CHAPTER 1

INTRODUCTION

1.1 GENERAL

Today the wireless communication has attracted around two third of world population due to its linear development both in technology as well as increase in user demand. Wireless communication a fast growing technology not only attracted the consumer but also the developing business man, research students and enthusiastic engineers all around the globe.

The advances in mobile telephony can be traced in successive generations from the early "0G" services like Mobile Telephone Service (MTS), to first generation “1G” analog cellular network, second generation “2G” digital cellular networks, third generation “3G” broadband data services to the current state of the art, fourth generation “4G” native-IP networks. Fifth generation, “5G” will bring us perfect real world wireless are called World Wide Wireless Web (WWWW). 5G would be the most intelligent technology that interconnects the entire world without limits i.e. completed “WWWW” would be possible and explained in www.Evolution (2005), Amit Kumar (2010).

4G technology combines different current existing and future wireless network technologies to ensure freedom of movement and seamless roam from one technology to another. It provides multimedia applications to
the end user by different technologies through continuous and always best connection possible.

4G networks are integrated with core network and several radio access networks. The core interface is used for communication with the core network and radio access networks, the collection of radio interfaces are used for communication with the radio access networks and mobile users. This kind of integration combines multiple radio access interfaces into single network to provide seamless roaming/handoff Shanthi (2003).

The main distinguishing factor between 3G and 4G is data rate. 4G can support at least 100Mbps peak rate in full-mobility wide area coverage and 1Gbps in low-mobility local area coverage where as the speed of 3G can be up to 2Mbps, which is much lower than the speeds of 4G. However, 4G standard will base on broadband IP-based entirely applying packet switching method of transmission with seamlessly access convergence. It means that 4G integrates all access technologies, services and applications unlimitedly through wireless backbone and wire-line backbone by using IP address. https://www.google.co.inFuture+wireless+ Communication (Amit Kumar 2010).

OFDM stands for orthogonal frequency division multiplexing, which transmits large amount of digital data over the radio wave. OFDM works by splitting the radio signal into multiple smaller sub signals and then transmit simultaneously at different frequencies to the receiver. Large Area Synchronized Code Division Multiple Access (LAS-CDMA) enables high-speed data and increases voice capacity. Multi-Carrier Code Division Multiple Access (MC-CDMA), which is designed for running on wide area, called macro cell. The Local Multipoint Distribution System, (LMDS), designed for micro cell is used to carry voice, data, internet and video services in 25GHz and higher spectrum (Syed M. Zafi 2011).
Adaptive modulation based MC-CDMA system and Adaptive modulation based OFDMA system are the two transmission systems suitable for future wireless communication ensures large bandwidth, high data rate and error free data communication. It was accomplished by sending the output from Turbo encoder to serial to parallel converter. The output from serial to parallel converter was inhibited to various Adaptive modulation schemes like m-aryPSK, m-aryQAM, m-aryCPM, and m-aryMHPM. The modulated output was then fed to multiple access techniques like MC-CDMA system and OFDMA system in fast fading Rayleigh environment. The BER performance, SNR ratio and over all throughput were observed for both the MC-CDMA system and OFDMA system inhibiting various modulation schemes and the corresponding simulation results in Matlab and Simulink software were plotted.

1.2 OBJECTIVE OF THE RESEARCH

Long term Evolution (LTE) or 4G would be the next generation mobile standard to be introduced shortly. The salient features of LTE includes increased download and upload rates, use of Multiple Input and Multiple Output (MIMO) as antenna technology, Orthogonal Frequency Division Multiplexing (OFDM) as the modulation technique, improved spectral efficiency, quality of service, better integration with existing standards, embedded security and an all ‘IP’ network.

The objective of the research was to implement an Adaptive modulation system i.e. Adaptive modulation based schemes like m-aryPSK, m-aryQAM, m-aryCPM, and m-aryMHPM system in Multi carrier code division multiple access (MC-CDMA) and Orthogonal frequency division multiple access (OFDMA) system incorporated with Turbo encoder was designed as our system for the user rate of up to 32 numbers. The simulation
results were taken up in Rayleigh fading channel using Matlab and Simulink software and explained by Chatterjee (2003).

Here the MC-CDMA system was initially designed and design parameters were assigned below. Initially the 1/2 rated Turbo coder was implied for a channel bandwidth of 20 megahertz with 1024 sub carriers for a symbol rate of 640ksymbols/s in a free Rayleigh channel for up to 32 users. The User Vs BER graph was plotted for various Adaptive modulation schemes like m-aryPSK, m-aryQAM, m-aryCPM, and m-aryMHPM system in Multi carrier code division multiple access (MC-CDMA) and Orthogonal frequency division multiple access (OFDMA) system (Chatterjee 2003).

Then 1/3 rated Turbo coder was implied for a channel bandwidth of 20MHz with 1024 sub carriers for a symbol rate of 640ksymbols/s in a free Rayleigh channel for up to 32 users. Again the Users Vs BER graph was plotted for various adaptive modulation schemes like m-aryPSK, m-aryQAM, m-aryCPM, and m-aryMHPM system in MC-CDMA and OFDMA system.

From the graphical analysis, the best Adaptive modulation system with improved rated Turbo coder was chosen and concluded it as the best Adaptive modulation technique for both MC-CDMA and OFDMA techniques for future wireless communication.

Through the result it was clear that the m-aryMHPM was the best modulation technique for future wireless communication followed by m-aryCPM, m-aryQAM and m-aryPSK.
1.3 LITERATURE REVIEW

1.3.1 Brief History of Cellular Communication

Wireless communication touches the lives of more than two-third of the world population. It is an ever-evolving field and emerged as the one of the fastest growing technology sectors from the consumer, business and research perspective. In the research work, it was tried to put together a few trends that can define the direction of the wireless world in the next few years. The technical aspects are specifically analyzed from the Industry perspective. The target audiences for this thesis are the professionals and businessmen new to the field of wireless and students looking into specialization in this area (Shanthi 2003, Amit Kumar 2010).

1.3.1.1 Evolution of mobile communication in terms of generations

First Generation (1G): 1G was based on analog technology and basically intended for analog phones. It was launched in the early 1985. It introduced the first basic framework for mobile communications like the basic architecture, frequency multiplexing, roaming concept etc. Access technology used was Advances Mobile Phone Service (AMPS) and explained by Sivakumar Gupta (2011).

1.3.1.2 2G (Second Generation)

2G was a revolution that marked the switching of mobile communication technology from analog to digital. It was introduced in 1992 and it adopted digital signal processing techniques. GSM was one of the main attractive sides of 2G and it introduced the concept of Subscriber Identity Module (SIM) cards. Main access technologies are CDMA (Code Division Multiple Access) and GSM (Global System for Mobile Communication) were explained by Shanthi (2003).
1.3.1.3 2.5G (2.5Generation)

2.5G implemented in 1995 was an extension of 2G with packet switching incorporated to 2G. It implemented hybrid communication which connected the internet to mobile communication was explained by Shanthi (2003).

1.3.1.4 3G (Third Generation)

The basic idea of 3G was to deploy new systems with new services instead of providing higher bandwidth and data rate. Support for multimedia transmission is another striking feature of 3G. It employed both circuit switching and packet switching strategies. The main access technologies were CDMA, Wideband CDMA (WCDMA), and Time division synchronous CDMA (TS-SDMA) and explained by Shivakumar Guptha (2011).

1.3.1.5 4G (Fourth Generation)

The term 4G is used broadly to include several types of broadband wireless access communication systems. One of the terms used to describe 4G is MAGIC (mobile multimedia, anytime anywhere, global mobility support, integrated wireless solution, and customized personal service). The 4G systems not only supports the next generation mobile service, but also the fixed wireless networks was explained by Shanthi (2003).

Three Paths of 4G:

- Worldwide Interoperability for Microwave Access (WiMAX):
  Intel, Sprint, others
- Long Term Evolution (LTE): Ericsson, Nokia, others
- Ultra Mobile Broadband (UMB): Qualcomm.
1.3.2 Evolution of (LTE)

1.3.2.1 Long term evolution (LTE)

Long Term Evolution (LTE) preferred as the best solution, is evolved from GSM and WCDMA networks. Salient features of LTE includes increased download and upload rates, improved spectral efficiency, quality of service, better integration with existing standards, embedded security and an all ‘IP’ network was explained in Shanthi (2003), Amit Kumar (2010).

The above merits can be achieved by use of Adaptive modulation techniques, use of Multiple Input and Multiple Output (MIMO) as antenna technology and Orthogonal Frequency Division Multiplexing (OFDM) based multiple access techniques. The world’s first LTE network was launched in Sweden. In most parts of the world, including US, LTE is still in testing stages and is expected to be launched shortly was explained by Shivakmar Guptha (2011).

1.3.2.2 WiMAX

Worldwide Interoperability for Microwave Access is the name given to the IEEE 802.16 standard. It is used to provide last mile mobile broadband as well as backhaul or network access applications. WiMAX is especially considered a viable solution for developing countries to provide coverage in the rural areas. The biggest debate in the telecom industry is which technology WiMAX or LTE is to be widely adopted for next 5 years. It was explained in Shanthi (2003).
1.3.2.3 UMB

UMB is the brand name for a project 3GPP2 to improve the CDMA 2000 mobile phone standard for next generation applications and requirements.

Like LTE, the UMB system is also based upon Internet (TCP/IP) networking technology running over a next generation radio system, with peak rates of up to 100 Mbits/s. UMB was designed to support handoffs with other technologies including existing CDMA2000 1X and 1xEV-DO systems. However 3GPP added this functionality to LTE to become the single upgrade path for all wireless networks. No carrier had announced plans to adopt UMB and most CDMA carriers in Australia, USA, Canada, China, Japan and South Korea have already announced plans to adopt either WIMAX or LTE as their 4G technology and explained by Theodore Rappport (2002).

1.3.3 Vision of 4G

The new generation of wireless is intended to replace the 3G systems, perhaps in 5 to10 years.

The 4G key infrastructures are accessing information anywhere, anytime, with a seamless connection, receiving large volume of information, data, pictures, video, and others. The 4G infrastructure will consist of a set of various networks using IP (Internet protocol) as a common protocol so that users are able to choose all applications and environment Shanthi (2003).

Based on the developing trends of mobile communication, 4G will have broader bandwidth, higher data rate, quicker handoff and seamless service across the wireless systems and networks. The key concept is to
integrate all the existing mobile technologies through advanced technology was explained in Shanthi (2003).

Application and adaptability supports the users' traffic, air interfaces, radio environment, and quality of service (QOS). It provides connection among networks correctly and efficiently.

The fourth generation encompasses all systems including public to private networks, Operator-driven broadband networks, personal networks and ad hoc networks to integrate and work in seamless environment.

The 4G system interoperates with 2G system, 3G system as well as with Digital (broadband) broadcasting systems. In addition, 4G system as it is fully IP-based wireless Internet intends to integrate from satellite broadband to Wireless Local Loop (WLL) and Fixed Wireless Access (FWA) was explained by Theodore S. Rappaport (2002).

1.3.3.1 Need of going to 4G

Difficulty of FDMA, TDMA and CDMA to provide higher data rate, necessity for increased data rate, bandwidth to meet the multimedia requirements, limitation of spectrum and its allocation, inability to roam between different services, introduction of better system with reduced cost and requirement of seamless end-to-end transport were explained by Theodore S. Rappaport (2002).

1.3.3.2 Concept of 4G

The 4G mobile system is an all IP-based network system. The features of 4G may be summarized with one word integration. The 4G systems seamlessly integrates different technologies and networks to satisfy increasing user demands 4G technologies shall combine different current
existing and future wireless network technologies (e.g., IPv6, OFDM, MC-CDMA, LAS-CDMA and Network-LMDS) to ensure freedom of movement and seamless roam from one technology to another. It provides multimedia applications to a mobile user by different technologies through a continuous and always best connection possible.

4G networks are integrated with one core network and several radio access networks. A core interface is used for communication with the core network and radio access networks and a collection of radio interfaces are used for communication with the radio access networks and mobile users. This kind of integration combines multiple radio access interfaces into a single network to provide seamless roaming/handoff and the best connected services.

The main distinguishing factor between 3G and 4G is the data rate. 4G can support at least 100Mbps peak rates in full-mobility wide area coverage and 1Gbps in low-mobility local area coverage. The speeds of 3G can be up to 2Mbps which is much lower than the speeds of 4G. However 4G standard is based on broadband IP-based entirely packet switching method of transmission with seamlessly access convergence. It means that 4G integrates all access technologies, services and applications using IP address.

But 5G will bring us perfect real world wireless or called “WWW: World Wide Wireless Web”. The idea of WWWW, World Wide Wireless Web is started from 4G technologies. The following evolution will be based on 4G and completed its idea to form a real wireless world. Thus, 5G shall make an important difference and add more services and benefit to the world over 4G. 5G shall be the most intelligent technology that interconnects the entire world without limits. Completed WWWW was explained by Xichun Li and Abdullah Gani (2008).
1.3.3.3 Description of LTE mobile system

LTE an all IP-based network system can be summarized with a single word called integration. It seamlessly integrates different technologies and networks to satisfy the increase in user demands. LTE technology combines different current existing and future wireless network technologies like Internet protocol version-6 (IPv6), Orthogonal Frequency Division Multiple Access (OFDM), Multi Carrier CDMA (MC-CDMA), Large Area Synchronized CDMA (LAS-CDMA) and Network-Local Multi point Distribution System (Network-LMDS) to ensure freedom of movement and seamless roaming from one technology to another. It provides multimedia applications to the mobile user by different technologies through continuous and best connection was explained by Theodore S. Rappaport (2002).

LTE networks are integrated with one core network and several radio access networks. The core interface is used for communication with the core network and radio access networks. The collection of radio interfaces are used for communication with the radio access networks and mobile users. This kind of integration combines multiple radio access interfaces into a single network to provide seamless roaming/handoff and the best connected services. The main distinguishing factor between 3G and 4G is the data rate. Here LTE can support at least 100Mbps peak rate in full-mobility wide area coverage and 1Gbps in low mobility local area coverage was explained by Theodore S. Rappaport (2002).

LTE standard is based on broadband IP-based packet switching method of transmission with seamlessly access convergence. The LTE integrates all access technologies, services and applications using IP address was explained by Amit Kumar (2010).
5G bring perfect real world wireless are called World Wide Wireless Web (WWW). 5G will be most intelligent technology that interconnects the entire world without limits was explained by Xichun Li and Abdullah Gani (2008).

1.3.3.4 Advantages of LTE

LTE refers to the fourth generation cellular wireless standards. It provides a wide range of data up to ultra-broadband (gigabit-speed) internet access to mobile as well as stationary users. LTE cellular system have a target peak data rate of up to 100 Mbit/s for high mobility such as mobile access and up to approximately 1 Gbit/s for low mobility local wireless access. The scalable bandwidth of up to 40 MHz is provided was explained by Theodore S. Rappaport (2002).

LTE system is expected to provide a comprehensive and secure all-IP based solution where facilities such as IP telephony, ultra-broadband Internet access, gaming services and HDTV streamed multimedia will be provided to end users. The first LTE release has a theoretical net bit rate capacity of up to 100 Mbit/s in the downlink and 50 Mbit/s in the uplink while utilization of 20 MHz bandwidth channel.

4G or LTE is developed to accommodate the QOS and rate requirements set by existing 3G applications like wireless broadband access, multimedia messaging service (MMS), video chat, mobile TV, and also new services like High Definition Television (HDTV) content, minimal services like voice and data etc. It allows roaming with wireless local area networks and explained by Theodare Rapapport (2002) and https://www.google.co.in Future+wireless+Communication, Abdullah Gani (2008).
Figure 1.1 Evolutions of LTE/ 4G  https://www.google.co.in
Future+wireless+Communication

Table 1.1  Comparison between various Generations and drafted Shanthi (2003)

<table>
<thead>
<tr>
<th>Generation</th>
<th>Technology</th>
<th>Features</th>
</tr>
</thead>
</table>
| 1G wireless | Advanced Mobile Phone Service (AMPS). | • Analog voice service.  
<pre><code>                       |                                                   | • No Data service.                           |
</code></pre>
<p>| 2G wireless | • CDMA                             | • Digital voice service 9.6K to 14.4K bit/sec.     |
|             | • GSM                               | • CDMA, Time Division Multiple Access (TDMA) and PDC offer one-way data transmissions only. |
|             | • Personal digital Cellular (PDC)  | • Enhanced calling features like caller ID (Identification). |
|             | • WCDMA                            |                                                   |</p>
<table>
<thead>
<tr>
<th>Generation</th>
<th>Technology</th>
<th>Features</th>
</tr>
</thead>
</table>
| 3G wireless | - Based on the Interim standard-95 CDMA  
- Standard (CDMA 2000)  
- Time-division synchronous code division multiple Access (TD-SDM) | - Superior voice quality and data communication.  
- Up to 2M bits/s.  
- Broadband data services like video and multimedia.  
- Enhanced roaming.  
- Circuit and packet switched networks. |
| 4G /LTE | - OFDM & WOFDM  
- MC-CDMA)  
- LAS-CDMA  
- Adaptive modulation techniques | - Converged data and voice over Internet Protocol (IP).  
- Entirely packet switched networks.  
- All network elements are digital...  
- Higher bandwidth to provide. Multimedia services at lower cost (up to 100Mbits/sec). |

1.3.3.5 Objectives of the 4G Wireless Communications

Flexible channel bandwidth between 5 and 20 MHz, optionally up to 40 MHz is preferred. The nominal data rate of 100 Mbit/s while the client physically moves at high speeds relative to the station, and 1 Gbit/s while client and station are in relatively fixed positions. The data rate of 100 Mbit/s between any two points in the world.

Smooth handoff across heterogeneous networks, seamless connectivity, global roaming across multiple networks, high quality of service for next generation multimedia support real time audio, high speed data, HDTV video content, mobile TV, etc were explained by Shivakumar Guptha (2011).
1.3.3.6  Features of 4G wireless communication

Necessity for Adaptive modulation, high Speed, high capacity, cost per bit, Global mobility, Service portability, Scalability, Seamless switching, Quality of Service (QOS) requirements, Scheduling, call admission control techniques, Adhoc networks and Multi hop networks are the main features to be considered in 4G was explained by https://www.google.co.in Future+wireless+ Communication, Shanthi (2003).

1.3.3.7  Applications

Virtual presence: LTE provides user services at all times, even if the user is off-site.

Virtual navigation: LTE provides users with virtual navigation through which a user can access a database of the streets, buildings etc.

Tele-geoprocessing applications: This is a combination of (Geographical Information System (GIS) and Global Positioning System (GPS) in which a user can get the location by querying.

Tele-medicine and education: LTE will support remote health monitoring of patients. For people who are interested in lifelong education, 4G provides a good opportunity.

Crisis management: Natural disasters can cause break down in communication systems. In today’s world it might take days or 7 weeks to restore the system. But in LTE, it is expected to restore such crisis issues in a few hours (Shanthi 2003).
1.3.3.8 Multimedia – video services

LTE wireless systems are expected to deliver efficient multimedia services at very high data rates.

Basically there are two types of video services: bursting and streaming video services. Streaming is performed when a user requires real-time video services, in which the server delivers data continuously at a playback rate was explained by Theodore S. Rappaport (2002).

1.3.4 Access Schemes

As the wireless standards evolved, the access techniques used also requires increase in efficiency, capacity and scalability.

The first generation wireless standards used plain TDMA and FDMA. In the wireless channels, TDMA proved to be less efficient in handling the high data rate channels as it requires large guard periods to elevate the multipath impact. Similarly, FDMA consumed more bandwidth for guard to avoid inter carrier interference was explained by Theodore S. Rappaport (2002).

So in second generation systems, one set of standard used the combination of FDMA and TDMA and the other set introduced an access scheme called CDMA. Usage of CDMA increased the system capacity, but as a drawback placed a soft limit on it rather than the hard limit (i.e. a CDMA network do not reject new clients when it approaches its limits, resulting in a denial of service to all clients when the network overloads). Data rate was also increased as this access scheme (providing the network not reaching its capacity) was efficient enough to handle the multipath channel was explained by Theodore S. Rappaport (2002).
This enabled the third generation systems to use CDMA as the access scheme. However, the issue with CDMA is that it suffers from poor spectral flexibility and computationally intensive time-domain equalization (high number of multiplications per second) for wideband channels.

Recently, new access schemes like Orthogonal FDMA (OFDMA), Single Carrier FDMA (SC-FDMA), Interleaved FDMA and Multi-carrier CDMA (MC-CDMA) gain more importance for the next generation systems. These are based on efficient FFT algorithm and frequency domain equalization. The implementation of such schemes will control the bandwidth and form the spectrum in a flexible way was explained by Shivakumar Guptha (2011) and Xichun Li and Abdullah Gani (2008).

1.3.4.1 The other important advantages are

Orthogonal FDMA (OFDMA), Single Carrier FDMA (SC-FDMA), Interleaved FDMA and Multi-carrier CDMA (MC-CDMA) require less complexity for equalization at the receiver.

In addition to improvements in the multiplexing system, Adaptation of modulation techniques is followed. Earlier standards largely used Phase shift keying, where as adaptive modulation systems such as m-aryPSK, m-ary QAM, m-aryCPM and m-aryMHPM are proposed for use with 3GPP LTE standards was explained by Chatterjee (2003).

1.3.4.2 IPv6 support

Unlike 3G, which was based on two parallel infrastructures of circuit switched and packet switched network nodes. 4G will be based on packet switching and require low-latency data transmission.
The process of IPv4 address exhaustion is expected to be in its final stages. Therefore, in the context of 4G, IPv6 support becomes essential in order to support a large number of wireless-enabled devices. By increase in number of IP addresses, IPv6 removes the need for Network Address Translation (NAT), a method of sharing a limited number of addresses among a larger group of devices and explained in 4G from

1.3.4.3 Multiple access mechanisms

- Basically the multiple access techniques used for Cellular and Mobile Communication were
  - Time Division Multiple Access
  - Frequency Division Multiple Access
  - Code Division Multiple Access

a. FDMA

FDMA allows user for simultaneous transmission but every user gets only a small portion of total bandwidth. Shiv Kumar Gupta (2011).

b. TDMA

In TDMA, every user utilize entire system bandwidth and has to wait for next turn to come. Shiv Kumar Gupta (2011).

c. CDMA

Before CDMA, FDMA (Frequency division multiple access) and TDMA (Time division multiple access) were used. Since TDMA and FDMA were time limited and frequency limited, CDMA came into picture.
It allows each user in the system to use the total bandwidth as well as simultaneous transmission. Hence much attention was paid into it as it revolutionizes the spectral efficiency of multiple access schemes.

Each user was given one unique code (PN sequence of period N, hence ‘N’ should be large), and transmits the data after spreading the message in time domain. As only the desired receiver knows the correct PN sequence used in the transmitter, it could only decode, and others could not decode and it seemed to be the reason for choosing of CDMA. Theodore Rapapport (2002).

i. **Features of CDMA**

CDMA was based on spread spectrum transmission schemes originally developed for the military due to their resistance to jamming and low probability of intercept (i.e. relatively low power spectral density). These properties, combined with inherent resistance to multipath, make CDMA beneficial for commercial wireless networks.

The noise like properties of spread spectrum signals allowed CDMA to provide several key advantages over TDMA technology. CDMA was superior because the interference caused to co-channel users behaves like Additive White Gaussian Noise (AWGN), was tolerable. Specifically, the noise-like interference allows the system design to be based on average interference conditions as opposed to worst-case conditions, thereby allowing nearby transmitters to use the same carrier frequency (universal frequency reuse).

Further, CDMA allows more efficient statistical multiplexing of simultaneous users by taking advantage of voice activity and universal
frequency reuse and facilitates soft handoff which provides large-scale diversity advantage in cellular systems.

The second area in which CDMA technologies excel in its applications was wireless local area networks (WLAN). Due to the propensity of WLANs to cover small areas and to be uncoordinated with other WLANs, the networks were restricted to unlicensed bands. To allow uncoordinated networks to share the same frequency band, spread spectrum multiple access has been exploited, since it results in noise-like interference which increased the number of users supported by the system. The unlicensed bands for WLANs have fostered the widespread use and acceptance of CDMA throughout the world was explained by Theodore Rapaport (2002).

ii. Challenges of CDMA

In the early days of CDMA cellular systems, it was widely believed that the IS-95 uplink, with its asynchronous transmission, would be the bottleneck in system capacity. However, experience has shown that the downlink was typically the system bottleneck. In the uplink, power control for each mobile user ensures that, at the base station, each user has approximately the same signal level. However, in the downlink there were smaller numbers of unequally-powered signals, not conforming well to the assumption that each signal should look like AWGN to all other signals, arriving at a particular mobile station from the co-channel base stations.

The effect combined with the lack of sufficient channel diversity in slow fading, non-handoff scenarios, have caused lower capacities to be experienced in the downlink.

Third-generation CDMA networks mitigate this problem by adding fast power control and transmit diversity to the downlink.
Adding fast power control reduces the variability of the received signal strength in slow and moderate fading conditions. This, along with transmit diversity, significantly reduced the required power for slow-fading conditions. It was generally believed that future wireless networks will be highly asymmetric, with much larger capacity requirements necessary on the downlink (for Web browsing). Thus, gives the uncertainty in data usage.

The first challenge to high data rate (HDR) was currently being met by parallel groups within 3GPP and 3GPP2. 3GPP2 attempted to combine voice and data efficiently on a single carrier by evolving CDMA 2000 to the Evolution - Data and Voice (1xEV-DV) standard. Hence improvement in data efficiency of CDMA was achieved by implementation of shared downlink packet channel, high order adaptive modulation, hybrid ARQ schemes, and fast packet scheduling.

The key issue of practical limitation of CDMA was its performance inside buildings, where the multipath delay spread was much smaller than in outdoor settings. Qualcomm’s IS-95 used only 1.25 MHz bandwidth and a 1.2288 Mc/s chipping rate. Historically, this bandwidth decision was based on the fact that the early adopter carriers were originally only willing to allocate 10 percent of their 12.5 MHz U.S. cellular spectrum band for CDMA trials. The CDMA Rake receiver was therefore only able to exploit and distinguish multipath that exceeds single chip duration, or about 800 nanoseconds.

For multipath delays less than 800 ns, the CDMA signal fades as same as a conventional narrowband signal. Thus, indoor deployments of CDMA (where delay spreads were typically only 100-200 ns either use a link budget that accounts for Rayleigh or Ricean fading). Multipath was induced within the buildings by adding propagation delays in a distributed antenna system (DAS). In addition, a GPS clock was required for each CDMA base station, and it was difficult to bring such a clock signal into a large building.
New fiber-based distribution systems, however, allow the entire cellular/PCS spectrum to be transmitted into buildings from an external or roof mounted base station, and microcells located outside the buildings were able to provide coverage into buildings with sufficient time diversity in the channel. It was worth noting that 3G CDMA systems with greater bandwidth allows the spreading code to have multipath diversity benefit inside buildings was explained by Theodor Repapport (2002).

1.3.4.4 Attempt to go for multi carrier or OFDM communication

The failure of the above listed attempted to improve CDMA influences the design of fourth-generation wireless networks where OFDM is considered as the physical layer. CDMA versions of OFDM, MC-CDMA and OFDMA considered for 4G. www. wireless communication.nl / reference /chaptr05/... / mcm1.htm., Syed M. Zafi S. Shah (2011).

a. Multi-Carrier Modulation (MCM)

The Multi-Carrier Modulation (MCM) transmits data by dividing the stream into several bit streams, with much lower bit rate, and these sub streams are used to modulate several carriers.

The MCM was used in military HF radio links in the late 1950s and early 1960s. A special form of MCM, called OFDM, with densely spaced sub carriers with overlapping spectra of the modulating signal, was patented in the U.S. in 1970.

OFDM abandoned the use of steep band pass filters which separates the spectrum of individual sub carriers.

Instead, OFDM time-domain waveforms are chosen such that mutual orthogonal is ensured even though spectra overlap exist. The
waveform can be generated using FFT at the transmitter and receiver. www.wirelesscommunication.nl/ reference/chaptr05/.../mcm1.htm.

After many years of further intensive research in the 1980's, MCM has become part of several standards. It was explained by Vasu Chakravarthy (2005).

1.3.4.5 OFDM

In OFDM or multicarrier systems, multiple bits are transmitted in parallel over multiple sub carriers. If one sub carrier is in fade, the other may not. Error correction coding can be used to correct bit errors on faded sub carriers. Rapid fading (Doppler) erode the orthogonal of closely spaced sub carriers. It was explained by Shivkumargupta (2011) and Arnon Friedmann (2006), Syed M. Zafri S. Shah (2011).

a. Necessity to choose multi carrier communication

Orthogonal Frequency Division Multiplexing (OFDM) is a special form of multicarrier transmission where a single high-speed data stream is transmitted over a number of lower-rate sub carriers.

The discrete Fourier transform implementation of OFDM is pioneers in the early 1970s. OFDM is a strong candidate for commercial high-speed broadband wireless communications, due to advancement in very-large-scale-integration (VLSI) technology. In addition, OFDM technology possesses number of unique features which makes it an attractive choice for high-speed broadband wireless communications:

OFDM is robust against multipath fading and intersymbol interference as the symbol duration increases for the lower rate parallel subcarriers. (For a given delay spread, the implementation complexity of an
OFDM receiver is considerably simpler than that of a single carrier with an equalizer.

OFDM allows efficient use of the available radio frequency (RF) spectrum by using adaptive modulation and power allocation across the subcarriers that are matched to slowly varying channel conditions using programmable digital signal processors, thereby enabled bandwidth-on-demand technology and achieve higher spectral efficiency.

Current trends suggest that OFDM will be the best choice for fourth-generation broadband multimedia wireless communication systems; however there are several hurdles need to be overcome before OFDM finds widespread use in modern wireless communication systems and explained by Arnon Friedmann (2006) and Yongzhe Xie (2003).

b. **Advantages of OFDM**

The primary advantage of OFDM system is its robustness on frequency selective fading and multipath delay spread. As it truncates the long data packet into symbols with narrow bandwidth. Since the time-bandwidth product is constant, the duration of the symbol length is made high and hence the relative amount of dispersion caused by multipath delay spread is decreased.

Another advantage is implementation of frequency diversity. As different symbols arrive at the receiver on independent orthogonal subcarriers, they are uncorrelated at the receiver. Hence frequency diversity scheme is used for correlation in the receiver.

The next advantage is the quantity of subcarriers. As N no. of subcarriers are used, N no of modulators are required in the transmitter, but
instead of that, N point FFT on input binary data stream is implemented and the same reverse effect is produced with N demodulators in the receiver. Hence the system becomes faster with out complexity in transmitter-receiver circuit and explained by Arnon Friedman (2006) and Syed M. Zafi S. Shah (2011).

c. Disadvantages of OFDM

The orthogonal sub carriers are not properly synchronized. It is more sensitive to frequency offset and phase noise.

1.3.4.6 OFDMA

In OFDM, usable bandwidth is divided into a large number of smaller bandwidths that are mathematically orthogonal using FFT. Reconstruction of the band is performed by the IFFT. FFTs and IFFTs are well-defined algorithms implemented when sized as powers of 2.

Typical FFT sizes for OFDM systems are 128, 256, 512, 1024 or 2048 possibilities. The bandwidths that are supported are 5, 10 and 20 MHz. One beneficial feature of this technique is the ease of adaptation to different bandwidths.

The smaller bandwidth unit can remain fixed, even as the total bandwidth utilization is hanged. For example, a 10-MHz bandwidth allocation is divided into 1,024 smaller bands, whereas a 5 MHz allocation can be divided into 512 smaller bands. These smaller bands are referred to as sub carriers and are typically on the order of 10 kHz (Arnon Friedmann 2006).
a. **Features of OFDMA**

OFDMA is developed to move OFDM technology from a fixed-access wireless system to a true cellular system with mobility. The underlying technology is the same, but more flexibility is defined in the operation of OFDMA system. In OFDMA, subcarriers are grouped into larger units, referred to as sub channels, and these sub channels are further grouped into bursts can be allocated to wireless users.

Each burst allocation can be changed from frame to frame as well as with in the modulation order. This allows the base station to dynamically adjust the bandwidth usage according to the current system requirements.

In addition, since each user consumes only a portion of the total bandwidth, the power of each user is modulated according to the current system requirements.

Quality of service (QOS) is another feature that can be adapted for different users depending on their specific application, such as voice, streaming video, or internet access (Arnon Friedmann 2006)

b. **Bandwidth flexibility**

OFDM and OFDMA allow systems to easily adapt to the available spectrum. The stated goals of both the long term evolution of 3GPP and WiMAX are to support bandwidth allocations from 1.25 to 20 MHz.

In addition, the system supports either time division or frequency division multiplexing. All of this flexibility will allow service providers to roll out 4G systems in different ways for different areas according to the necessity of the market and explained by Arnon Friedmann (2006).
c. **Operation Principle of OFDMA**

OFDMA is the multi-user version of the OFDM digital modulation scheme. For achieving multiple accesses, subsets of subcarriers are provided to the individual users in OFDMA. This allows simultaneous low data rate transmission from several users.

d. **Advantages of OFDMA**

Flexibility of deployment across various frequency bands with little modification to the air interface.

Average interferences from neighboring cells, by using different basic carrier permutations between users in different cells. Interferences within the cell are averaged by using allocation with cyclic permutations.

OFDMA enables single frequency network coverage, where coverage problem does not exist and gives excellent coverage. It offers frequency diversity by spreading the carriers all over the used spectrum and allows per channel or per sub channel power control.

e. **Disadvantages of OFDMA**

Higher sensitivity to frequency offset and phase noise. Asynchronous data communication services such as web access are characterized by short communication bursts at high data rate where as few users in the base station cell transfer data simultaneously at low constant data rate.

The OFDM diversity gain, and resistance to frequency-selective fading, might partly be lost if very few sub-carriers are assigned to each user, and if the same carrier is used in every OFDM symbol. Adaptive sub-carrier
assignment based on fast feedback information about the channel, or sub-carrier frequency hopping, is therefore desirable.

Dealing with co-channel interference from nearby cells is more complex in OFDM than in CDMA. It requires dynamic channel allocation with advanced coordination among adjacent base stations.

The fast channel feedback information and adaptive sub-carrier assignment is more complex than CDMA fast power control and explained by Imran Rajeswari (2012).

1.3.4.7 The Necessity for MC-CDMA

CDMA is based on spread spectrum concept. In CDMA message signal is multiplied with pseudo noise sequence and spread the time domain message over the larger bandwidth, hence make the signal power lower than the noise power. It is difficult to demodulate/decode the message signal in the middle of transmission. Only in the receiver with appropriate PN sequence, the message signal can be decoded.

Future mobile communication system requires high data rate and good quality of service to consumers. The higher data rate is achieved with the use of multi carrier system such as OFDM. MC-CDMA combines DS-CDMA technique with OFDM. Hence MC-CDMA achieves higher data rate with optimum spectrum efficiency. Also MC-CDMA is robust technique for multipath fading environment and it was explained by Vasu Chakravarthy (2005).

MC-CDMA is a multiple access scheme used in OFDM-based telecommunication systems, allowing the system to support multiple users at the same time.
MC-CDMA is a form of CDMA in which spreading is applied in frequency domain rather than time domain. MC-CDMA is a form of Direct sequence CDMA where an FFT is performed after spreading. MC-CDMA is a form of OFDM, but an orthogonal matrix operation to the user bits is initially applied before spreading. Therefore, MC-CDMA might call as "CDMA-OFDM" and it was explained by Vasu Chakravarthy (2005).

Even though MC-CDMA is a form of direct sequence CDMA, the Fourier transform of Walsh Hadamard sequence is used as the code sequence. Each bit is transmitted simultaneously (in parallel) on many different subcarriers. Each subcarrier has a (constant) phase offset. The set of frequency offsets form a code to distinguish different users exhibiting frequency diversity.

a. **Compared to DIRECT SEQUENCE (DS) CDMA**

DS-CDMA is a method in which the spectrum is shared among multiple simultaneous users. The RAKE receiver is used to exploit frequency diversity. However, in dispersive multipath channel, DS-CDMA with a spread factor N can accommodate N simultaneous users only if highly complex interference cancellation techniques are used. In practice, this is difficult to implement. MC-CDMA can handle N simultaneous users with good BER, using standard receiver techniques and it was explained Shinsuke Hara (1997)

b. **OFDMA Vs MC-CDMA in performance**

OFDMA with frequency spreading is called MC-CDMA. OFDMA with time spreading is called MC-DS-CDMA(Multi carrier Direct sequence Code division multiple access and MT-CDMA(Multi tone Code division multiple access) where as OFDMA with both time and frequency spreading is called Orthogonal Frequency Code Division Multiple access (OFCDMA).
In MC-DS-CDMA, OFDM is used as the modulation scheme. The data symbols on the individual subcarriers are spread in time by multiplying the chips on a PN code by the data symbol on the subcarrier. The PN code chips consist of \{1, -1\} and the data symbol on the subcarrier is $-j$. The symbol being modulated for symbols 0 and 1, are $-j$ for symbol 0 and $+j$ for symbol 1.

c. Multi-carrier code division multiple access

Here in OFDMA with both time and frequency spreading (Orthogonal Frequency Code Division Multiple Access (OFCDMA)), 2-dimensional spreading in both the frequency and time domains is also possible. Such a scheme using 2-D spreading is called as VSF-OFCDM (variable spreading factor orthogonal frequency code-division multiplexing).

The combination of MC and CDMA techniques leads to MC-CDMA is illustrated in the Figure 1.2. The MC-CDMA transmitter spreads the user data stream using a given sub carrier and N chips (in Figure 1.3 C1… C4, N=4, chip duration is T) per symbol are transmitted in parallel on different subcarriers and at a much lower rate. Chip duration after the serial to parallel converter becomes NT. In practice, N is chosen as large as enough to reduce ISI and it was explained Shinsuke Hara (1997).
Figure 1.2 OFDM-TDMA signal structures

Figure 1.3 OFDM-FDMA signal structures

Figure 1.4 MC-CDMA multi-user signal structure

The above figs were drafted and explained in Syed M. Za S. Shah (2011).


32

d. Advantages of MC-CDMA over DS-CDMA and OFDM

Compared to DS-CDMA, MC-CDMA share spectrum among multiple simultaneous users. Moreover, it can exploit frequency diversity, using RAKE receiver. However, in a dispersive multipath channel, DS-CDMA with a spread factor N can accommodate only N simultaneous users if highly complex interference cancellation techniques are used. In practice this is difficult to implement. Whereas MC-CDMA can handle N simultaneous users with good BER by using standard receiver techniques. And it was explained by Shinsuke Hara (1997) and www.wireless communication.nl/reference/.../mccdma/mccdma.htm.

To avoid excessive bit errors on sub carriers that are in a deep fade, OFDM typically applies coding. Hence, the number of sub carriers need is larger than the number of bits or symbols transmitted simultaneously. MC-CDMA replaces this encoder by an NxN matrix operation and provides good BER.

One way of interpreting MC-CDMA was to regard it as a direct-sequence CDMA signal (DS-CDMA) which was transmitted after passing through an inverse FFT (Fast Fourier Transform) and it was explained by Vasu Chakravarthy (2005), Syed M.Zafi (2011).

1.3.5 Analog Modulation Techniques

Amplitude modulation (AM): Here the amplitude of the carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal.
Angle modulation: which is approximately constant envelope. Frequency modulation (FM): Here the frequency of the carrier signal is varied in accordance to the instantaneous frequency of the modulating signal.

Phase modulation (PM): Here the phase shift of the carrier signal is varied in accordance to the instantaneous phase of the modulating signal.

1.3.5.1 Digital modulation methods

In digital modulation, the analog carrier signal is modulated by a discrete signal. Digital modulation methods can be considered as digital-to-analog conversion, and the corresponding demodulation or detection as analog-to-digital conversion. The changes in the carrier signal are chosen from a finite number of \( M \) alternative symbols (the modulation alphabet).

1.3.5.2 Types of Common digital modulation techniques

The most common digital modulation techniques are:

- Phase-shift keying (PSK)
  - Binary PSK (BPSK), using \( M=2 \) symbols
  - Quadrature PSK (QPSK), using \( M=4 \) symbols
  - 8PSK, using \( M=8 \) symbols
  - 16PSK, using \( M=16 \) symbols
  - Differential PSK (DPSK)
  - Differential QPSK (DQPSK)
  - Offset QPSK (OQPSK)
  - \( \pi/4 \)-QPSK
- Frequency-shift keying (FSK)
  - Audio frequency-shift keying (AFSK)
  - Multi-frequency shift keying (M-ary FSK or MFSK)
  - Dual-tone multi-frequency (DTMF)
- Amplitude-shift keying (ASK)
- On-off keying (OOK), the most common ASK form
  - M-ary vestigial sideband modulation, for example 8VSB
- Quadrature amplitude modulation (QAM), a combination of PSK and ASK
  - Polar modulation like Gamma combination of PSK and ASK.
- Continuous phase modulation (CPM) methods
  - Minimum-shift keying (MSK)
  - Gaussian minimum-shift keying (GMSK)
  - Continuous-phase frequency-shift keying (CPFSK)
  - Multi-hop phase modulation (MHPM).
- Orthogonal frequency-division multiplexing (OFDM) modulation:
  - Discrete Multitone (DMT) - includes adaptive modulation and bit-loading.
- Wavelet modulation
- Trellis coded modulation (TCM), also known as trellis modulation
MSK and GMSK are particular cases of continuous phase modulation. Indeed, MSK is a particular case of the sub-family of CPM known as continuous-phase frequency-shift keying (CPFSK) which is defined by a rectangular frequency pulse i.e. a linearly increasing phase pulse of one symbol-time duration (Krishna Pillai 2008)

1.3.5.3 Miscellaneous modulation techniques

- The use of on-off keying to transmit Morse code at radio frequencies is known as continuous wave (CW) operation.
- Adaptive modulation techniques
- Space modulation a method whereby signals are modulated within airspace, are used in instrument landing systems.

1.3.5.4 Features of Adaptive Modulation

RF power output has been a major planning aspect for engineers since the start of radio transmission. Undoubtedly important, RF Power level is one of the many factors that determine a successful wireless network.

To evaluate and differentiate between various microwave systems and link performance, several key aspects of RF power output, propagation and antennas set aside, parameters such as receiver threshold, modulation type and RF power level are the most important factors of consideration.

Adaptive modulation schemes and ATPC (Automatic Transmit Power Control) provide point-to-point microwave systems with a high degree of flexibility, ensures better efficiency under changing weather conditions. RF output power can be controlled dynamically so as to ensure the highest power efficiency under changing modulations.
The goal of Adaptive modulation is to improve the operational efficiency of microwave links by increase of network capacity over the existing infrastructure thereby reducing sensitivity to environmental interferences.

Adaptive modulation means dynamically varying the modulation in an errorless manner in order to maximize the throughput under momentary propagation conditions. In other words, a system can operate at maximum throughput under clear sky conditions, and decreases gradually under rain fade.

1.3.5.5 Benefits of Adaptive Modulation

Operators evaluate point to point microwave links according to capacity and availability parameters. This in turn imply having a detailed plan of frequencies, channel bandwidth, modulation, antenna size, link configuration, diversity schemes, transmission power and more. Radio network planners can increase the capacity of given link at any time.

Microwave links allow more bits per Hz for any given spectrum, antenna size and transmitter power. Assignment of different availability of class to different types of service over a single radio link allows more efficient planning of link capacity. Voice and real-time video applications will continue to be assigned for 24*7 hours. By utilization of Adaptive Coding & modulation (ACM) some of the data under fading conditions are dropped and allow the constant flow of high priority bits at all time, hence the overall radio capacity is maximized at no extra cost.
1.3.5.6 Phase-shift keying

Three major classes of digital modulation techniques are

- Amplitude-shift keying (ASK)
- Frequency-shift keying (FSK)
- Phase-shift keying (PSK)

Phase-shift keying (PSK) is a digital modulation scheme which conveys data by modulating, the phase of a reference signal (the carrier wave). The digital modulation scheme uses a finite number of distinct signals to represent digital data. PSK uses a finite number of phases, which are assigned a unique pattern of binary digits.

The demodulator designed specifically for the symbol set is used by the modulator to determine the phase of the received signal and maps it back to the symbols, thus recovers the original data. This requires the receiver to be able to compare the phase of the received signal to a reference signal.

The demodulator determines the changes in the phase of the received signal rather than the phase itself. If the scheme depends on the difference between successive phases, it is termed differential phase-shift keying (DPSK). DPSK can be significantly simpler to implement than ordinary PSK as there is no need for the demodulator to have a copy of the reference signal to determine the exact phase of the received signal (it is a non-coherent scheme).
a. **Working procedure of PSK**

All convey data by changing some aspect of a base signal, the carrier wave (usually a sinusoid), in response to a data signal. In the case of PSK, the phase is changed to represent the data signal. The two fundamental ways of utilizing the phase of a signal are:

- View the phase itself as conveying the information.
- View the change in the phase as conveying information.

A convenient way to represent PSK schemes is on the constellation diagram. This shows the points in the complex plane where the real and imaginary axes are termed the in-phase and quadrature axes respectively due to their 90° separation. The amplitude of each point along the in-phase axis is used to modulate a cosine (or sine) wave and the amplitude along the quadrature axis to modulate a sine (or cosine) wave (Kamilo Feher 2004, Riaz Ahamed 2005).

b. **Quadrature Phase-Shift Keying (QPSK)**

The QPSK can be used to double the data rate compared with a BPSK system while maintaining the same bandwidth of the signal or the data rate of BPSK with half bandwidth needed. The BER of QPSK is exactly the same as the BER of BPSK.

QPSK transmits twice the data rate in a given bandwidth compared to BPSK at the same BER. The engineering penalty is that QPSK transmitters and receivers are more complicated than the ones for BPSK were explained by Riaz (2005).
c. Higher-order PSK

Bit Error Rate

For the general M-PSK there is no simple expression for the symbol-error probability if $M > 4$. Unfortunately, it can be obtained as

$$P_s = 1 - \int_{\pi}^{0} \frac{M}{M} p_{\phi}(\theta_r) \, d\theta_r$$

(1.1)

Where

$$p_{\phi}(\theta_r) = \frac{1}{2\pi} e^{-2\sin^2 \phi} \int_{0}^{\infty} V e^{-(V \sqrt{\gamma \cos r})^2/2} \, dv,$$

(1.2)

$$V = \sqrt{r_1^2 + r_2^2}$$

(1.3)

$$\theta_r = \tan^{-1}(r_2/r_1),$$

(1.4)

$$\gamma_s = \frac{E_s}{N_0}$$

and

(1.5)

$$T_1 \sim N \left( \sqrt{E_s N_0/2} \right) \text{ and}$$

(1.6)

$$T_2 \sim N(0, N_0/2) \text{ are jointly Gaussian}$$

(1.7)

This is approximated for high $M$ and high $E_b/N_0$ by:

$$P_s \approx 2Q \left( \sqrt{\frac{2\gamma_s}{M}} \sin \frac{\pi}{M} \right)$$

(1.8)

The bit-error probability for M-PSK can be determined only if the bitmapping is known. However, when gray coding is used, the most
probable error from one symbol to the next produces only a single bit-error is given by

\[ P_b = \frac{1}{k} P_s \] (1.9)

### 1.3.5.7 Quadrature amplitude modulation

**Quadrature amplitude modulation (QAM)** is a combination of both analog and digital modulation scheme.

It conveys two analog message signals or two digital bit streams by changing (modulating) the amplitudes of two carrier waves using the amplitude-shift keying (ASK) digital modulation scheme or amplitude modulation (AM) analog modulation scheme was explained in Figure 1.5.

![Digital-to-Analog Modulation Techniques](image)

**Figure 1.5 Types of digital-to-analog modulation**

In Analog QAM the two carrier waves, usually sinusoids are out of phase with each other by 90° and are thus called quadrature carriers or quadrature components. The modulated waves are summed and the resulting
waveform is a combination of both phase-shift keying (PSK) and amplitude-shift keying (ASK) or (in the analog case) of phase modulation (PM) and amplitude modulation (AM).

In the digital QAM, a finite number of at least two phases and at least two amplitudes are used. PSK modulators are often designed using the QAM principle, but are not considered as QAM since the amplitude of the modulated carrier signal is constant. QAM is used extensively as a modulation scheme for digital telecommunication systems. Arbitrarily high spectral efficiency is achieved with QAM by selection of suitable constellation size, limited only by the noise level and linearity of the communications channel (Riaz Ahamed 2005, Kamilo Feher 2004 and Chaehag yi 2008).

a. Digital QAM

Like other modulation schemes, QAM conveys data by changing some of the aspects of the carrier signal, or the carrier wave, (usually a sinusoid) in response to the data signal. In the case of QAM, the amplitude of two waves, 90° out-of-phase with each other (in quadrature) are changed (modulated or keyed) to represent the data signal. Amplitude modulating two carriers in quadrature can be equivalently viewed as both amplitude modulation and phase modulation of single carrier.

Phase modulation (analog PM) and phase-shift keying (digital PSK) are be regarded as a special case of QAM, where the magnitude of the modulating signal remains constant, with the change of phase (Riaz Ahamed 2005, Chaehag yi 2008).
1.3.5.8 Continuous phase modulation (CPM)

In contrast to other coherent digital phase modulation techniques where the carrier phase abruptly resets to zero at the start of every symbol (e.g. m-aryPSK), in CPM the carrier phase is modulated in a continuous manner. For instance with QPSK the carrier instantaneously jumps from a sine to a cosine (i.e. a 90 degree phase shift) whenever one of the two message bits of the current symbol differs from the two message bits of the previous symbol. This discontinuity requires large percentage of the power to occur outside the intended band, (high fractional out-of-band power) leading to poor spectral efficiency. Furthermore CPM is typically implemented as a constant-envelope waveform i.e. the transmitted carrier power is constant. Therefore in CPM, the phase continuity yields high spectral efficiency and the constant envelope yields excellent power efficiency. The primary drawback is the high implementation complexity required for an optimal receiver and explained by Dr.Kamilo Feher (2004).

a. Phase memory

Each symbol is modulated by gradually changing the phase of the carrier from the starting value to the final value over the symbol duration. The modulation and demodulation of CPM is complicated by the fact that the initial phase of each symbol is determined by the cumulative total phase of all previous transmitted symbols which is called as phase memory. Therefore the optimal receiver cannot make decisions on any isolated symbol without taking the entire sequence of transmitted symbols into account. This requires a Maximum Likelihood Sequence Estimator (MLSE), which is efficiently implemented using the Viterbi algorithm and explained by Kamilo Feher (2004).
b. **Phase trajectory**

Minimum-shift keying (MSK) is another name for CPM with excess bandwidth of 1/2 and linear phase trajectory. Although the linear phase trajectory is continuous, it is not smooth since the derivative of the phase is not continuous. The spectral efficiency of CPM is further improved by using a smooth phase trajectory. This is typically accomplished by filtering the phase trajectory prior to modulation by use of raised cosine or a Gaussian. The raised cosine filter has zero crossings offset by exactly one symbol time and so it can yield a full-response CPM waveform to prevent ISI. It was explained by Kamilo Feher (2004).

c. **Working procedure of CPM**

The CPM modulator baseband block modulates using continuous phase modulation. The output is a baseband representation of the modulated signal. The m-ary parameter ‘m’ is the size of the input alphabet. (‘m’ must have the form $2^k$ for some positive integer K).

Continuous phase modulation uses pulse shaping to smooth the phase transitions of the modulated signal.

d. **MSK**

MSK is a special case of OK-QPSK or form of FSK modulation. The baseband signal is filtered sinusoidal produces transition from one binary state to another.

MSK is the binary modulation technique with symbol interval $T_b$, frequency deviation $\pm 1/4T_b$ and phase continuity of the modulated RF carrier at the bit transitions. RF phase varies linearly exactly +90 degrees with
respect to the carrier over one bit period $T_b$ was explained by Kamilo Feher (2004).

e. GMSK

The use of a premodulation low pass filter (Gaussian characteristics) with the MSK approach achieves the constant envelope in addition to spectral containment. This modulation scheme is known as GMSK. The GMSK filter is used to suppress out of band and adjacent channel interference. GMSK provides high spectrum efficiency, constant amplitude, enhances utilization of class C power amplifiers, thus minimizing power consumption, weight and cost. It was explained by Kamilo Feher (2004).

f. TFM

In MSK even though the phase continuity is achieved the derivative of the phase is still discontinued. If the phase change is made still smoother, a much narrow spectrum is achieved. A scheme involving prefiltering combined with an algorithm for selecting the carrier phase shift according to original data values is developed. The TFM has spectral containment characteristics similar to GMSK and explained by Kamilo Feher (2004).

1.3.5.9 MHPM

Multi-h phase-coded modulation (MHPM) is a bandwidth efficient modulation scheme which offers substantial coding gain over conventional digital modulations. A new concept of MHPM with asymmetric modulation indices corresponding to the bipolar data +1 and -1 is considered. The
performance improvements on the error probability over conventional MHPM with essentially no bandwidth expansion are achieved.

Multi-h phase-coded modulation (MHPM), described in detail by Anderson and Taylor provides efficient signaling schemes for the transmission of digital data as compared to techniques such as minimum shift keying (MSK) or quaternary phase shift keying (QPSK). In the MHPM schemes, cyclically varying modulation indices are used in a prescribed manner, such that the transmitted signal has phase slope variation changing from one symbol interval to the next in response to the data symbols being transmitted. The delays in the merge of neighboring phase trellis paths thus results in longer minimum Euclidean distances for MHPM schemes than those for MSK and hence provide high coding gain. In practice, the modulation indices for MHPM are always restricted to be multiples of $1/q$, where $q$ was an integer, and the finite number of phase states can therefore be used to demodulate the data in the receiver. For MHPM asymmetric modulation indices corresponding to the bipolar data $+1$ and $-1$ are used as compared to the symmetric indices used in conventional MHPM schemes. In this new approach, the modulation indices $h+i$ for the data $+1$ and $h–J$ for the data $-1$ are not necessarily equal, more phase states and better flexibilities are therefore available for the designers to optimize the system performance and it was explained by Hong-Kuang Hwang (2009) Kevin C. Kreitzer (1999).

1.3.6 Rayleigh Fading Channel

The requirements are many scatters availability and non availability of line of sight between the transmitter and receiver i.e. (many buildings and other objects attenuate, reflect, refract, and diffract the signal). The path between the base station and mobile station of terrestrial mobile communication is characterized by various obstacles and reflections.
The radio waves transmitted from the base station radiates in all directions including reflected waves, diffracted wave, scattering wave and the direct wave from the base station to the mobile station. Since the path length of the direct, reflected, diffracted, and scattering waves are different, the time taken to reach the mobile station is different for scattered waves.

The reception environment characterized by superposition of delayed waves is called a multipath propagation environment. In a multipath propagation environment, the total received signal is the vector sum of individually delayed signals.

Further more, time variance of the channel is due to Doppler spread, and realized by fast fading or slow fading. In the frequency domain, signal distortion due to fast fading increases as the Doppler spread increases, thus causing the channel impulse response to change rapidly within the symbol duration (Riaz Ahamed 2005, Samreen Amir 2011, Mohamed Slimalouini 2000).

### 1.3.6.1 Channel coding

The task of channel coding is to encode the information sent over a communication channel in such a way that in the presence of channel noise, errors can be detected or corrected. The two types of coding methods are

**Backward error correction (BEC)** requires only error detection: if an error is detected, the sender is requested to retransmit the message. The method is simple and sets lower requirements on the code’s error-correcting properties and on the other hand requires duplex communication and causes undesirable delays in transmission
Forward error correction (FEC) requires that the decoder is capable of correcting a certain number of errors, i.e. it should be capable of locating the positions where the errors occurred. Since FEC codes require only simplex communication, they are especially attractive in wireless communication systems, will help to improve the energy efficiency of the system (Tom Richardson 2008, Yongzhe Xie 2003, Jagan Mohan 2010, Cheng Yang Li 2003).

1.3.6.2 The search for good codes

For around 45 years the code and information theorists invented several classes of codes offering good performance. They were Block codes (memory less) such as BCH, Reed-Solomon codes Convolutional (with memory) codes Concatenated codes, a mixture of the two above etc. (Tom Richardson 2008, Yongzhe Xie 2003, Jagan Mohan 2010, Cheng Yang Li 2003).

1.3.6.3 Concept of conventional convolutional codes

Convolutional codes differ from block codes in the sense that they will not break the message stream into fixed-size blocks. Instead, redundancy is added continuously to the whole stream. The encoder keeps M previous input bits in memory. Each output bit of the encoder depends on the current input bit as well as the M stored bits.

The encoder produces two output bits per every input bit, defined by the equations

\[ y_{1,i} = x_i + x_{i-1} + x_{i-3}, \]  

(1.32)

\[ y_{2,i} = x_i + x_{i-2} + x_{i-3}. \]  

(1.33)
For the encoder, $M = 3$, since the $i^{th}$ bits of output depend on input bit $i$, as well as three previous bits $i-1$, $i-2$, $i-3$. The encoder is nonsystematic.

An important parameter of a channel code is the code rate. If the input size (or message size) of the encoder is ‘$k$’ bits and the output size (the code word size) is ‘$n$’ bits, and then the ratio ‘$k/n$’ is called the code rate ‘$r$’. Since sample convolution encoder produces two output bits for every input bit, its rate is $1/2$. Code rate express the amount of redundancy in the code.

Finally, the Hamming weight or simply the weight of a code word was the number of non-zero symbols in the code word. In the case of binary codes, the weight of a code word was the number of ones in the word.

1.3.6.4 Necessity for better codes

Design of channel code is always a tradeoff between energy efficiency and bandwidth.

Efficiency: Codes with lower rate (i.e. bigger redundancy) usually correct more errors. If more errors are to be corrected, the communication system shall operate with a lower transmit power and higher data rate. The above property makes the code energy efficient. On the other hand, low-rate codes have a large overhead and consume heavy bandwidth. The decoding complexity grows exponentially with code length and long (low-rate) codes set high computational requirements to conventional decoders.

Shannon Capacity: For every combination of bandwidth (W), channel type, signal power (S) and received noise power (N), there is a theoretical upper limit on the data transmission rate (R), for which error-free
data transmission is possible. This limit is called channel capacity or also Shannon capacity. The formula is given as

\[
R < W \log_2 \frac{1+S/N}{2} \text{ bits/second.}
\]  

(1.10)

Instead, error-free data transmission is interpreted in a way that the bit error probability can be an arbitrarily small constant. The bit error probability, or bit error rate (BER) used in benchmarking is often chosen to be $10^{-5}$ or $10^{-6}$.

Hence, new codes are sought that allow easy decoding. One way of making the task of the decoder easier is use of code with mostly heavy weight code words. Heavy weight code words, i.e. code words containing more ones and less zeros, can be distinguished more easily. Another strategy involves combining simple codes in a parallel fashion, so that each part of the code can be decoded separately with less complex decoders and each decoder can gain from information exchange with others. This is called the divide-and-conquer strategy. Thus the concept of turbo codes is based on divide and conquers strategy. Tom Richardson (2008).

### 1.3.7 Main Characteristics of Turbo Codes

**Principle of Turbo Codes**

It is theoretically possible to approach the Shannon limit by use of block code with large block length or a convolutional code with a large constraint length. The processing power require to decode such long codes made this approach impractical. Turbo codes overcome this limitation by use of recursive coders and iterative soft decoders. The recursive coder makes convolutional codes with short constraint length, and the iterative soft decoder progressively improves the estimate of the received message.
1.3.7.1 Turbo encoding

Recursive Systematic Convolution Code (RSC)

Convolution encoding results by passing of encrypted information through a linear shift register as shown in Figure 1.6 below. The encoder shown here is nonsystematic because no version of the encoded input is part of the output. Convolution encoder is represented by their generator polynomials. For the encoder below, \( g(1) = [111] \) and \( g(2) = [101] \).

![Figure 1.6 Constraint length K = 2 convolutional encoder](image)

Convolutional encoding is a continuous process where the output depends on the \( K \) previous inputs of the encoder. The linear shift register introduces a deterministic component to the randomly generated input. This component can be tracked through a trellis diagram. For Turbo codes, the recursive systematic convolutional codes are chosen as they exhibit better performance at low signal to noise ratios (SNR) and explained by Emilia Kasper and Tom Richardson (2008).

1.3.7.2 Encoding of parallel concatenated convolutional codes

Turbo code was initially presented by Berrou, Glavieux and Thitimajshima in 1993. They are the result of the parallel concatenation of two or more RSC. Here in this case only two RSC are used. The information
is encoded by the first recursive systematic encoder, interleaved and then encoded by the second RSC at the same time. The size of the interleave determines the length of the codeword.

The code’s behavior is described by trellis diagram. In the trellis diagram, all possible transitions between states are shown along with the input and output associated with it. Transitions not drawn on the trellis will not represent valid codeword and therefore classified as errors.

The block MxN interleaver is used. Here the M bits are fed into the interleave column wise and N bits are read out row-wise. The interleaver will then alleviate burst errors by spreading them so that one error occurs every M bits and thus reduce the correlation between the input and output. The presence of the interleaver adds to a difficult trellis termination problem. The trellis of conventional convolution encoder can be terminated by appending a few zeros at the end of the input sequence. For the recursive variety of encoders, the termination bits depend on the state of the encoder as it is forced back to the zero state. Therefore, the tails bits cannot be known until the encoder completely encode the data. Moreover, the additional bits used for trellis termination of RSC #1 is interleaved and therefore useless in terminating RSC #2, thus become data for the latter. One can see how difficult it becomes to successfully compute a sequence of tail bits that would terminate both trellis. One solution is to only terminate the trellis of RSC #1 and leave the other open. One can modify a turbo code with punctured code and puncture pattern decides which parity bits are to be retained after puncturing. Commonly used patterns include selection of xth bit every 2^k parity bits, k > 0. For most rates, when commonly used patterns are applied to both parity sequences, turbo codes exhibit very good performance and explained by Turbo codes at Charles worth (2000), Tom Richardson (2008).
1.3.8 Simulation

For simulation purposes, a complete OFDM WLAN physical layer simulation in Matlab and Simulink software is preferred. The program simulates a 64 subcarrier OFDM system with Turbo code generator rates of 1/2 or 1/3. The system supports 4 modulation schemes, m-aryPSK, m-aryQAM, m-aryCPM and m-aryMHPM. Frequency jitter can be added to the system that supports two channel models, namely AWGN and flat Rayleigh fading. The desired length of the delay spread can be provided as input. The cyclic prefix is chosen and specific average signal to noise ratio is desired.

The experiment is followed up with a study of BER performance of the system in Rayleigh environments. Rayleigh fading emerges when multiple time-shifted or delayed versions of the originally transmitted signal emerge at the receiver.