scheme with the relative receiver design is proposed and analyzed in this dissertation.

(4) Numerical comparison among existing and proposed multiple access schemes, such as Time Hopping-Pulse Position Modulation (TH-PPM), Time Hopping-Pulse Shift Keying (TH-PSK), Direct Sequence-Pulse Shift Keying (DS-PSK) and DS-TH. The system performance of each scheme is analyzed under a multipath fading environment.

2.4 Dissertation Outline

In Chapter II, the fundamentals of WB communications are described. Several signal shapes are expressed in both time and frequency domains. This is followed by the introduction of pulse modulation techniques, and then the proposed system model, including the transmitter, channel and receiver model. The expressions of UWB system performance parameters are also defined, namely Signal-to-Noise plus Interference Ratio (SNIR), Bit Error Rate (BER), and outage probability. Finally, the introduction of diversity combiners is presented. From Chapter III to Chapter V, each chapter covers one proposed multiple access schemes. The review of each scheme is presented first. Then basic modulation/demodulation techniques and mathematical expressions are given for performance analysis. Analytical expressions for different diversity combiners and signal waveforms are deduced. Finally, the average SNIR, BER and outage probability performances are plotted as the function of the data transmission rate, the number of active users, the multipath components, the fading parameters and the signal power. Finally, Chapter VI concludes this dissertation with a summary of the work completed and a list of suggested future research topics.

CHAPTER 3

WB COMMUNICATIONS SYSTEM:

3.1 Definition of WB Signals

A WB signal is defined as any signal whose fractional bandwidth is greater than 25% or occupies at least 500 MHz in the allocated spectrum of the 3.1-10.6 GHz band .The fractional bandwidth as defined by the FCC is given by

$$B_\Delta = \frac{2\,(f_H - f_L)}{f_H + f_L},$$

where fH and fL are the upper and lower frequencies of the -10 dB emission point respectively. Furthermore, the center frequency of the transmission could be defined as the average of the upper and lower -10 dB points, i.e.,

fc = (fH + fL) /2.

In order to protect both UWB communication systems and existing services against interference, emission limits were specified for UWB devices. For indoor communication systems, the Equivalent Isotropic Radiated Power (EIRP) density must remain below -41.3 dBm/MHz, as shown in Figure 2.1.

It is noticed that this regulation is based on a frequency band with power limitations, not on the type of data modulation and multiple access schemes. This brings more flexibility in the use of UWB techniques in wireless communication systems.

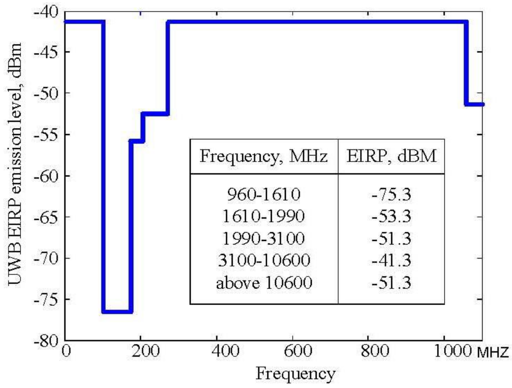

Figure 2.1 FCC emission specification for indoor applications.

3.2 WB Pulse Shapes:

In WB communications, information is contained in trains of extremely short pulses, typically from a few tens of picoseconds to a few nanoseconds .Since WB systems use the baseband technology, the choice of pulse shapes will affect the system performance. Several possible pulse shapes, with the signal amplitude A, are discussed in this section.

(a) Rectangular Pulse Shape

The simple rectangular shaped signal is shown in Figure 2.2 (a), which is mathematically defined as

$$p(t) = \begin{cases} A, & -\frac{T_p}{2} \leq t \leq \frac{T_p}{2} \\ 0 & \text{elsewhere,} \end{cases}$$

where Tp is the pulse width in picoseconds and the pulse p(t) is centered at t = 0. For a sinc function, the main lobe is between f = −1/Tp and f = 1/Tp, and there are energy lobes outside.

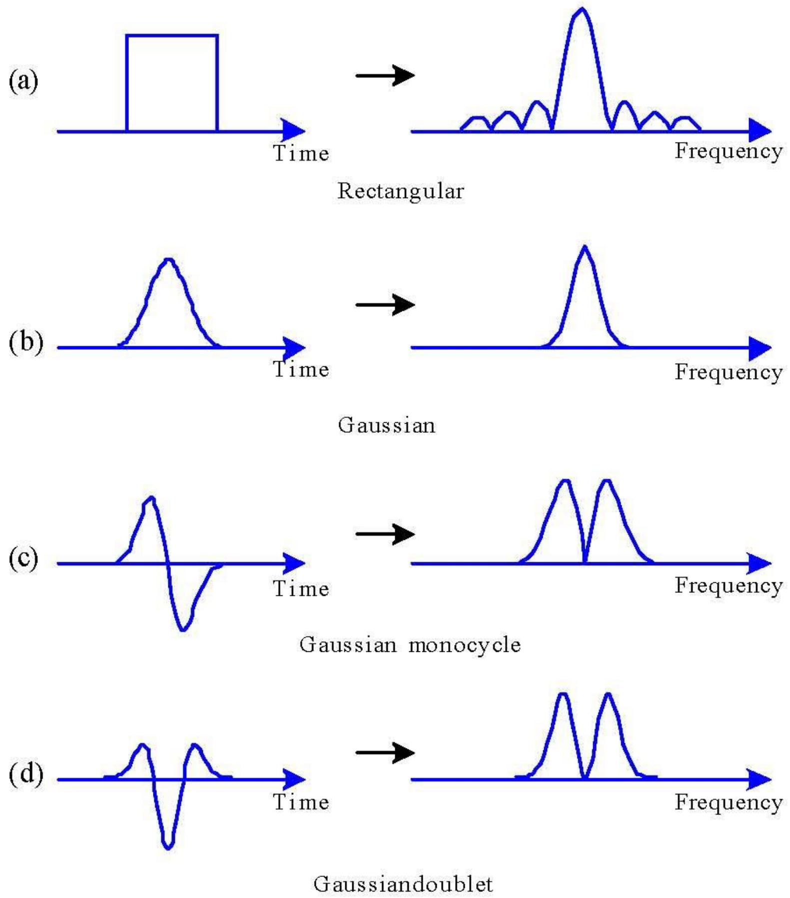

Figure 2.2: UWB pulses in both time and frequency domains.

b) Gaussian Pulse Shape

A Gaussian pulse, as shown in Figure 2.2 (b), can be represented by

$$p(t) = A \exp\left(-\frac{t}{t_m}\right)^2,$$

where tm is a time-scaling factor, which determines the width of the pulse.

(c) Gaussian Monocycle Pulse Shape:

The first derivative of a Gaussian pulse, called a Gaussian monocycle, is also commonly used, as shown in Figure 2.2 (c). A Gaussian monocycle pulse is described as

$$p(t) = -\frac{2At}{t_m^2} \exp\left(-\frac{t}{t_m}\right)^2$$

(d) Gaussian Doublet Pulse Shape:

The second derivative of a Gaussian signal is called a Gaussian doublet, as shown in Figure 2.2 (d). In the time domain, it is defined by

$$p(t) = \frac{2A}{t_m^2}\left(1 - \frac{2t^2}{t_m^2}\right) \exp\left(-\frac{t}{t_m}\right)^2,$$

The rectangular and Gaussian pulses can be compared in terms of their design parameters by making the 10 dB bandwidths in the frequency domain exactly equal. Clearly, the main lobes between 10 dB points are nearly identical, but the side lobes are quite different. Gaussian pulses have the lower side lobe energy, but rectangular pulses show the opposite.

Even though a rectangular pulse contains more energy outside of the main lobe, it is still one of the popular shapes for the system performance analysis due to its simplicity. It must be pointed out that many more different pulse shapes may be used for WB applications. The choice of pulse shapes is usually driven by the system design and application requirements. Gaussian pulse shapes are chosen since they are relatively easy to generate. The most important factor to use the Gaussian waveform is the effect of filtering due to antennas at both transmitters and receivers. After being filtered by the transmitter antenna, the output signal is the first derivative of the input signal, while the same effect occurs at the receiver . Therefore, a Gaussian doublet is always considered as the received pulse shape if a Gaussian pulse is used at the input of the transmitter antenna.

3.3 Pulse Modulation:

One pulse by itself does not contain a lot of information. User data needs to be modulated onto a sequence of pulses, or a pulse train. In WB systems, information can be encoded in a variety of ways, including amplitude, time and phase modulation. Three popular modulation schemes used for WB communications are Pulse Position Modulation (PPM), Pulse Amplitude Modulation (PAM) and Phase Shift Keying (PSK). These modulation techniques are briefly discussed.

3.3.1 Pulse Position Modulation:

Pulse Position Modulation (PPM) is based on the principle of encoding information with two or more positions in time, with references to the nominal pulse position. For example, in a binary system, as shown in Figure 2.3, a pulse transmitted at the nominal position represents a bit 0, and a pulse transmitted after the nominal position represents a bit 1. This is achieved by time shifting, which is typically a fraction of a nanosecond and less than the time duration between nominal positions to avoid interference between impulses.

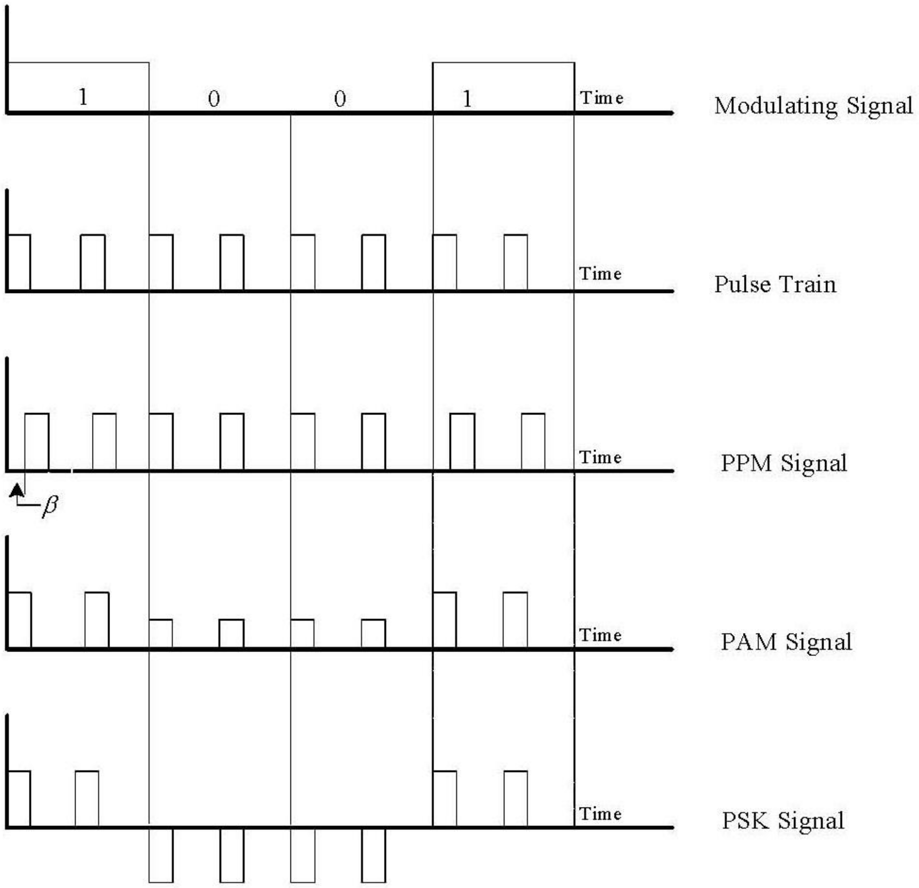

Figure 2.3 Illustration of pulse modulation techniques

3.3.2 Pulse Amplitude Modulation:

Pulse Amplitude Modulation (PAM) is based on the principle that the amplitude of the pulses is encoded by data. Digital PAM is also called Amplitude Shift Keying (ASK), alternatively referred to as On-Off Keying (OOK) for two level PAM. Figure 2.3 shows a two-level amplitude modulation. Bit 1s and bit 0s are transmitted by different amplitudes.

3.3.3 Pulse Shift Keying:

Pulse Shift Keying (PSK) is based on modulating information by phase shifting. The Binary PSK is also referred to as bi-phase modulation. In this case, the modulated signal shifts the phase of pulses, for example, at zero degrees for transmitting bit 1s and 180 degrees for transmitting bit 0s.

Among the three pulse modulation techniques, PPM and PSK could be used in WB systems. The advantages of PPM mainly arise from its simplicity and the ease with which the time shift may be controlled [14]. Fine time control is necessary for UWB systems due to the extremely short pulses. The benefit of using PSK is its power efficiency and reduced jitter requirements. PAM might not be the preferred method for most short-range communications due to the extremely low signal power.

3.4 Spread Spectrum Techniques

It is commonly believed that UWB communication systems will be required for future generation wireless communication systems. Thus, the successful deployment of the WB technology strongly depends on efficient multiple access techniques. In cellular communication systems, spread spectrum techniques gain their popularity in dense Multiple Access Interference (MAI) propagation channels. Spread spectrum (SS) is an RF communication modulation technique in which the base band signal bandwidth is intentionally spread over a larger bandwidth and the spread signal is composed of the information signal and the spreading sequence. In contrast to narrowband communications, spread spectrum techniques are able to communicate through environments of severe interference. The robustness of SS in interference-prone environments makes the technology suitable for both military and commercial applications. Different spread spectrum techniques are available. Figure 2.4 shows a classification of spread spectrum techniques; namely, Direct Sequence Spread Spectrum (DSSS), Frequency

Hopping Spread Spectrum (FHSS), Time Hopping Spread Spectrum (THSS) and the hybrid spread spectrum.

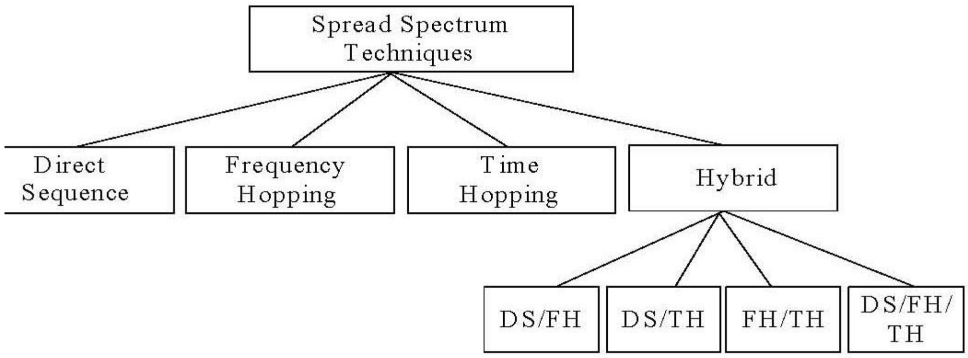

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Figure 2.4: Classification of different spread spectrum systems.

(a) Direct Sequence Spread Spectrum:

Direct Sequence Spread Spectrum (DSSS) spreads the energy of the signal over a large bandwidth. Thus, the energy per unit frequency is reduced and the interference is decreased. This allows multiple signals to share the same frequency band. To an unintended receiver, DSSS signals appear as low-power wideband noise and are rejected. DSSS combines the data stream with a high-speed digital code. Each data bit is mapped into a common pattern of bits (also called chips), named a Pseudo-random Noise (PN) code , and known only to the transmitter and the corresponding receiver. A key parameter for DSSS systems is the number of chips per bit, called the processing gain. If there is an interference or jammer in the same band, it will be spread out by a factor equivalent to the processing gain. The chipping code is a redundant bit pattern for each bit that is transmitted, which increases the signal's resistance to interference. A key feature of DSSS is that multiple access capability can be achieved without synchronization between different transmitters. But the transmitter and receiver must be synchronized to operate properly.

(b) Time Hopping Spread Spectrum:

Time Hopping Spread Spectrum (THSS) modulates a signal by using a PN sequence to control the time of a transmission. Within a THSS system, the time axis is virtually divided into frames and slots within the frame. Only one slot is used out of n possible slots within one frame for a single user. Thus, the sending data rate is n times higher in contrast to the situation where the whole frame is used. MAI can be minimized if proper coordination between transmitters can be achieved. If more than one transmitter uses the same time slot, the receivers will not be able to detect either of the signals correctly. For such cases, error correction schemes must be applied.

(c) Frequency Hopping Spread Spectrum:

Frequency Hopping Spread Spectrum (FHSS) spreads the signal by using a different carrier frequency at different times. The change of frequency is based on a pattern known to both the transmitter and the receiver and the process of changing the carrier from one frequency to another (i.e., a hopping) is facilitated by the PN sequence. During a small amount of time, the signal at the given frequency is constant. However, the sequence of different channels, which determines the hopping pattern, is pseudo random. Since it is difficult to predict the next frequency at which a system will transmit data, FHSS system is secure against interception. A comparison of the three basic spread spectrum techniques shows that FHSS and THSS do not spread the signal directly unlike DSSS, but they use spreading code sequences to determine the hopping sequence or the timing sequence. Furthermore, hopping is usually more costly and more complicated because it needs extra circuits for synchronizing in both time and frequency. Between the two hopping schemes, the application of THSS is much simpler than that of FHSS. FHSS has the shortest acquisition time among these techniques, THSS has the highest bandwidth efficiency, and DSSS shows most effective throughput in the time domain. Obviously, combining two or more pure spread spectrum techniques can offer a combination of their advantages, but it may increase the complexity of the transmitter and receiver. In this research, the DS-TH technique is applied to UWB communication systems. This is a hybrid technique combining the techniques of DSSS and THSS.

3.5 Generalized WB System Model

The block diagram of a generalized WB system is illustrated in Figure 2.5. Throughout this dissertation, it is considered that multiple users' WB signals are simultaneously transmitted through a multipath fading channel. At the receiver, the desired signal is corrupted by MAI and noise.

3.5.1 Transmitter Model

In the system model, K simultaneous users transmit WB signals through a multipath channel. User binary data can be from any application, such as an e-mail client, a web browser, a personal digital device or a digital stream from a DVD player.

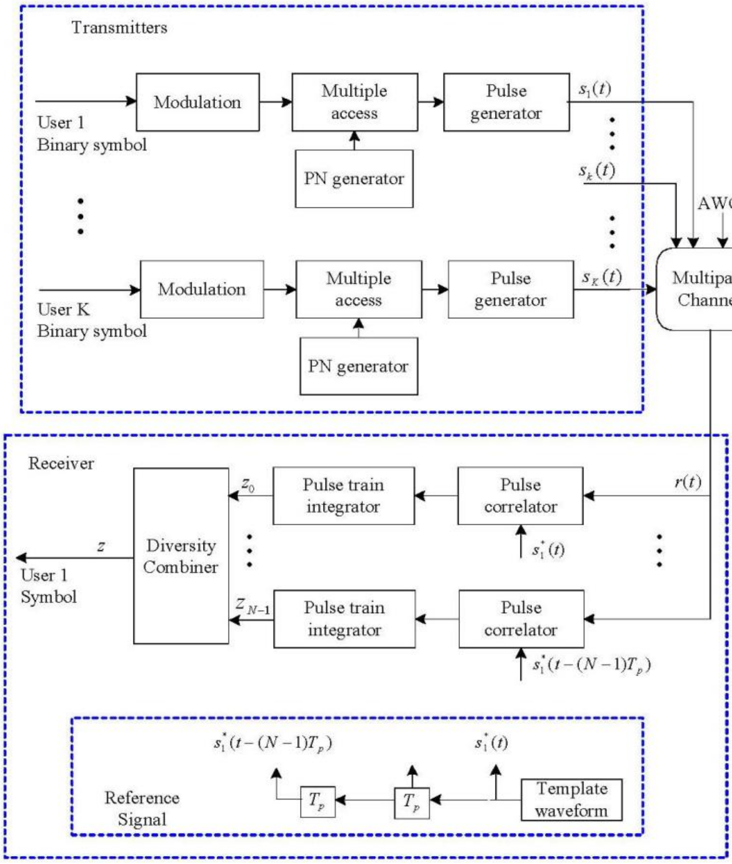


Figure 2.5: Block diagram of UWB communication system model.

After coding, the binary data are then mapped from bits to symbols. Our concern is that of mapping symbols to analog pulse shapes, and then generating and transmitting pulses through the antenna. This requires precise timing circuits and optional amplifier circuits, although they are not shown in Figure 2.5. Basically, each transmitter generates a pulse train modulated by the data stream. In this case, multiuser information transmissions can be achieved by pulse modulation and SS techniques.

In general, the transmitted symbol of the kth user can be expressed as

$$s_k(t) = \sqrt{\frac{E_s}{N_s}} \sum_{j=0}^{N_s-1} F(j,k) p_{tr} \left[t - t_k - G(j,k) \right],$$

where Ns is the number of pulses per symbol; Es is the symbol energy; t is the first user's transmitter clock time; tk is the relative transmission time difference compared to the first user with t1 = 0; and ptr represents the transmitted waveform based on the particular pulse shape. In addition, F(j, k) and G(j, k) are used to generalize modulation functions, which are varied from each multiple access scheme and will be derived in the following chapters.

For simplicity, we define

$$Ptrjk(t) = F(j, k)ptr\,[t - tk - G(j, k)]\ ,$$

Multiplexing:

In telecommunications and computer networks, multiplexing (known as muxing) is a term used to refer to a process where multiple analog message signals or digital data streams are combined into one signal over a shared medium. The aim is to share an expensive resource. For example, in telecommunications, several phone calls may be transferred using one wire.

The multiplexed signal is transmitted over a communication channel, which may be a physical transmission medium. The multiplexing divides the capacity of the low-level communication channel into several higher-level logical channels, one for each message signal or data stream to be transferred. A reverse process, known as demultiplexing, can extract the original channels on the receiver side.

A device that performs the multiplexing is called a multiplexer (MUX), and a device that performs the reverse process is called a demultiplexer (DEMUX).

Inverse multiplexing (IMUX) has the opposite aim as multiplexing, namely to break one data stream into several streams, transfer them simultaneously over several communication channels, and recreate the original data stream.

Categories of multiplexing:

The two most basic forms of multiplexing are time-division multiplexing (TDM) and frequency-division multiplexing (FDM), both either in analog or digital form. FDM requires modulation of each signal.

In optical communications, FDM is referred to as wavelength-division multiplexing (WDM).

Variable bit rate digital bit streams may be transferred efficiently over a fixed bandwidth channel by means of statistical multiplexing, for example packet mode communication. Packet mode communication is an asynchronous mode time-domain multiplexing, which resembles but should not be considered as time-division multiplexing.

Digital bit streams can be transferred over an analog channel by means of code-division multiplexing (CDM) techniques such as frequency-hopping spread spectrum (FHSS) and direct-sequence spread spectrum (DSSS).

In wireless communications, multiplexing can also be accomplished through alternating polarization (horizontal/vertical or clockwise/counterclockwise) on each adjacent channel and satellite, or through phased multi-antenna array combined with a Multiple-input multiple-output communications (MIMO) scheme.

What is multiplexing why is multiplexing needed in data communication?

In electronics, telecommunications and computer networks, multiplexing (short muxing) is a term used to refer to a process where multiple analog message signals or digital data streams are combined into one signal over a shared medium. The aim is to share an expensive resource. For example, in electronics, multiplexing allows several analog signals to be processed by one analog-to-digital converter (ADC), and in telecommunications, several phone calls may be transferred using one wire. In communications, the multiplexed signal is transmitted over a communication channel, which may be a physical transmission medium. The multiplexing divides the capacity of the low-level communication channel into several higher-level logical channels, one for each message signal or data stream to be transferred. A reverse process, known as demultiplexing, can extract the original channels on the receiver side.

Multiplexing technique is designed to reduce the number of electrical connections or leads in the display matrix. Whereas driving signals are applied not to each pixel (picture element) individually but to a group of rows and columns at a time. Besides reducing the number of individually independent interconnections, multiplexing also simplifies the drive electronics, reduces the cost and provides direct interface with the microprocessors. There are limitations in multiplexing due to complex electro-optical response of the liquid crystal cell. However, fairly reasonable level of multiplexing can be achieved by properly choosing the multiplexing scheme, liquid crystal mixture and cell designing.

Digital Multiplex System:

Figure: A DMS-10 used as a lab trainer at Southeast Missouri State University.

Digital Multiplex System (DMS) is the name shared among several different telephony product lines from Nortel Networks for wire line and wireless operators. Among them are the DMS-1 (originally named the DMS-256[1]) Rural/Urban digital loop carrier, DMS-10 telephone switch, the DMS Super Node family of telephone switches (DMS-100, DMS-200, DMS-250, DMS-300, DMS-500, DMS-GSP, DMS-MSC, DMS-MTX), and the S/DMS optical transmission system.

Exploratory development on the technology began at Northern Telecom's Bell Northern Research Labs (Ottawa, Canada) in 1971. The first Class 5, the DMS-10 switch, began service on 21 October 1977 in Fort White, Florida and the first toll switch, the DMS-200, entered service in 1979 in Ottawa, Canada. DMS was the first commercially successful Class 5 digital switch in the North American market and revolutionized the industry. Of the numerous digital switching products introduced in the North American telephone market in the late 1970s, only the Nortel DMS Family is still in production.

Previously, new technology had entered the telecommunications industry slowly, with the telephone companies amortizing equipment over periods as long as forty years. AT&T was intending to delay the introduction of digital switching until the 1990s. The success of DMS changed the industry by creating a technological imperative that has lasted until this day. DMS, with its massive introduction of digital technology, was one of the antecedents that encouraged the Internet to grow large.

On October 16, 2006, Nortel received a special recognition award from Canada's Telecommunications Hall of Fame for its role in pioneering digital communications with the Digital Multiplex System.

Figure: A DMS-250 used by an operator to offer local and long distance services in France.

The DMS name arose from a designation for a switching matrix design that was developed in the exploratory phase of the project. The Digital Multiplexed Switch was selected as the basic switching design for the project. The product was intended as a successor for Nortel's first electronic switch the SP1 and was to be called the SP2. However the DMS acronym was mellifluous and was eventually (1975) adopted as the designation for the DMS-10 and DMS-100 family of products with the 'S' standing for 'system' rather than 'switch.' It was then applied to the entire digital switching family as well as the DMS-1 family of Digital Transmission Concentrators.

DMS is favored by many European cable operators as the switching platform for their voice networks. The DMS-10 is widely used by rural wire line providers. DMS-100 and 200 switches are widely deployed throughout the U.S. and Canada by Regional Bell Operating Companies, Bell Canada and independent telephone companies as well as the US Military. The DMS-250 is the backbone of a number of carriers long distance networks, the DMS-300 is an international gateway switch and the DMS-500 is commonly deployed by competitive local exchange carriers because it combines DMS-100 and 250 capabilities.

Digital multiplex hierarchy:

In telecommunications, a digital multiplex hierarchy is a hierarchy consisting of an ordered repetition of tandem digital multiplexers that produce signals of successively higher data rates at each level of the hierarchy.

Digital multiplexing hierarchies may be implemented in many different configurations depending on;

(a)The number of channels desired,
(b)The signaling system to be used, and
(c)The bit rate allowed by the communications media.

Some currently available digital multiplexers have been designated as Dl-, DS-, or M-series, all of which operate at T-carrier rates.

In the design of digital multiplex hierarchies, care must be exercised to ensure interoperability of the multiplexers used in the hierarchy.

Orthogonal frequency-division multiplexing (OFDM)

essentially identical to Coded OFDM (COFDM) and Discrete multi-tone modulation (DMT) — is a frequency-division multiplexing (FDM) scheme utilized as a digital multi-carrier modulation method. A large number of closely-spaced orthogonal sub-carriers are used to carry data. The data are divided into several parallel data streams or channels, one for each sub-carrier. Each sub-carrier is modulated with a conventional modulation scheme (such as quadrature amplitude modulation or phase shift keying) at a low symbol rate, maintaining total data rates similar to conventional single-carrier modulation schemes in the same bandwidth.

OFDM has developed into a popular scheme for wideband digital communication, whether wireless or over copper wires, used in applications such as digital television and audio broadcasting, wireless networking and broadband internet access.

The primary advantage of OFDM over single-carrier schemes is its ability to cope with severe channel conditions — for example, attenuation of high frequencies in a long copper wire, narrowband interference and frequency-selective fading due to multipath — without complex equalization filters. Channel equalization is simplified because OFDM may be viewed as using many slowly-modulated narrowband signals rather than one rapidly-modulated wideband signal. The low symbol rate makes the use of a guard interval between symbols affordable, making it possible to handle time-spreading and eliminate intersymbol interference (ISI). This mechanism also facilitates the design of single-frequency networks, where several adjacent transmitters send the same signal simultaneously at the same frequency, as the signals from multiple distant transmitters may be combined constructively, rather than interfering as would typically occur in a traditional single-carrier system.

Advantages

- Can easily adapt to severe channel conditions without complex equalization
- Robust against narrow-band co-channel interference
- Robust against Intersymbol interference (ISI) and fading caused by multipath propagation

- High spectral efficiency
- Efficient implementation using FFT
- Low sensitivity to time synchronization errors
- Tuned sub-channel receiver filters are not required (unlike conventional FDM)
- Facilitates Single Frequency Networks, i.e. transmitter macro diversity.

Disadvantages

- Sensitive to Doppler shift.
- Sensitive to frequency synchronization problems.
- High peak-to-average-power ratio (PAPR), requiring linear transmitter circuitry, which suffers from poor power efficiency

Channel coding and interleaving

OFDM is invariably used in conjunction with channel coding (forward error correction), and almost always uses frequency and/or time interleaving.
Frequency (sub carrier) interleaving increases resistance to frequency-selective channel conditions such as fading. For example, when a part of the channel bandwidth is faded, frequency interleaving ensures that the bit errors that would result from those sub carriers in the faded part of the bandwidth are spread out in the bit-stream rather than being concentrated. Similarly, time interleaving ensures that bits that are originally close together in the bit-stream are transmitted far apart in time, thus mitigating against severe fading as would happen when traveling at high speed.
However, time interleaving is of little benefit in slowly fading channels, such as for stationary reception, and frequency interleaving offers little to no benefit for narrowband channels that suffer from flat-fading (where the whole channel bandwidth is faded at the same time).
The reason why interleaving is used on OFDM is to attempt to spread the errors out in the bit-stream that is presented to the error correction decoder, because when such decoders are presented with a high concentration of errors the decoder is unable to correct all the bit errors, and a burst of uncorrected errors occurs.
A common type of error correction coding used with OFDM-based systems is convolution coding, which is often concatenated with Reed-Solomon coding. Convolution coding is used as the inner code and Reed-Solomon coding is used for the outer code — usually with additional interleaving (on top of the time and frequency interleaving mentioned above) in between the two layers of coding. The reason why this combination of error correction coding is used is that the Viterbi decoder used for convolution decoding produces short errors bursts when there is a high concentration of errors,

and Reed-Solomon codes are inherently well-suited to correcting bursts of errors.
Newer systems, however, usually now adopt the near-optimal types of error correction coding that use the turbo decoding principle, where the decoder iterates towards the desired solution. Examples of such error correction coding types include turbo codes and LDPC codes. These codes only perform close to the Shannon limit for the Additive White Gaussian Noise (AWGN) channel, however, and some systems that have adopted these codes have concatenated them with either Reed-Solomon (for example on the Media FLO system) or BCH codes (on the DVB-S2 system) to improve performance further over the wireless channel.

Application areas:

Telegraphy:
The earliest communication technology using electrical wires, and therefore sharing an interest in the economies afforded by multiplexing, was the electric telegraph. Early experiments allowed two separate messages to travel in opposite directions simultaneously, first using an electric battery at both ends, then at only one end. In 1874, the quadruplex telegraph developed by Thomas Edison transmitted two messages in each direction simultaneously, for a total of four messages transiting the same wire at the same time. Alexander Graham Bell was investigating a frequency-division multiplexing technique with his "Harmonic Telegraph", which led to the telephone.

Telephony:
In telephony, a customer's telephone line now typically ends at the remote concentrator box down the street, where it is multiplexed along with the telephone lines for that neighborhood or other similar area. The multiplexed signal is then carried to the central switching office on significantly fewer wires and for much further distances than a customer's line can practically go. This is likewise also true for digital subscriber lines (DSL).
Fiber in the loop (FITL) is a common method of multiplexing, which uses optical fiber as the backbone. It not only connects POTS phone lines with the rest of the PSTN, but also replaces DSL by connecting directly to Ethernet wired into the home. Asynchronous Transfer Mode is often the communications protocol used.
Because all of the phone (and data) lines have been clumped together, none of them can be accessed except through a demultiplexer. This provides for more-secure communications, though they are not typically encrypted.
The concept is also now used in cable TV, which is increasingly offering the same services as telephone companies. IPTV also depends on multiplexing.

Video processing:

In video editing and processing systems, multiplexing refers to the process of interleaving audio and video into one coherent transport stream (time-division multiplexing).

In digital video, such a transport stream is normally a feature of a container format which may include metadata and other information, such as subtitles. The audio and video streams may have variable bit rate. Software that produces such a transport stream and/or container is commonly called a statistical multiplexor or muxer. A demuxer is software that extracts or otherwise makes available for separate processing the components of such a stream or container.

Digital broadcasting:

In digital television and digital radio systems, several variable bit-rate data streams are multiplexed together to a fixed bit rate transport stream by means of statistical multiplexing. This makes it possible to transfer several video and audio channels simultaneously over the same frequency channel, together with various services.

In the digital television systems, this may involve several standard definition television (SDTV) programmes (particularly on DVB-T, DVB-S2, and ATSC-C), or one HDTV, possibly with a single SDTV companion channel over one 6 to 8 MHz-wide TV channel. The device that accomplishes this is called a statistical multiplexer. In several of these systems, the multiplexing results in an MPEG transport stream.

On communications satellites which carry broadcast television networks and radio networks, this is known as multiple channel per carrier or MCPC. Where multiplexing is not practical (such as where there are different sources using a single transponder), single channel per carrier mode is used.

In digital radio, both the Eureka 147 system of digital audio broadcasting and the in-band on-channel HD Radio, FMeXtra, and Digital Radio Mondiale systems can multiplex channels. This is essentially required with DAB-type transmissions (where a multiplex is called an ensemble), but is entirely optional with IBOC systems.

Analog broadcasting:

In FM broadcasting and other analog radio media, multiplexing is a term commonly given to the process of adding sub carriers to the audio signal before it enters the transmitter, where modulation occurs. Multiplexing in this sense is sometimes known as MPX, which in turn is also an old term for stereophonic FM, often seen on stereo systems of the 1960s and 1970s.

Other meanings: In spectroscopy the term is used in a related sense to indicate that the experiment is performed with a mixture of frequencies at once and their respective response unraveled afterwards using the Fourier transform principle.

Multiplexing may also refer to a juggling technique where multiple objects are released from one hand at the same time.
In computer programming, it may refer to using a single in-memory resource (such as a file handle) to handle multiple external resources (such as on-disk files).
Some electrical multiplexing techniques do not require a physical "multiplexer" device; they refer to a "keyboard matrix" or "Charlieplexing" design style:
Multiplexing may refer to the design of a multiplexed display (non-multiplexed displays are immune to the Dorito effect).
Multiplexing may refer to the design of a "switch matrix" (non-multiplexed buttons are immune to "phantom keys" and also immune to "phantom key blocking").

Chapter4

SPREAD SPRECTRUM

SPREAD SPRECTRUM (SS)

Spread-spectrum techniques are methods by which energy generated at one or more discrete frequencies is deliberately spread or distributed in time or frequency domains. This is done for a variety of reasons, including the establishment of secure communications, increasing resistance to natural interference and jamming, and to prevent detection.

4.1 Spread-spectrum telecommunications

This is a technique in which a (telecommunication) signal is transmitted on a bandwidth considerably larger than the frequency content of the original information.

Spread-spectrum telecommunications is a signal structuring technique that employs direct sequence, frequency hopping or a hybrid of these, which can be used for multiple access and/or multiple functions. This technique decreases the potential interference to other receivers while achieving privacy. Spread spectrum generally makes use of a sequential noise-like signal structure to spread the normally narrowband information signal over a relatively wideband (radio) band of frequencies. The receiver correlates the received signals to retrieve the original information signal. Originally there were two motivations: either to resist enemy efforts to jam the communications (anti-jam, or AJ), or to hide the fact that communication was even taking place, sometimes called low probability of intercept (LPI).

Frequency-hopping spread spectrum (FHSS), direct-sequence spread spectrum (DSSS), time-hopping spread spectrum (THSS), chirp spread spectrum (CSS), and combinations of these techniques are forms of spread spectrum. Each of these techniques employ pseudorandom number sequences — created using pseudorandom number generators — to determine *and* control the spreading pattern of the signal across the allotted bandwidth. Ultra-wideband (UWB) is another modulation technique that accomplishes the same purpose, based on transmitting short duration pulses. Wireless Ethernet standard IEEE 802.11 uses either FHSS or DSSS in its radio interface.

Notes

- "Spread" radio signals over a wide frequency range several magnitudes higher than minimum requirement. The core principle of spread spectrum is the use of noise-like carrier waves, and, as the name implies, bandwidths much wider than that required for simple point-to-point communication at the same data rate.
- Two main techniques:
  - Direct sequence (DS)
  - Frequency hopping (FH)
- Resistance to jamming (interference). DS is better at resisting continuous-time narrowband jamming, while FH is better at resisting pulse jamming. In DS systems, narrowband jamming affects detection performance about as much as if the amount of jamming power is spread over the whole signal bandwidth, when it will often not be much stronger than background noise. By contrast, in narrowband systems where the signal bandwidth is low, the received signal quality will be severely lowered if the jamming power happens to be concentrated on the signal bandwidth.
- Resistance to eavesdropping. The spreading code (in DS systems) or the frequency-hopping pattern (in FH systems) is often unknown by anyone for whom the signal is unintended, in which case it "encrypts" the signal and prevents the adversary from making sense of it. What's more, for a given noise power spectral density (PSD), spread-spectrum systems require the same amount of energy per bit before spreading as narrowband systems and therefore the same amount of power if the bit rate before spreading is the same, but since the signal power is spread over a large bandwidth, the signal PSD is much lower, often significantly lower than the noise PSD, therefore the adversary may be unable to determine if the signal exists at all. Note that these effects can also be achieved by using encryption and a very low-rate channel code, which can also be viewed as a spread-spectrum method, albeit more complex.
- Resistance to fading. The high bandwidth occupied by spread-spectrum signals offer some frequency diversity, i.e. it is unlikely that the signal would encounter severe multipath fading over its whole bandwidth, and in other cases the signal can be detected using e.g. a Rake receiver.

- Multiple access capability. Multiple users can transmit simultaneously on the same frequency (range) as long as they use different spreading codes. See CDMA.

4.2 Pseudorandom Sequence

A Pseudorandom number sequence is a sequence of numbers that has been computed by some defined arithmetic process but is effectively a random number sequence for the purpose for which it is required.

Although a pseudorandom number sequence in this sense often appears to lack any definite pattern, any pseudorandom number generator with a finite internal state will repeat after a very long sequence of numbers. This can be proved using the pigeonhole principle.

Pigeonhole principle

The inspiration for the name of the principle: pigeons in holes. Here $n = 7$ and $m = 9$.

The pigeonhole principle, also known as Dirichlet's box (or drawer) principle, states that, given two natural numbers n and m with $n > m$, if n items are put into m pigeonholes, then at least one pigeonhole must contain more than one item. Another way of stating this would be that m holes can hold at most m objects with one object to a hole; adding another object will force you to reuse one of the holes (provided that m is finite; otherwise, see below on infinite sets). More formally, the theorem states that there does not exist an injective function on finite sets whose co-domain is smaller than its domain.

The pigeonhole principle is an example of a counting argument which can be applied to many formal problems, including ones involving infinite sets that cannot be put into one-to-one correspondence. In Diophantine approximation the quantitative application of the principle to the existence of integer solutions of a system of linear equations goes under the name of *Siegel's lemma*.

The first statement of the principle is believed to have been made by Dirichlet in 1834 under the name *Schubfachprinzip* ("drawer principle" or "shelf principle"). In Italian too, the original name "principio dei cassetti" is still in use; in some other languages (for example, Russian) this principle is called the *Dirichlet principle* (not to be confused with the minimum principle for harmonic functions of the same name).

Examples

An easy example of the pigeonhole principle involves the situation when there are five people who want to play softball (n = 5 objects), but there are only four softball teams (m = 4 holes). This would not be a problem except that each of the five refuses to play on a team with any of the other four. To prove that there is no way for all five people to play softball, the pigeonhole principle says that it is impossible to divide five people among four teams without putting two of the people on the same team. Since they refuse to play on the same team, at most four of the people will be able to play.

Although the pigeonhole principle may seem to be a trivial observation, it can be used to demonstrate possibly unexpected results. For example, *there must be at least two people in London with the same number of hairs on their heads*. Demonstration: a typical head of hair has around 150,000 hairs. It is reasonable to assume that no one has more than 1,000,000 hairs on their head (m = 1 million holes). There are more than 1,000,000 people in London (n is bigger than 1 million objects). If we assign a pigeonhole for each number of hairs on a head, and assign people to the pigeonhole with their number of hairs on it, there must be at least two people with the same number of hairs on their heads.

Another example is: *Presume that in a box there are 10 black socks and 12 blue socks and you need to get one pair of socks of the same colour. Supposing you can take socks out of the box only once and only without looking, how many socks do you have to pull out together?* When asked point-blank, people may sometimes unthinkingly give answers such as "thirteen", before realizing that the correct answer is obviously "three". To have at least one pair of the same colour (m = 2 holes, one per colour), using one pigeonhole per colour, you need only three socks (n = 3 objects).

If there are n *persons (at least two) who can arbitrarily shake hands with one another, there is always a pair of persons who shake the same number of hands.* Here, the 'holes' correspond to number of hands shaken. Each person can shake hands with anywhere from 0 to $n - 1$ other people. This creates n possible holes, but either the '0' or the '$n - 1$' hole must be empty (if one person shakes hands with everybody, it's not possible to have another person who shakes hands with nobody, and vice versa). This leaves n people to be placed in at most $n - 1$ non-empty holes, guaranteeing duplication.

The pigeonhole principle often arises in computer science. For example, collisions are inevitable in a hash table because the number of possible keys exceeds the number of indices in the array. No hashing algorithm, no matter how clever, can avoid these collisions. This principle also proves that any lossless compression algorithm that makes at least one input file smaller will make some other input file larger. (Otherwise, two files would be compressed to the same smaller file and restoring them would be ambiguous.)

Several additional examples are given by Grimaldi (see References).

Generalizations of the pigeonhole principle

A generalized version of this principle states that, if n discrete objects are to be allocated to m containers, then at least one container must hold no fewer than $\lceil n/m \rceil$ objects, where $\lceil x \rceil$ is the ceiling function, denoting the smallest integer larger than or equal to x.

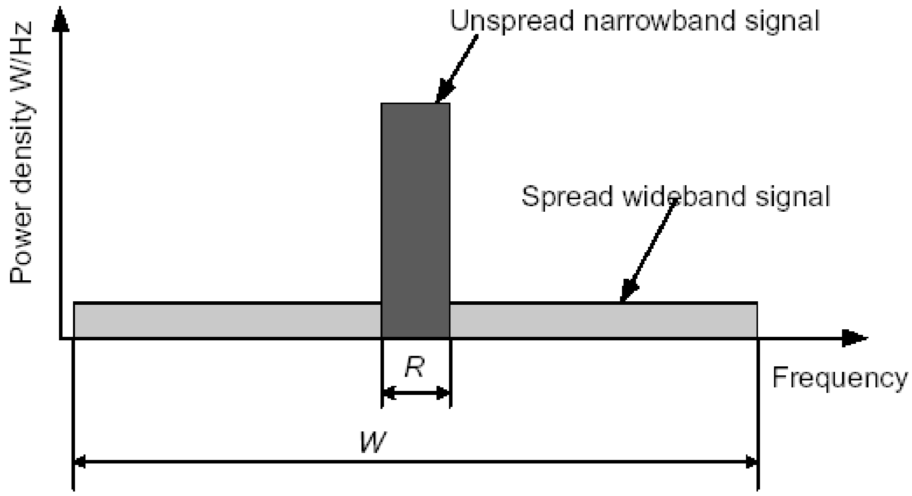
A probabilistic generalization of the pigeonhole principle states that if n pigeons are randomly put into m pigeonholes with uniform probability $1/m$, then at least one pigeonhole will hold more than one pigeon with probability

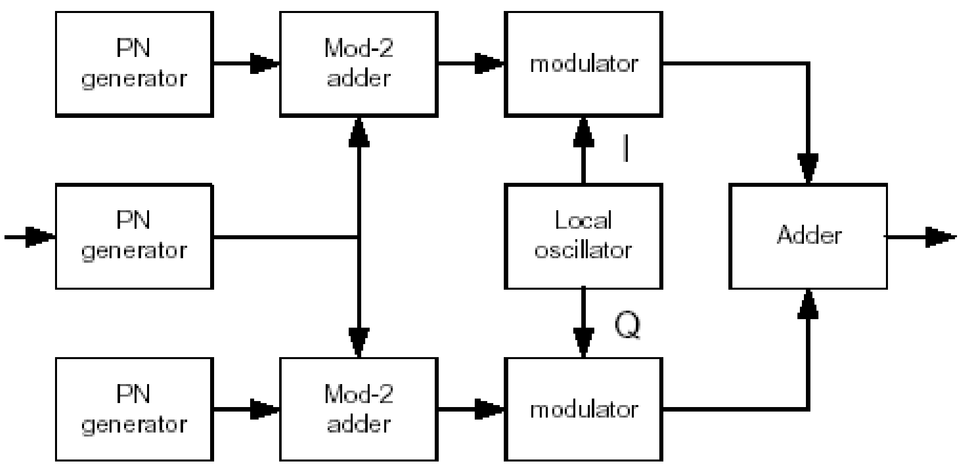
$$1 - \frac{m!}{(m-n)!\, m^n} = 1 - \frac{(m)_n}{m^n},$$

where $(m)_n$ is a falling factorial. For $n = 0$ and for $n = 1$ (and $m > 0$), that probability is zero; in other words, if there is just one pigeon, there cannot be a conflict. For $n > m$ (more pigeons than pigeonholes) it is one, in which case it coincides with the ordinary pigeonhole principle. But even if the number of pigeons does not exceed the number of pigeonholes ($n \leq m$), due to the random nature of the assignment of pigeons to pigeonholes there is often a substantial chance that clashes will occur. For example, if 2 pigeons are randomly assigned to 4 pigeonholes, there is a 25% chance that at least one pigeonhole will hold more than one pigeon; for 5 pigeons and 10 holes, that probability is 69.76%; and for 10 pigeons and 20 holes it is about 93.45%. This problem is treated at much greater length at birthday paradox.

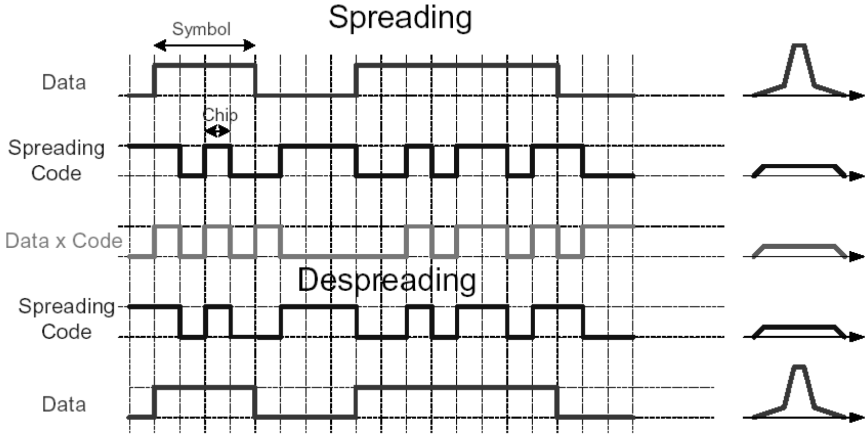
4.3 Process gain

• A narrowband signal is spread to a wideband signal.
• Information rate at the input of the encoder is R *bits/s*
• Available bandwidth is *W Hz*
• In order to utilize the entire available bandwidth the phase of the modulator is shifted
Pseudo randomly, according to the pattern from the PN generator at a rate W *times/s.*
• The duration of is called chip interval
• High bit rate means less processing gain and higher transmit power or smaller coverage.

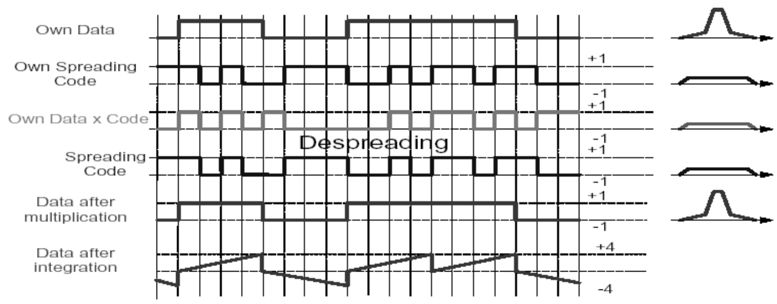




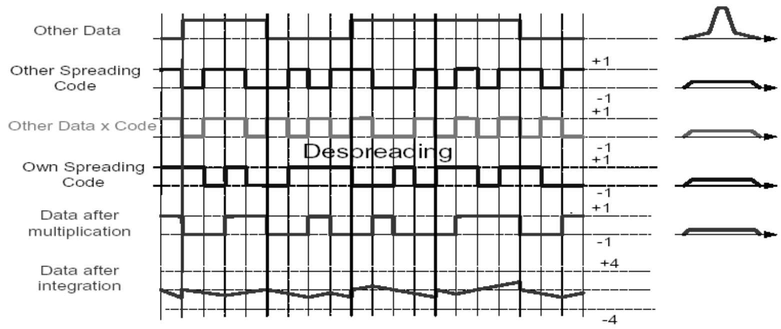
Spreading



Detection own signal



Detection other signal



4.4 Codes

Some features are described below —

• Requirements for the spreading codes:

– Good auto-correlation properties. For separating different paths.

– Good cross-correlation properties. For separating different channels.

Channelization codes used for channel separation from the same source.

• Same codes from all the cells.

• Short codes: used for channel separation in Uplink and Downlink.

– Othogonality property, reduce interference.

– Different spreading factors, different symbol rates.

– Limited resource, must be managed.

– Do not have good correlation properties, need for additional long code.

Scrambling codes.

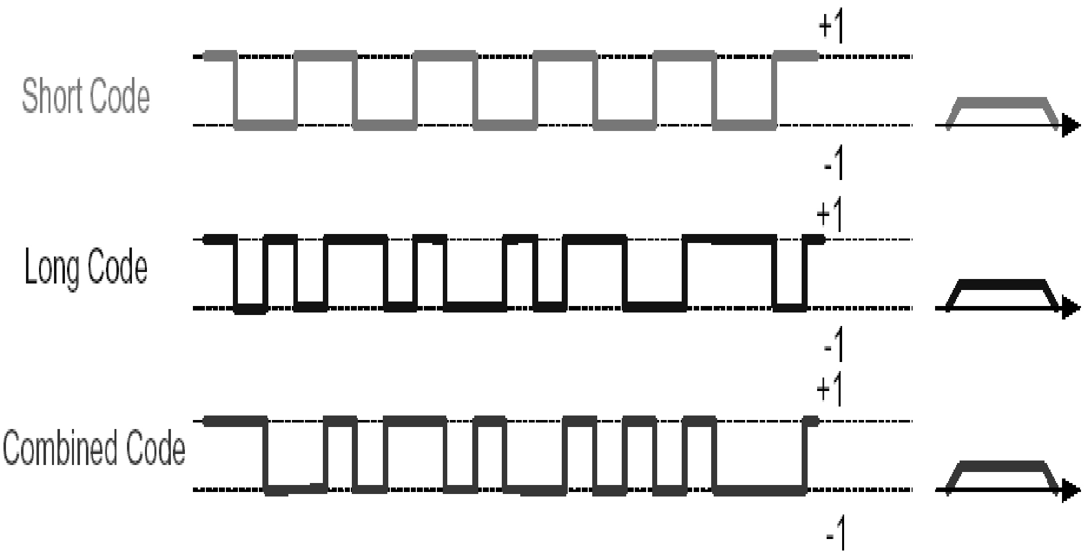
• Long Codes:

– Good correlation properties.

– Uplink: different users.

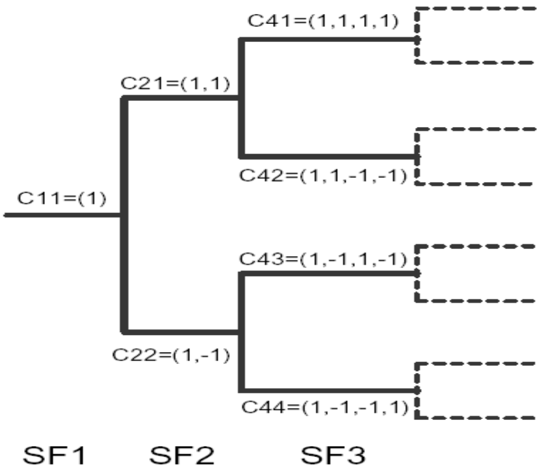
– Downlink: different BS.

<u>Long and Short Codes</u>



The tree of orthogonal codes

Orthogonal short codes will only be useful if channel can be synchronized in the symbol level.
– Mainly used in DL.
• Orthogonal Variable Spreading Factor technique.



• Orthogonality preserved across the different symbol rates.
• Codes must be allocated in RNC.
• Code tree may become fragmented code reshuffling may be needed.
• Provision of multiple code trees within one sector by concatenation with multiple sector specific long codes.

Generation of a scrambling codes:

• Spreading code is output of the binary shift register generator.

• Pseudo random codes are used: cyclic.

• Maximal length codes m-sequences: sequences that have maximal possible sequence given
the length of the shift registers.

• UL long scrambling code: complex scrambling codes, sum of two sequences (Gold sequence) generators:
– *X25+X3+1.*
– *X25+X3+X2+X+1.*

• UL short scrambling codes.

– Used to supporting Multiuser detection.

– Sequence length around 255 chip.

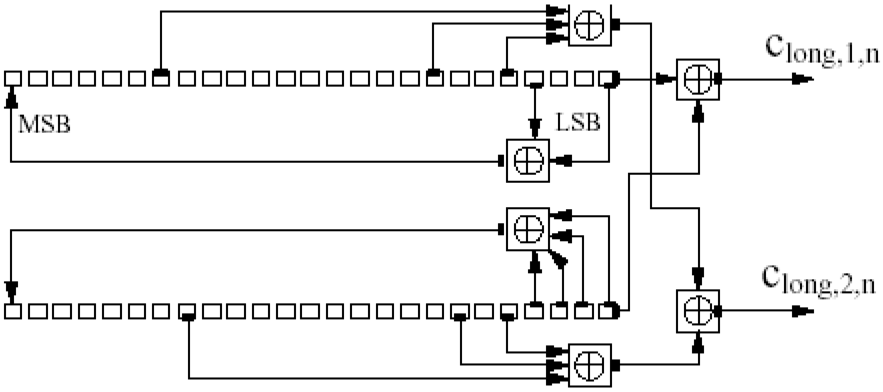


Fig : Configuration of uplink long scrambling sequence generator

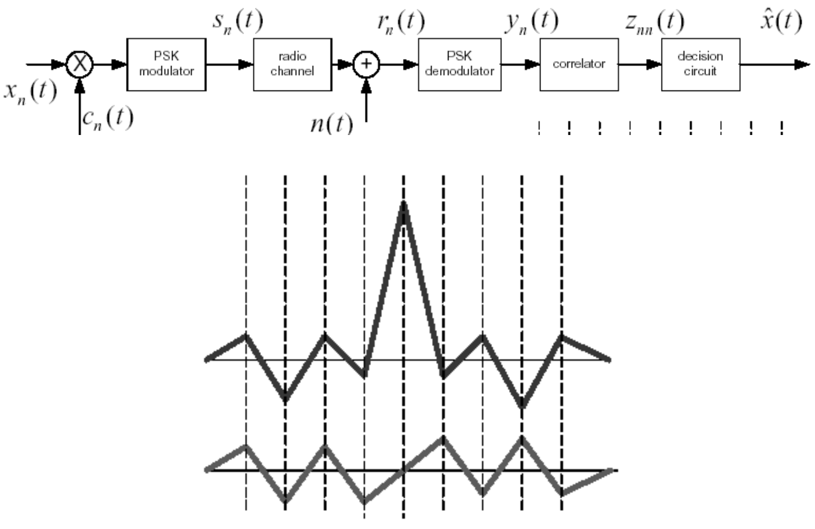
• DL scrambling sequences:

– Constructed by combining two real sequences with generator polynomials:

– *1+X7+X18*

4.5 Direct Sequence

By applying the above principles we can depict Direct sequence (DS) Spread Spectrum as following:



X n (t) user *n* information signal.

C n (t) user *n* spreading code.

S n (t) user *n* transmit signal.

Chapter5

CELLULAR NETWORK

5.1 Cell Types

For an optimal UMTS performance, it is proposed that UMTS network is planned with a hierarchical cell structure (HCS) using macro, micro and pico cells. In general, the more stringent the QoS and capacity requirements, the smaller the cell needs to be. A possible use of the hierarchical cell structure is shown in Figure. Large cells guarantee a continuous coverage for fast moving mobiles, while small cells are necessary to achieve good spectrum efficiency and high capacity for hot spot areas. With flexible deployment, it could be possible for an operator to redeploy pico cell channels for macro cells outside of urban cells in some locations.

F1/f1 Pico and macro cells
F2 macro cell layer
F3 Macro cell layer

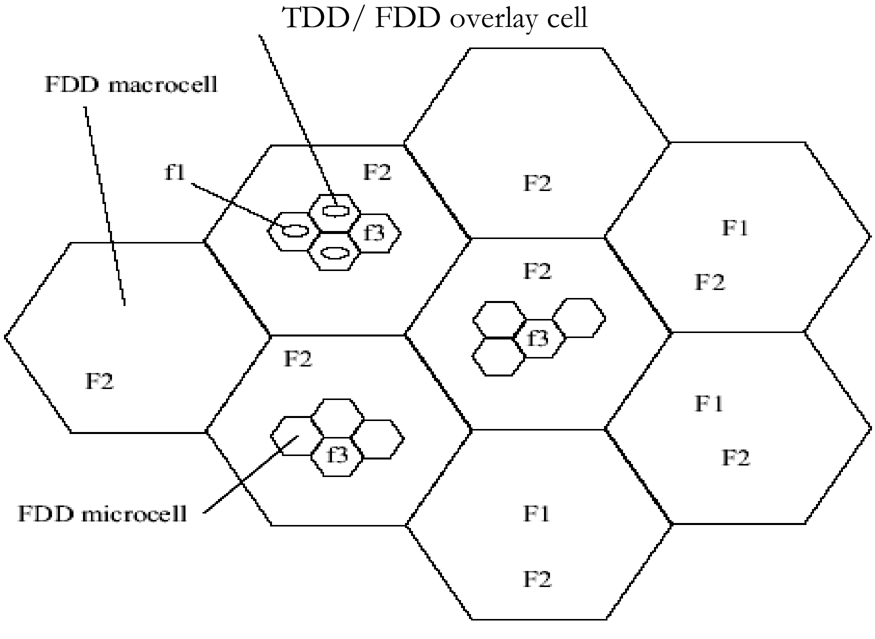


Figure UMTS hierarchical cell scenario.

The FDD macro cellular network provides the wide area coverage and it is used for high-speed mobiles. The micro cells are used at street level for outdoor coverage to provide extra capacity where macro cells could not scope. It would seem likely that these micro cells would not be hexagonal in shape but rather canyon like, reflecting the topography of a street and be perhaps 200–400 m in distance. The pico cell would be deployed mainly indoors in areas where there is a demand for high data rate services such as laptops networking or multimedia conferencing. Such cells may be of the order of 50 m in distance. A limiting factor will be the range of these terminals when used for high data rate services given the high demand this will place on batteries. Maximum bit rate for macro cells is to be 384 kbps and 2 Mbps for pico cells.

5.2 General Characteristic of a cellular network

<u>A cellular network</u>

is a radio network made up of a number of radio cells (or just cells) each served by a fixed transmitter, known as a cell site or base station. These cells are used to cover different areas in order to provide radio coverage over a wider area than the area of one cell. Cellular networks are inherently asymmetric with a set of fixed main transceivers each serving a cell and a set of distributed (generally, but not always, mobile) transceivers which provide services to the network's users.

Cellular networks offer a number of advantages over alternative solutions:

- increased capacity
- reduced power usage
- better coverage

A good (and simple) example of a cellular system is an old taxi driver's radio system where the taxi company will have several transmitters based around a city each operated by an individual operator.

<u>General characteristics</u>

The primary requirement for a network to be succeed as a cellular network is for it to have developed a standardized method for each distributed station to distinguish the signal emanating from its own transmitter from the signals received from other transmitters. Presently, there are two standardized solutions to this issue: · frequency division multiple access (FDMA) and code division multiple access (CDMA).

FDMA works by using varying frequencies for each neighboring cell. By tuning to the frequency of a chosen cell the distributed stations can avoid the signal from other cells. The principle of CDMA is more complex, but achieves the same result; the distributed transceivers can select one cell and listen to it. Other available methods of multiplexing such as polarization division multiple access (PDMA) and time division multiple access (TDMA) cannot be used to separate signals from one cell to the next since the effects of both vary with position and this would make signal separation practically impossible. Time division multiple access, however, is used in combination with either FDMA or CDMA in a number of systems to give multiple channels within the coverage area of a single cell.

In the case of the aforementioned taxi company, each radio has a knob. The knob acts as a channel selector and allows the radio to tune to different frequencies. As the drivers move around, they change from channel to channel. The drivers know which frequency covers approximately what area, when they don't get a signal from the transmitter, they also try other channels until they find one which works. The taxi drivers only speak one at a time, as invited by the operator (in a sense TDMA).

Broadcast messages and paging

Practically every cellular system has some kind of broadcast mechanism. This can be used directly for distributing information to multiple mobiles, commonly, for example in mobile telephony systems, the most important use of broadcast information is to set up channels for one to one communication between the mobile transceiver and the base station. This is called **paging**.

The details of the process of paging vary somewhat from network to network, but normally we know a limited number of cells where the phone is located (this group of cells is called a Location Area in the GSM or UMTS system, or Routing Area if a data packet session is involved). Paging takes place by sending the broadcast message to all of those cells. Paging messages can be used for information transfer. This happens in pagers, in CDMA systems for sending SMS messages, and in the UMTS system where it allows for low downlink latency in packet-based connections.

Our taxi network is a very good example here. The broadcast capability is often used to tell about road conditions and also to tell about work which is available to anybody. On the other hand, typically there is a list of taxis waiting for work. When a particular taxi comes up for work, the operator will call their number over the air. The taxi driver acknowledges that they are listening, then the operator reads out the address where the taxi driver has to go.

Frequency reuse

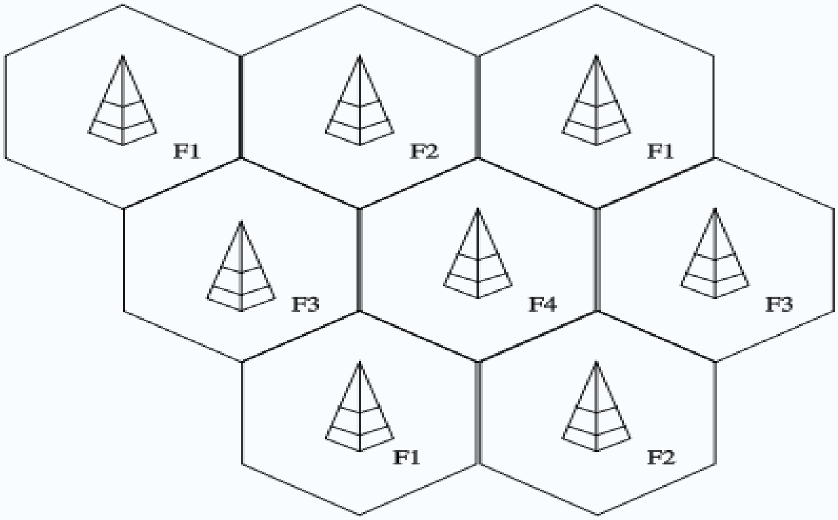


Fig : Frequency reuse in a cellular network

The increased capacity in a cellular network, compared with a network with a single transmitter, comes from the fact that the same radio frequency can be reused in a different area for a completely different transmission. If there is a single plain transmitter, only one transmission can be used on any given frequency. Unfortunately, there is inevitably some level of interference from the signal from the other cells which use the same frequency. This means that, in a standard FDMA system, there must be at least a one cell gap between cells which reuse the same frequency.

The frequency reuse factor is the rate at which the same frequency can be used in the network. It is 1/n where n is the number of cells which cannot use a frequency for transmission. A common value for the frequency reuse factor is 7.

Code division multiple access-based systems use a wider frequency band to achieve the same rate of transmission as FDMA, but this is compensated for by the ability to use a frequency reuse factor of 1. In other words, every cell uses the same frequency and the different systems are separated by codes rather than frequencies.

Depending on the size of the city, a taxi system may not have any frequency-reuse in its own city, but certainly in other nearby cities, the same frequency can be used. In a big city, on the other hand, frequency-reuse could certainly be in use.

5.3 Cellular telephony

Cell site

The most common example of a cellular network are mobile phone (cell phone) networks. A mobile phone is a portable telephone which receives or makes calls through a cell site (base station), or transmitting tower. Radio waves are used to transfer signals to and from the cell phone. Large geographic areas (representing the coverage range of a service provider) are split up into smaller cells to deal with line-of-sight signal loss and the large number of active phones in an area. In cities, each cell site has a range of up to approximately ½ mile, while in rural areas, the range is approximately 5 miles. Many times in clear open areas, a user may receive signal from a cell 25 miles away. Each cell overlaps other cell sites. All of the cell sites are connected to cellular telephone exchanges "switches", which in turn connect to the public telephone network or another switch of the cellular company.

As the phone user moves from one cell area to another, the switch automatically commands the handset and a cell site with a stronger signal (reported by the handset) to go to a new radio channel (frequency). When the handset responds through the new cell site, the exchange switches the connection to the new cell site.

With CDMA, multiple CDMA handsets share a specific radio channel; the signals are separated by using a pseudo noise code (PN code) specific to each phone. As the user moves from one cell to another, the handset sets up radio links with multiple cell sites (or sectors of the same site) simultaneously. This is known as "soft handoff" because, unlike with traditional cellular technology, there is no one defined point where the phone switches to the new cell.

Modern mobile phones use cells because radio frequencies are a limited, shared resource. Cell-sites and handsets change frequency under computer control and use low power transmitters so that a limited number of radio frequencies can be reused by many callers with less interference. CDMA handsets, in particular, must have strict power controls to avoid interference with each other. An incidental benefit is that the batteries in the handsets need less power.

Since almost all mobile phones use cellular technology, including GSM, CDMA, and AMPS (analog), the term "cell phone" is used interchangeably with "mobile phone"; however, an exception of mobile phones not using cellular technology is satellite phones.

Old systems predating the cellular principle may still be in use in places. The most notable real hold-out is used by many amateur radio operators who maintain phone patches in their clubs' VHF repeaters.

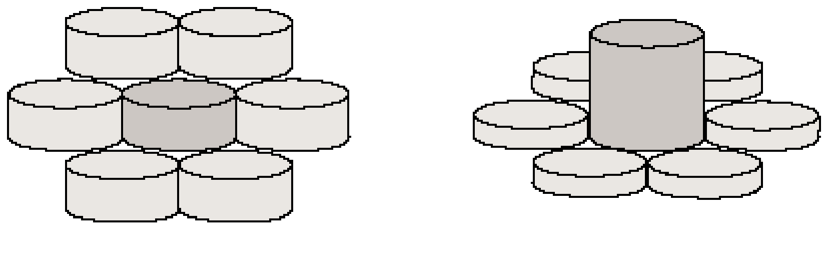
5.4 Call Admission Strategies

As described the cell coverage and the quality of ongoing connections will decline below the planned level, if cell interference is allowed to increase excessively. For this reason, certain radio resource management algorithms are needed to limit the amount of interference in the system. Call admission control (CAC or shortly AC) regulates the establishment and modification of radio access bearers to prevent the system from becoming overloaded. AC is used to achieve high traffic capacity and to maintain the stability of radio access network. Here, AC is examined from the uplink point of view. Uplink and downlink can be decoupled, since admission control decision in radio resource management is made practically independently in uplink and downlink.

Admission control in WCDMA is inherently different from the systems whose resources are finite and specified. The number of channels per sector is fixed, for instance, in frequency division and time division multiple access systems such as GSM. Thus, the capacity limit in those systems is a hard limit, and the AC only has to take care of the allocation of available channels, for instance, time slots for the users. CDMA has no hard limit on the maximum capacity, which makes admission control a complex soft capacity management problem. The impact of admission control algorithm is significant for the performance of WCDMA system, as the AC affects capacity, coverage and quality of service. Several admission control strategies have been proposed. One design choice is to restrict the admission by fixing the amount of resources, for instance, the maximum number of connections or the maximum total bit rate of the cell. The AC blocks calls at a base station, when the measured total power at that base station exceeds the predetermined threshold.

The total interference limiting admission controls essentially retains the soft-capacity feature of WCDMA. This is anticipated considering the inherent interference limitation of CDMA. The advantage of the total interference-based AC approach is intuitively the fact that all interference is treated equal without any explicit assumptions concerning the strength of the interference source. The total wideband interference is measured, and the admission control algorithm estimates the load increase that the establishment of a new bearer would cause. If the new resulting total interference would be unacceptably high, according to a predefined threshold value, the radio access bearer request is rejected. The approach described above directly utilizes the soft capacity feature. The less interference that there is coming from the neighboring cells, the more capacity there is available in the middle cell. This can be seen directly from the reuse efficiency factor, *F*. Some AC methods such as the throughput-based algorithm does not implicitly take into account the interference from the adjacent cells, but the other-cell interference is included as an estimated parameter.

Interference-based admission control treats different types of services in a uniform manner, and it is adaptive to load changes between the cells. However, on the minus side, it may be difficult to judge precisely what values should the threshold parameters, such as the maximum allowed total received power, have. This issue will be investigated by using quality of service monitoring approach.



(a) (b)

Figure : (a) Soft capacity of the middle cell when cells are equally loaded
(b) Soft capacity of the middle cell when there is less interference in the neighboring cells.

To allow vendor and operator specific solutions and to promote the development of efficient algorithms, the admission control details of UMTS are not specified.

Chapter6

RADIO ACCESS

6.1 Multiple Accesses

The basis for any mobile system is its air interface design, and particularly the way the common transmission medium is shared between users, that is, multiple access scheme. Multiple access schemes define how the radio spectrum is divided into channels, and how the channels separate the different users of the system. WCDMA is the multiple access method selected by ETSI as basis for UMTS air interface technology. Multiple access schemes can be classified into groups according to the nature of the protocol. The basic branches are contention less (scheduling) and contention (random access) protocols. The contention less protocols avoids the situation in which two or more users access the channel at the same time by scheduling the transmissions of the users. This can be done in a fixed fashion by allocating each user a static part of the transmission capacity, or in a demand-assigned fashion, in which scheduling only takes place between the users that have something to transmit. The fixed-assignment technique is used in frequency division multiple access (FDMA) and time division multiple access (TDMA), which are combined in many contemporary mobile radio systems such as GSM. In a FDMA system, the total system bandwidth is divided into several frequency channels that are allocated to users. In a TDMA system, one frequency channel is divided into time slots that are allocated to users, and the users only transmit during their assigned timeslots. Examples of demand-assignment contention less protocol are token bus and token ring LANs described by the IEEE in the 802.4 and 802.5 standards.

With the contention protocols, a user cannot be certain that the transmission will not collide, since other users may be accessing the channel at the same time. If several users transmit simultaneously, all of their transmissions will fail. Contention protocols, for example ALOHA-type protocols, resolve conflicts by waiting a random amount of time until retransmitting the collided message.

CDMA, and thus WCDMA, is very different from the techniques explained above. In principle, it is a contention less protocol allowing a number of users to transmit at the same time without conflict. However, contention will occur if the number of simultaneously transmitting users rise above some threshold.

In CDMA, each user is assigned a distinct code sequence (spreading code) that is used to encode the user's information-bearing signal. The receiver retrieves the desired signal by using the same code sequence at the reception. The division of TDMA, FDMA and CDMA channels into time-frequency plane is illustrated in Figure

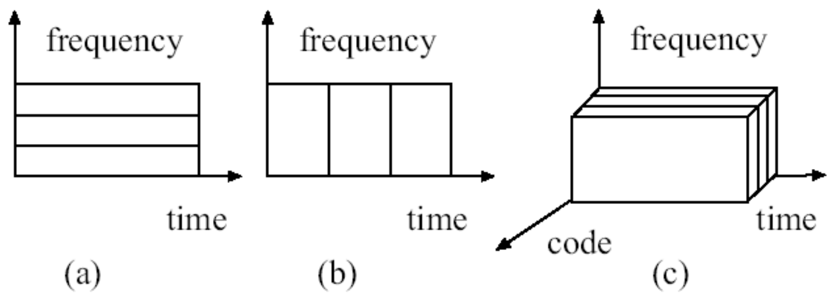


Figure : Multiple access schemes: (a) FDMA (b) TDMA (c) CDMA

6.2 CDMA Principles

Spread spectrum techniques use transmission bandwidth that is many times greater than the information bandwidth of any user. All radio resources are allocated to all users simultaneously. In CDMA, all communicating units transmit at the same time and over the same frequency. Multiple accesses are achieved by assigning each user or channel a distinguished spreading code (chip code). This chip code is used to transform a user's narrowband signal to a much wider spectrum prior to transmission. The receiver correlates the received composite signal with the same chip code to recover the original information-bearing signal.

The ratio of the transmitted bandwidth *Bt* to information bandwidth *Bi* is an important concept in CDMA systems. It is called the processing gain or the spreading factor, *Gp*, of the spread spectrum system. The capacity of the system and its ability to reject interference are directly proportional to *Gp*. Wide CDMA bandwidth, that is, high chip code rate gives higher processing gains, and thus better system performance.

$$G_P = \frac{B1}{B2}$$

When multiple users transmit a spread spectrum signal at the same time, the receiver is able to distinguish the information signal, since each user's distinct code has good auto- and cross-correlation properties. Thus, as the receiver decodes (despreads) the received signal, the transmitted signal power is increased above the noise, while the signals of the other users remain spread across the total bandwidth. The principle of the spreading and dispreading is illustrated in Figure. In Figure-a, the data signal of user 1 is spread into wideband signal. Figure -c shows the spreading operation for several other users.

Figure-b illustrates the received wideband signal, which consists of the signals from all the users, inclusive user 1. Figure-d shows the signal powers after the dispreading operation with the code of user 1. The signal of user 1 is retrieved by the receiver, whereas the rest of the signals appear random and are experienced as noise.

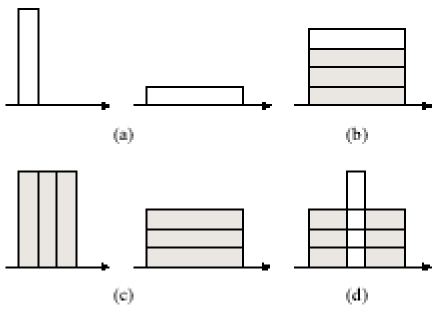


Figure: Principle of spread spectrum access:
(a) User 1 signal spreading (b) The received signal
(c) Spreading for several users (d) Dispread signal for user 1.

The described multiple access fully distinguishes CDMA from other multiple access systems. This makes the radio resource management of CDMA very challenging, since there is no absolute upper limit on the number of users that can be supported in each cell. This feature of CDMA is also called soft capacity. If the users are allowed to enter the system without any restrictions, the interference may increase to intolerable levels, thus damaging the quality of reverse links by causing power outage of some terminals. This thesis tries to find an efficient way to limit the number of uplink calls in order to improve the performance of the system. In general, the maximum number of users depends on many factors. These include interference that is generated at the base station by all the uplinked signals from own cell and other cells, and the propagation conditions which consist of path loss, shadowing and fast-fading. As already indicated, the components of the total interference are cross-correlation interference of users' signals and background noise. Overall, the CDMA systems are interference limited. The interference concept is essential for this thesis from admission control point of view and admission control related interference issues will be dealt separately in chapter 5. As described above, CDMA systems have limitations due to interference, and a brief summary is given next of the elements and technical solutions that are fundamental for the performance of a real CDMA network.

Power control (PC) Combats near-far problem. That is a situation, in which a mobile device close to a base station is received at higher power than a mobile located further away. Consequently, the reception of the mobile device's transmission is blocked. Power control solves this by increasing the output power as the mobile moves away from the base station, and by decreasing the transmit power as the mobile moves closer to the base station. Power control measures the signal-to-interference ratio (SIR) and sends commands to the transmitter on the other end to adjust the transmission power accordingly. Power control is used in both directions in WCDMA.

Soft handover – Handover (handoff) is the action of switching a call in progress from one cell to another without interruption when a mobile station moves from one cell to another. Neighboring cells in FDMA and TDMA cellular systems do not use the same frequencies. In those systems, a mobile station performs a hard handover when the signal strength of a neighboring cell exceeds the signal strength of the current cell with some threshold. In CDMA systems, the universal frequency reuse with factor of one is used. Thus, the previous approach would cause excessive interference in the neighboring cells. Neither is it feasible to perform an instantaneous handover, which would naturally solve this problem. The solution in CDMA systems is soft handover (soft handoff), in which a mobile user may receive and send the same call simultaneously from and to two or more base stations. In this way, the transmission power of a mobile can be controlled by the prevailing base station that receives the strongest signal.

Multipath signal reception - In a multi-path channel, the original transmitted signal reflects from obstacles such as buildings and mountains, and several copies of the signal, with slightly different delays, arrive at the receiver. From each multi-path signal's point of view, other multi-path signals can be regarded as interference and they are suppressed by the processing gain like other signals using the same channel. However, CDMA uses the Rake technique, in which the receiver has several parallel correlates that process the multipath components independently, and align them for optimal combining. Wideband CDMA is an extension of CDMA architecture using a large bandwidth of at least 5 MHz, and it has more advanced characteristics than the second generation CDMA systems. WCDMA is characterized by the following items:

. High chip rate (3.84 Mcps) and data rates (up to 2 Mbps)

. Provision of multi rate services

. Packet data

. Fast power control in the downlink

. Asynchronous base stations

. Seamless inter frequency handover

. Intersystem handovers, e.g., between GSM and WCDMA

Support for advanced technologies like multi-user detection (MUD and smart adaptive antennas There are two major techniques for obtaining a spread-spectrum signal: frequency hopping (FH) and direct sequence (DS) spreading.

Direct sequence spread spectrum (DS-SS). The data is directly coded by a high chip rate (spreading) code by multiplying the information-bearing signal with a pseudorandom binary waveform. The receiver knows how to generate the same code, and correlates the received signal with that code to extract the original data. UMTS is based on DS-CDMA. The important principles of DSCDMA will be discussed.

Frequency hopping spread spectrum (FH-SS). The carrier frequency at which the data is transmitted is changed rapidly according to the spreading code. By using the same code, the receiver knows where to find the signal at any given time.

6.3 Direct sequence CDMA

Direct sequence CDMA (DS-CDMA) has been described widely in the literature, for instance. In DS-CDMA the original information-bearing signal, that is, data signal is modulated on a carrier, which is spread by a high rate binary code sequence (chip code) to produce a bandwidth much larger than the original bandwidth. Logical binary symbols, bits 0 and 1, are suggested to be considered as mapped to real values –1 and +1 during the spreading operation. Various modulation techniques can be used for the code modulation, but usually some form of phase shift keying (PSK) such as binary phase shift keying (BPSK), quadrature phase shift keying (QPSK) or minimum phase shift keying (MSK) is employed.

The modulated wideband signal is transmitted through the radio channel. During the transmission, the modulated signal suffers from interference caused by the signals of other users. The desired signal together with interference reaches the receiver. At the reception, the receiver correlates the composite signal with the chip code of the desired signal. The multiplication by the distinct □□□ binary spreading waveform filters out large part of interference, and the original data is recovered. The cross-correlations of the code sequences of different users should be small so that the power ratio of the desired signal to the interfering signals will be large. The discrete cross-correlation between two different codes is given by:

$$R(k)=\frac{1}{N}\sum_{n=1}^{N} a_n b_{n+k}$$

where *an* and *bn* are the elements of the two sequences with code period *N,* and *k* is the time lag between the signals. In the following, the use of DS-CDMA will be illustrated by examples using the simplest form of spreading modulation, that is, BPSK. In BPSK modulation, the phase of the carrier is shifted 180 degrees in accordance with the transmitted digital bit stream. In the examples, a single bit transition, from 1 to 0 or from 0 to 1, causes a phase shift whereas two successive bits with equal values do not result in a phase shift. Let *x(t)* be the data stream that is to be modulated by a carrier having power *P* and radian frequency □ *0*. Then, the modulated stream, *sd(t)*, can be defined as:

$$s_d(t)=\sqrt{2P}\cdot x(t)\sin(\omega_0 t)$$

As an example, let the data stream being modulated to be (1 0) as in Figure 6-3a. The BPSK modulated data signal, *sd(t),* is shown in Figure-d. The wideband BPSK spreading is accomplished by multiplying *sd(t)* by a function *c(t)* that takes on values □□ 1. The transmitted wideband signal, *st(t)*, can thus be represented by:

$$s_t(t)=\sqrt{2P}\cdot c(t)x(t)\sin(\omega_0 t)$$

Let the chip code sequence *c(t)* to be (1 0 1 0), with processing gain four, as shown in Figure 3-3b. When modulo-2 addition is used, the spread data will be as shown in Figure 3-3c. The resulting transmission wave is depicted in Figure 6-3e. As previously noted, the signal will be despread at the receiving end using the same code as in transmission. After demodulation and despreading, the original data will be recovered. The received signal has a propagation delay *Td* that is determined by the path length. The signal, *st'(t)*, coming out of the receivers correlator is:

$$s_t'(t) = \sqrt{2P} \cdot c(t - T_d')c(t - T_d)x(t - T_d)\sin\left[\omega_0 (t - T_d)\right]$$

where *T'd* is the receiver's best estimate of the transmission delay. It can be seen that if the chip code *c(t)* at the receiver is correctly synchronized with the chip code at the transmitter (i.e., *T'd* = *Td*), the original data is recovered after dispreading and demodulation.

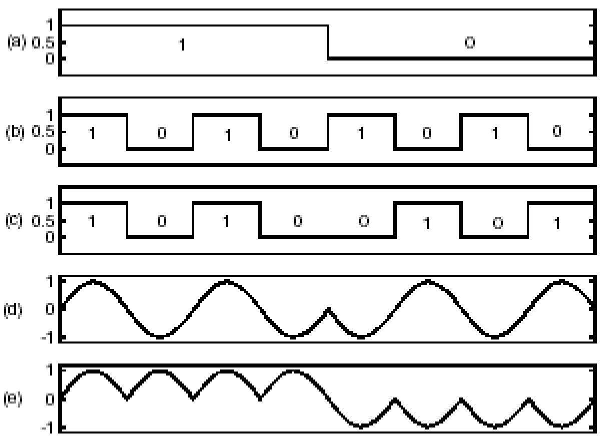


Figure 6-3 (a) User data (b) Spreading sequence (c) Spread data (d) Modulated data signal (e) Transmitted signal

As a second example, the spreading and dispreading is illustrated with three users. Let the data and the chip sequence of user 1, which are shown in Figure 6-4 a, be the same as those used in Figure. Let the data of user 2 be (1 0) and the spreading code (1 0 0 1). They are shown in Figure 6-4 c. In addition, let the data stream of user 3 to be (0 0) and the spreading code (1 1 0 0), as shown in Figure 6-4 e. The selected processing gain is again 4. Figure 6-3 b, Figure 6-3 d and Figure 6-4 f illustrate the spread signals of users 1, 2 and 3, respectively. The resulting composite signal of all the users is given by Figure 6-4 g. Figure 6-4 h shows the effect of the dispreading operation when the dispreading is applied to user 1. The decoded chips are integrated to give the decoded data. The retrieved signal is the original one, since the multiplication of the composite signal by the user 1 chip code cancels the interfering codes from others users. This is because the cross-correlation, *R(k)* , between the chip codes is zero, as the codes were selected orthogonal in the example.

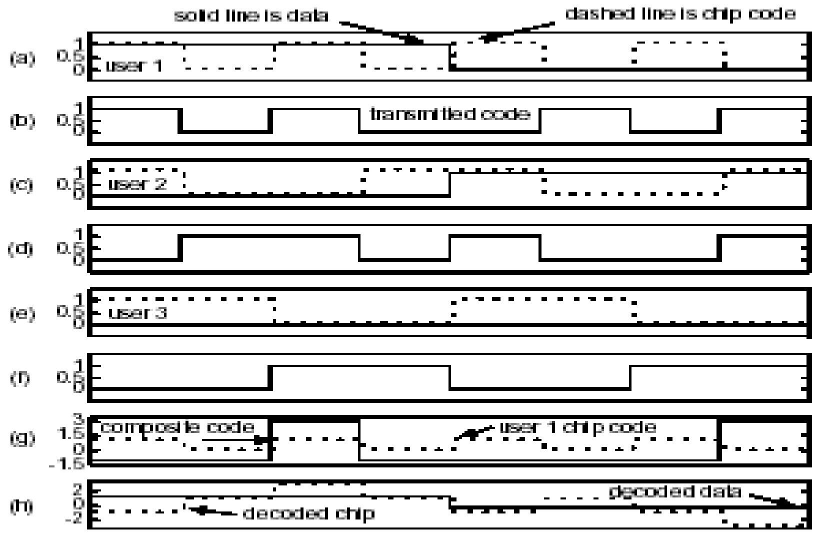


Figure 6-4 (a) User 1 data and the spreading sequence (b) Encoded user 1 data
(c) User 2 data and the spreading sequence (d) Encoded user 2 data
(e) User 3 data and the spreading sequence (f) Encoded user 3 data
(g) Composite data and spreading sequence for user 1
(h) Decoded chip and data for user 1.

It should be noted that orthogonal codes are completely orthogonal only for zero delay. For other delays, orthogonal codes have poor cross-correlation properties. Thus, they are suitable only if all the users of the same channel are synchronized in time to the accuracy of a small fraction of one chip. This is why PN (pseudo noise) codes are necessary in the reverse link. WCDMA uses Gold-sequences, which is a class of PN-code, for cell and user separation both in the downlink and in the uplink, and orthogonal codes for channel separation. The performance and interference resistance properties of Gold and other scrambling codes are evaluated. The choice of the spreading code is very important, as it is the basic building block of any CDMA system. Many families of spreading codes, with satisfactory auto- and cross-correlation properties,
In the actual systems, the processing gain is usually much larger than four, the value that was used in the previous examples. A large processing gain is, of course, highly beneficial in suppressing interference. For instance, the chip rate of WCDMA is 3.84 Mcps, which allows large spreading.

CONCLUSION

The access techniques discussed in this report are the latest technologies being used to expand the capacity and range of existing systems. FDMA was the first to be implemented in Phase I of P25 during the early 90's. Although difficult to plan, implementation was relatively simple and FDMA was the technology supported by the majority of the manufacturing community. To increase spectrum availability, Phase II of P25 is developing standards for TDMA technology as well as 6.25 kHz FDMA. Both of these technologies use digital technology but still provide the capability to interoperate with analog systems which provides interoperability between systems.

CDMA technology has many advantages and is the most recent multiple access technology to be considered. However, CDMA technology has not become a major player because it is more difficult and expensive to provide equipment that accommodates FDMA, TDMA, and CDMA systems. It is more difficult to provide backward compatibility, a primary objective of P25, if a CDMA system is implemented. The research conducted for this report did not reveal any plans to develop CDMA equipment for public safety radio equipment. However, the research did indicate that using a combination of CDMA and TDMA technologies could improve the quality of service and user capacity without loss of range in cellular telephone systems. As the demand for wireless services increase and technology advances, it is highly probable these techniques will continue to evolve within the public safety arena.

Summary of Results:

Previous studies of WB multiple access schemes mainly focused on the time hopping scheme. In this dissertation, we investigated the DS scheme combined with TH scheme. In this study, the system performance was evaluated in a multipath and multiuser fading environment. The Nakagami fading model was considered for the analysis. The criteria for evaluating system performance were the average SNIR, BER and outage probability.

Results obtained from this dissertation are consistent with published results and with characteristic behavior of WB systems.

Throughout this study, and based partially on interactions with selected U.S., European, and Japanese companies, it is recognized that there is substantial need for systems research for future wireless applications. The following research areas are either emerging or evolving and are considered important for future health of wireless communication systems:

- new decoding algorithms for turbo codes for wireless channels
- new coding/modulation techniques for reducing the peak-to-mean envelope ratio, maximizing the data rate and providing large coding gain
- new approaches to jointly designing modulation techniques, and power amplifiers to simultaneously obtain high power added efficiency along with bandwidth efficiency
- new demodulation/decoding techniques to simultaneously combat the near-far problem and do channel decoding in multi-rate DS-CDMA systems
- communication problems unique to high frequency systems (e.g., channel estimation)
- joint channel estimation and decoding/demodulation algorithms
- multiple-access techniques for multi-rate systems with variable quality of service requirements
- space-time coding for systems with multiple antennas
- analog decoding techniques for high speed, low power systems
- ultra wideband systems and hardware design
- research in methodologies for an integrated approach to wireless communications (device layer: e.g., power and low noise amplifiers, mixers, filters; physical layer: coding, modulation; medium access layer: CDMA/FDMA/TDMA; data link layer: hybrid ARQ; network layer: routing protocols)

APPENDICES:

- **Abbreviations**

3G	Third generation
3GPP	3rd generation partnership project
AC	Admission control
BPSK	Binary phase shift keying
BS	Base station
BTS	Base transceiver station
CAC	Call admission control
CDMA	Code division multiple access
CN	Core network
CS	Circuit switched
DL	Downlink
DS-CDMA	Direct sequence CDMA
DS-SS	Direct sequence SS
DTX	Discontinuous transmission
ETSI	European telecommunications standards institute
FDD	Frequency division duplex
FDMA	Frequency division multiple access
FH-SS	Frequency hopping spread spectrum
GGSN	Gateway GPRS support node
GMSC	Gateway MSC
GPRS	General packet radio system
GSM	Global system for mobile communications
HC	Handover control
HCS	Hierarchical cell structure
HLR	Home location register

HO	Handover
IEEE	Institute of electrical and electronic engineers
IMT-2000	International mobile telephony 2000
IT	Information technology
ITU	International telecommunications union
IP	Internet protocol
LAN	Local area network
LC	Load control
ME	Mobile equipment
MS	Mobile station
MSC	Mobile services switching center
MSK	Minimum phase shift keying
MUD	Multiuser detection
Node B	UMTS Base station
PC	Power control
PDF	Probability distribution function
PLMN	Public land mobile network
PN	Pseudo noise
PS	Packet schduler, Packet switched
PSK	Phase shift keying
QoS	Quality of service
QPSK	Quadrature phase shift keying
RAN	Radio access network
RAB	Radio access bearer
RNC	Radio network controller
RNS	Radio network sub system
RRM	Radio resource management
RTT	Radio transmission technology
SDU	Service data unit

SIM	Subscriber identity module
SIR	Signal to interference ratio
SGSN	Serving GPRS support node
MRC	Maximal ratio combining
OOK	On-off keying
PAM	Pulse amplitude modulation
PDA	Personal digital assistant
PDF	Probability density function
PHY	Physical
PPM	Pulse position modulation
PSK	Pulse shift keying
RF	Radio frequency
SD	Selection diversity
SNIR	Signal to noise plus interference ratio
SS	Spread spectrum
TH	Time hopping
UWB	Ultra wideband
WLAN	Wireless local area network
WPAN	Wireless personal area network

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