SIMULATION IN MATLAB AND IMPLEMENTATION IN C LANGUAGE OF AN OFDM MODEM

BY

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INTRODUCTION

The telecommunications industry faces the problem of providing telephone services to rural areas, where the customer base is small, but the cost of installing a wired phone network is very high. One method of reducing the high infrastructure cost of a wired system is to use a fixed wireless radio network. The problem with this is that for rural and urban areas, large cell sizes are required to get sufficient coverage. This presents extra problems as there are long delay times in multipath signal propagation.

Currently Global System for Mobile telecommunications (GSM) technology is being applied to fixed wireless phone systems in rural areas or Australia. However, GSM uses time division multiple access (TDMA), which has a high symbol rate leading to problems with multipath causing inter-symbol interference.

Several techniques are under consideration for the next generation of digital phone systems, with the aim of improving cell capacity, multipath immunity, and flexibility. These include CDMA and COFDM. Both techniques could be applied to provide a fixed wireless system for rural areas. However, each technique has peculiar features, making it more suited for specific applications.

COFDM is currently being used in several new radio broadcast systems including the proposal for high definition digital television (HDTV) and digital audio broadcasting (DAB). However, little research has been done into the use of COFDM as a transmission method for mobile systems.

In CDMA, all users transmit in the same broad frequency band using specialized codes as a basis of channelization. Both the base station as well as the mobile station know codes used to modulate the data sent.

OFDM/COFDM allows many users to transmit in an allocated band, by subdividing the available bandwidth into many narrow bandwidth carriers. Each user is allocated several carriers in which to transmit their data. The transmission is generated in such a way that the carriers used are orthogonal to one another, thus allowing them to be packed together much closer than standard frequency division multiplexing (FDM). This leads to OFDM/COFDM providing a high spectral efficiency.

1.1 Third Generation Wireless Networks:

The expansion of the use of digital networks has led to the need for the design of new higher capacity communications networks. The demand for cellular-type systems in Europe is predicted to be between 15 and 20 million users by the year 2000 [1], and is already over 30 million (1995) in the U.S. [2]. Wireless services have been growing at a rate greater than 50% per year [2], with the current second-generation European digital systems (GSM) being expected to be filled to capacity by the early 2000s[3]. The telecommunications industry is also changing, with a demand for a greater range of services such as video conferencing, Internet services, and data networks, and multimedia. This demand for higher capacity networks has led to the development of third generation telecommunications systems.

One of the proposed third generation telecommunications systems is the Universal Mobile Telecommunications System (UMTS), with the aim of providing more flexibility, higher capacity, and a more tightly integrated service. This section focuses on the services and aims of the UMTS. Other systems around the world are being developed, however many of these technologies are expected to be combined into the UMTS.

The World Wide Web (WWW) has become an important communications media, as its use has increased dramatically over the last few years. This has resulted in an increased demand for computer networking services. In order to satisfy this, telecommunications systems are now being used for computer networking, Internet access and voice communications. A WWW survey revealed that more then 60% of users access the Internet from residential locations [10], where the bandwidth is often limited to 28.8kbps [8]. This restricts the use of the Internet, preventing the use of real time audio and video capabilities. Higher speed services are available, such as integrated-services digital network (ISDN). These provide data rates up to five times as fast, but at a much increased access cost. This has led to the demand of a more integrated service, providing faster data rates, and a more universal interface for a variety of services. The emphasis has shifted away from providing a fixed voice service to providing a general data connection that allows for a wide variety of applications, such as voice, Internet access, computer networking, etc.

The increased reliance on computer networking and the Internet has resulted in an increased demand for connectivity to be provided “any where, any time”, leading to an increase in the demand for wireless systems. This demand has driven the need to develop new higher capacity, high reliability wireless telecommunications systems. The development and deployment of third generation telecommunication systems aim to overcome some of the downfalls of current wireless systems by providing a high capacity, integrated wireless network. There are currently several third generation wireless standards, including UMTS, cdmaOne, IMT 2000, and IS-95 [10].

1.2 Propagation Characteristics of mobile radio channels:

In an ideal radio channel, the received signal would consist of only a single direct path signal, which would be a perfect reconstruction of the transmitted signal. However in a real channel, the signal is modified during transmission in the channel. The received signal consists of a combination of attenuated, reflected, refracted, and diffracted replicas of the transmitted signal. On top of all this, the channel adds noise to the signal and can cause a shift in the carrier frequency if the transmitter or receiver is moving (Doppler Effect). Understanding of these effects on the signal is important because the performance of a radio system is dependent on the radio channel characteristics.

1.2.1 Attenuation:

Attenuation is the drop in the signal power when transmitting from one point to another. It can be caused by the transmission path length, obstructions in the signal path, and multipath effects. Figure 2 shows some of the radio propagation effects that cause attenuation. Any objects which obstruct the line of sight signal from the transmitter to the receiver, can cause attenuation.

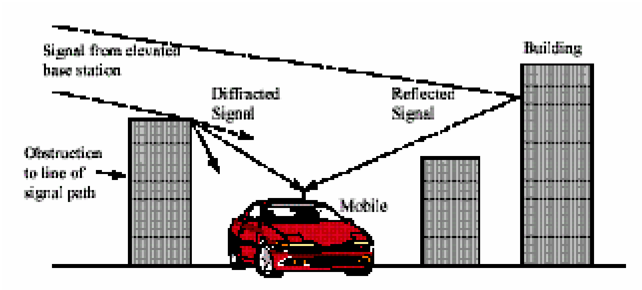


FIG 1.1

Shadowing of the signal can occur whenever there is an obstruction between the transmitter and receiver. It is generally caused by buildings and hills, and is the most important environmental attenuation factor.

Shadowing is most severe in heavily built up areas, due to the shadowing from buildings. However, hills can cause a large problem due to the large shadow they produce. Radio signals diffract off the boundaries of obstructions, thus preventing total shadowing of the signals behind hills and buildings. However, the amount of diffraction is dependent on the radio frequency used, with low frequencies diffracting more then high frequency signals. Thus high frequency signals, especially, Ultra High Frequencies (UHF), and microwave signals require line of sight for adequate signal strength. To overcome the problem of shadowing, transmitters are usually elevated as high as possible to minimize the number of obstructions. Typical amounts of variation in attenuation due to shadowing are shown .

Description	Typical Attenuation due to Shadowing
Heavily built-up urban center	20dB variation from street to street
Sub-urban area (fewer large buildings)	10dB greater signal power then built-up urban center
Open rural area	20dB greater signal power then sub-urban areas
Terrain irregularities and tree foliage	3-12dB signal power variation

Table 1

Shadowed areas tend to be large, resulting in the rate of change of the signal power being slow. For this reason, it is termed *slow-fading* or *log-normal shadowing*.

1.2.2 Multipath Effects

1.2.2.1 Rayleigh fading

In a radio link, the RF signal from the transmitter may be reflected from objects such as hills, buildings, or vehicles. This gives rise to multiple transmission paths at the receiver. Figure 3 show some of the possible ways in which multipath signals can occur.
The relative phase of multiple reflected signals can cause constructive or destructive interference at the receiver. This is experienced over very short distances (typically at half wavelength distances), thus is given the term *fast fading*. These variations can vary from 10-30dB over a short distance. Figure 4 shows the level of attenuation that can occur due to the fading.

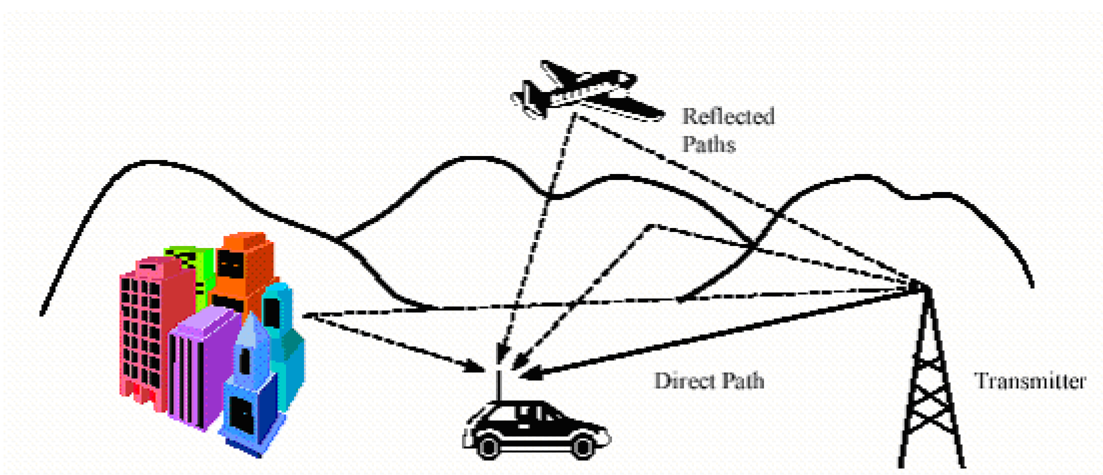


Fig 1.2 Multipath Signals

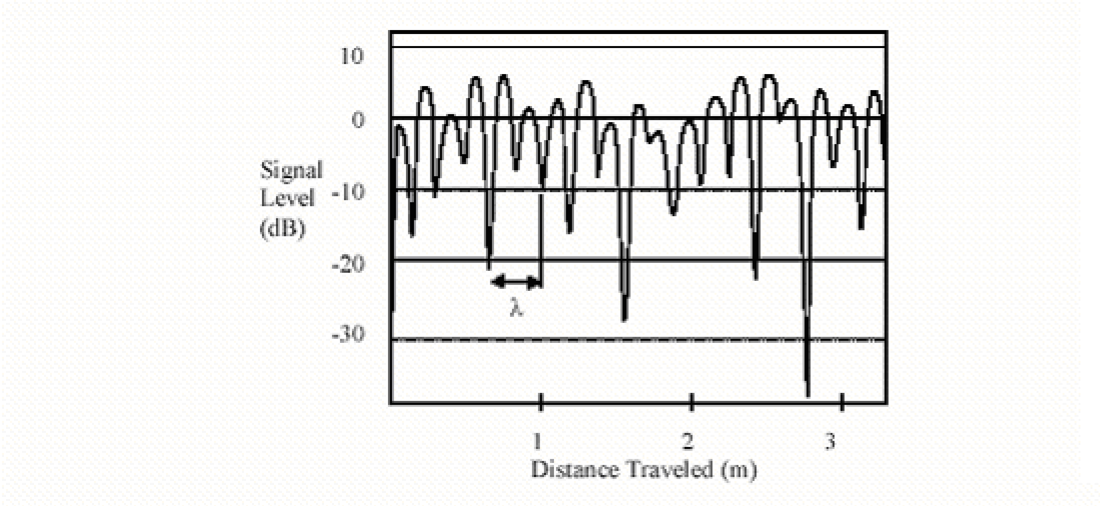


fig (1.3) Typical Rayleigh fading while the mobile unit is moving(for at 900 Mhz)

The Rayleigh distribution is commonly used to describe the statistical time varying nature of the received signal power. It describes the probability of the signal level being received due to fading. Table 7 shows the probability of the signal level for the Rayleigh distribution.

Signal Level (dB about median)	% Probability of Signal Level being less then the value given
10	99
0	50
-10	5
-20	0.5
-30	0.05

Table 2

1.2.2.2 Frequency Selective Fading:

In any radio transmission, the channel spectral response is not flat. It has dips or fades in the response due to reflections causing cancellation of certain frequencies at the receiver. Reflections off near-by objects (e.g. ground, buildings, trees, etc) can lead to multipath signals of similar signal power as the direct signal. This can result in deep nulls in the received signal power due to destructive interference.

For narrow bandwidth transmissions if the null in the frequency response occurs at the transmission frequency then the entire signal can be lost. This can be partly overcome in two ways.

By transmitting a wide bandwidth signal or spread spectrum as CDMA, any dips in the spectrum only result in a small loss of signal power, rather than a complete loss. Another method is to split the transmission up into many small bandwidth carriers, as is done in a COFDM/OFDM transmission. The original signal is spread over a wide bandwidth thus; any nulls in the spectrum are unlikely to occur at all of the carrier frequencies. This will result in only some of the carriers being lost, rather then the entire signal. The information in the lost carriers can be recovered provided enough forward error corrections are sent.

1.2.2.3 Delay Spread:

`The received radio signal from a transmitter consists of typically a direct signal, plus reflections of object such as buildings, mountings, and other structures. The reflected signals arrive at a later time than the direct signal because of the extra path length, giving rise to a slightly different arrival time of the transmitted pulse, thus spreading the received energy. Delay spread is the time spread between the arrival of the first and last multipath signal seen by the receiver.

In a digital system, the delay spread can lead to inter-symbol interference. This is due to the delayed multipath signal overlapping with the following symbols. This can cause significant errors in high bit rate systems, especially when using time division multiplexing (TDMA). Figure 5 shows the effect of inter-symbol interference due to delay spread on the received signal. As the transmitted bit rate is increased the amount of inter-symbol interference also increases. The effect starts to become very significant when the delay spread is greater then ~50% of the bit time.

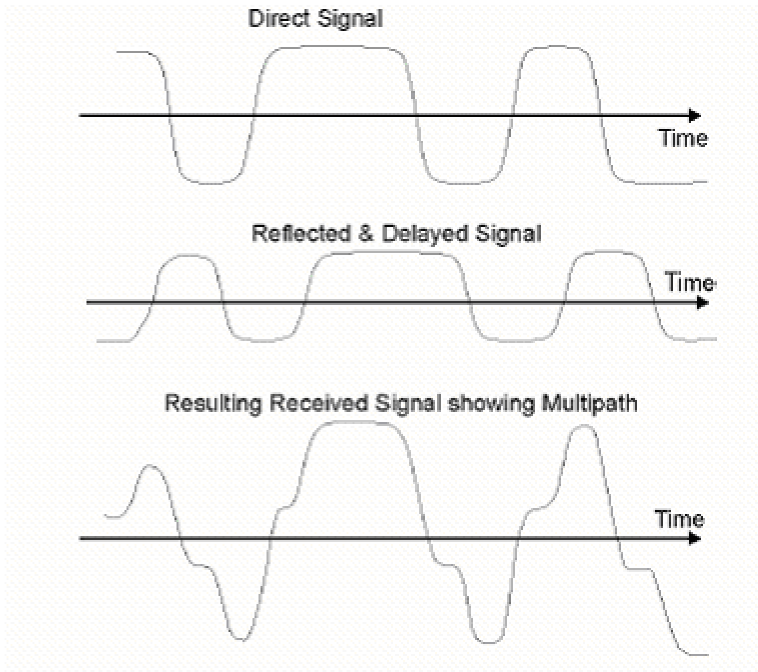


fig (1.4) Multipath Delay Spread

Table 8 shows the typical delay spread that can occur in various environments. The maximum delay spread in an outdoor environment is approximately 20usec, thus significant inter symbol interference can occur at bit rates as low as 25kbps.

Environment or cause	Delay Spread	Maximum Path Length Difference
Indoor (room)	40nsec - 200nsec	12m - 60 m
Outdoor	1usec - 20usec	300m - 6km

Table 3

Inter-symbol interference can be minimized in several ways. One method is to reduce the symbol rate by reducing the data rate for each channel (i.e. split the bandwidth into more channels using frequency division multiplexing). Another is to use a coding scheme which is tolerant to inter-symbol interference such as CDMA.

1.2.3 Doppler Shift:

When a wave source and a receiver are moving relative to one another the frequency of the received signal will not be the same as the source. When they are moving toward each other the frequency of the received signal is higher then the source, and when they are approaching each other the frequency decreases. This is called the Doppler Effect. An example of this is the change of pitch in a car's horn as it approaches then passes by. This effect becomes important when developing mobile radio systems.

The amount the frequency changes due to the Doppler Effect depends on the relative motion between the source and receiver and on the speed of propagation of the wave. The Doppler shift in frequency can be written:

$$\Delta f \approx \pm f_o \frac{v}{c}$$

Where ***.f*** is the change in frequency of the source seen at the receiver, ***fo*** is the frequency of the source, *v* is the speed difference between the source and transmitter, and *c* is the speed of light.

1.3 Multiple Access Techniques:

Multiple access schemes are used to allow many simultaneous users to use the same fixed bandwidth radio spectrum. In any radio system, the bandwidth which is allocated to it is always limited. For mobile phone systems the total bandwidth is typically 50MHz, which is split in half to provide the forward and reverse links of the system. Sharing of the spectrum is required in order increase the user capacity of any wireless network. FDMA, TDMA and CDMA are the three major methods of sharing the available bandwidth to multiple users in wireless system. There are many extensions, and hybrid techniques for these methods, such as OFDM, and hybrid TDMA and FDMA systems. However, an understanding of the three major methods is required for understanding of any extensions to these methods.

1.3.1 Orthogonal Frequency Division Multiplexing:

Orthogonal Frequency Division Multiplexing (OFDM) is a multicarrier transmission technique, which divides the available spectrum into many carriers, each one being modulated by a low rate data stream. OFDM is similar to FDMA in that the multiple user access is achieved by subdividing the available bandwidth into multiple channels, which are then allocated to users. However, OFDM uses the spectrum much more efficiently by spacing the channels much closer together. This is achieved by making all the carriers orthogonal to one another, preventing interference between the closely spaced carriers.

Coded Orthogonal Frequency Division Multiplexing (COFDM) is the same as OFDM except that forward error correction is applied to the signal before transmission. This is to overcome errors in the transmission due to lost carriers from frequency selective fading, channel noise and other propagation effects. For this discussion the terms OFDM and COFDM are used interchangeably, as the main focus of this thesis is on OFDM, but it is assumed that any practical system will use forward error correction, thus would be COFDM.

In FDMA each user is typically allocated a single channel, which is used to transmit all the user information. The bandwidth of each channel is typically 10 kHz-30 kHz for voice communications. However, the minimum required bandwidth for speech is only 3 kHz. The allocated bandwidth is made wider then the minimum amount required to prevent channels from interfering with one another. This extra bandwidth is to allow for signals from neighboring channels to be filtered out, and to allow for any drift in the center frequency of the transmitter or receiver. In a typical system up to 50% of the total spectrum is wasted due to the extra spacing between channels. This problem becomes worse as the channel bandwidth becomes narrower, and the frequency band increases.

Most digital phone systems use vocoders to compress the digitized speech. This allows for an increased system capacity due to a reduction in the bandwidth required for each user. Current vocoders require a data rate somewhere between 4-13 kbps [13], with depending on the quality of the sound and the type used. Thus each user only requires a minimum bandwidth of somewhere between 2-7 kHz, using QPSK modulation. However, simple FDMA does not handle such narrow bandwidths very efficiently.

TDMA partly overcomes this problem by using wider bandwidth channels, which are used by several users. Multiple users access the same channel by transmitting in their data in time slots. Thus, many low data rate users can be combined together to transmit in a single channel which has a bandwidth sufficient so that the spectrum can be used efficiently.

There are however, two main problems with TDMA. There is an overhead associated with the change over between users due to time slotting on the channel. A change over time must be allocated to allow for any tolerance in the start time of each user, due to propagation delay variations and synchronization errors. This limits the number of users that can be sent efficiently in each channel. In addition, the symbol rate of each channel is high (as the channel handles the information from multiple users) resulting in problems with multipath delay spread.

OFDM overcomes most of the problems with both FDMA and TDMA. OFDM splits the available bandwidth into many narrow band channels (typically 100-8000). The carriers for each channel are made orthogonal to one another, allowing them to be spaced very close together, with no overhead as in the FDMA example. Because of this there is no great need for users to be time multiplex as in TDMA, thus there is no over head associated with switching between users.

The orthogonality of the carriers means that each carrier has an integer number of cycles over a symbol period. Due to this, the spectrum of each carrier has a null at the centre frequency of each of the other carriers in the system. This results in no interference between the carriers, allowing then to be spaced as close as theoretically possible. This overcomes the problem of overhead carrier spacing required in FDMA.

Each carrier in an OFDM signal has a very narrow bandwidth (i.e. 1 kHz), thus the resulting symbol rate is low. This results in the signal having a high tolerance to multipath delay spread, as the delay spread must be very long to cause significant inter-symbol interference (e.g. > 500usec).

1.3.2 OFDM generation:

To generate OFDM successfully the relationship between all the carriers must be carefully controlled to maintain the orthogonality of the carriers. For this reason, OFDM is generated by firstly choosing the spectrum required, based on the input data, and modulation scheme used. Each carrier to be produced is assigned some data to transmit. The required amplitude and phase of the carrier is then calculated based on the modulation scheme (typically differential BPSK, QPSK, or QAM). The required spectrum is then converted back to its time domain signal using an Inverse Fourier Transform. In most applications, an Inverse Fast Fourier Transform (IFFT) is used. The IFFT performs the transformation very efficiently, and provides a simple way of ensuring the carrier signals produced are orthogonal.

The Fast Fourier Transform (FFT) transforms a cyclic time domain signal into its equivalent frequency spectrum. This is done by finding the equivalent waveform, generated by a sum of orthogonal sinusoidal components. The amplitude and phase of the sinusoidal components represent the frequency spectrum of the time domain signal. The IFFT performs the reverse process, transforming a spectrum (amplitude and phase of each component) into a time domain signal. An IFFT converts a number of complex data points, of length which is a power of 2, into the time domain signal of the same number of points. Each data point in frequency spectrum used for an FFT or IFFT is called a bin.

The orthogonal carriers required for the OFDM signal can be easily generated by setting the amplitude and phase of each bin, then performing the IFFT. Since each bin of an IFFT corresponds to the amplitude and phase of a set of orthogonal sinusoids, the reverse process guarantees that the carriers generated are orthogonal.

Figure 14 shows the setup for a basic OFDM transmitter and receiver. The signal generated is a baseband, thus the signal is filtered, then stepped up in frequency before transmitting the signal.

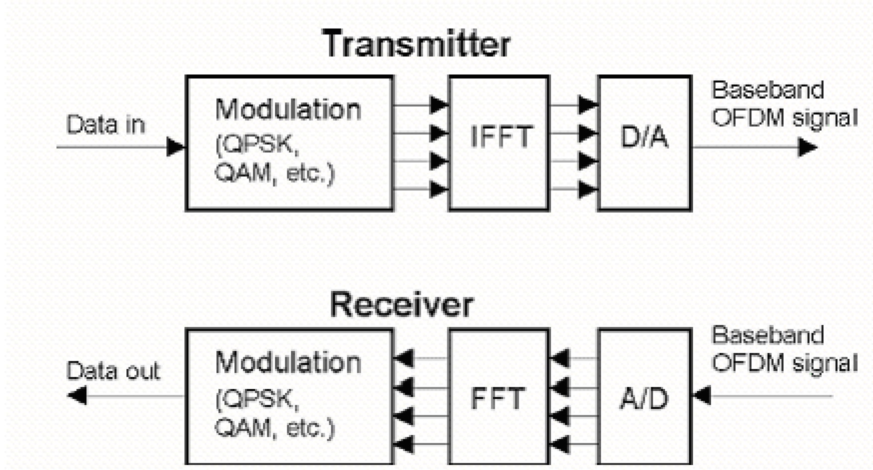


fig (1.5) Basic FFT,OFDM Transmitter and Receiver

1.3.3 Adding Guard Period to OFDM:

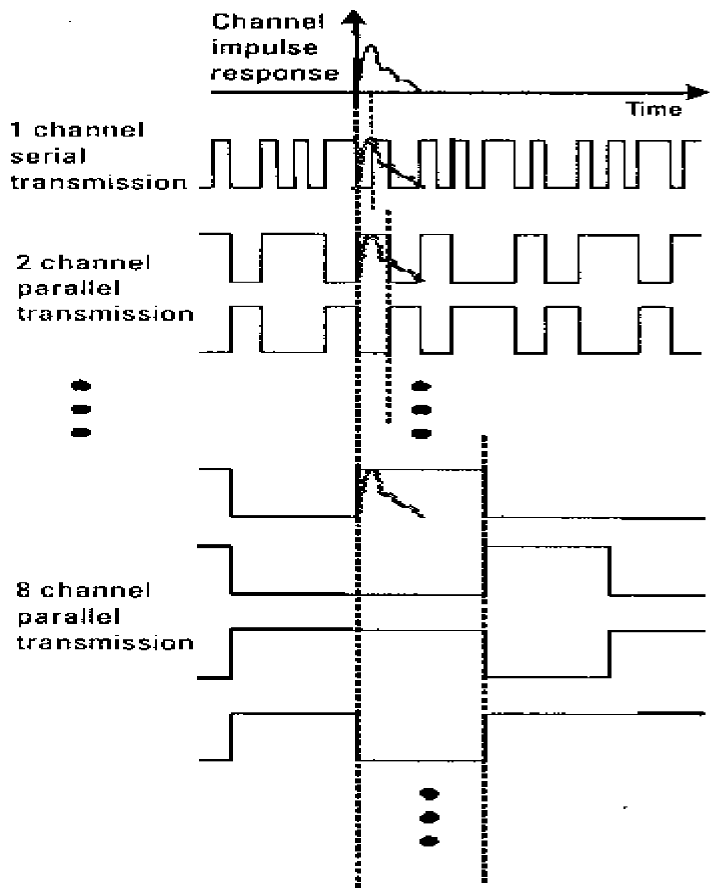
One of the most important properties of OFDM transmissions is the robustness against multipath delay spread. This is achieved by having a long symbol period, which minimizes the inter-symbol interference. The level of robustness, can infact is increased even more by the addition of a guard period between transmitted symbols. The guard period allows time for multipath signals from the pervious symbol to die away before the information from the current symbol is gathered. The most effective guard period to use is a cyclic extension of the symbol. If a mirror in time, of the end of the symbol waveform is put at the start of the symbol as the guard period, this effectively extends the length of the symbol, while maintaining the orthogonality of the waveform. Using this cyclic extended symbol the samples required for performing the FFT (to decode the symbol), can be taken anywhere over the length of the symbol. This provides multipath immunity as well as symbol time synchronization tolerance.

As long as the multipath delay echo's stay within the guard period duration, there is strictly no limitation regarding the signal level of the echoes: they may even exceed the signal level of the shorter path! The signal energy from all paths just adds at the input to the receiver, and since the FFT is energy conservative, the whole available power feeds the decoder. If the delay spread is longer then the guard intervals then they begin to cause intersymbol interference. However, provided the echoes are sufficiently small they do not cause significant problems. This is true most of the time as multipath echoes delayed longer than the guard period will have been reflected of very distant objects.

Other variations of guard periods are possible. One possible variation is to have half the guard period a cyclic extension of the symbol, as above, and the other half a zero amplitude signal. This will result in a signal as shown in Figure 15. Using this method the symbols can be easily identified. This possibly allows for symbol timing to be recovered from the signal, simply by applying envelop detection. The disadvantage of using this guard period method is that the zero period does not give any multipath tolerance, thus the effective active guard period is halved in length. It is interesting to note that this guard period method has not been mentioned in any of the research papers read, and it is still not clear whether symbol timing needs to be recovered using this method.

2.1 Concept of the Parallel Transmission Scheme:

The multipath fading environment in which not only a direct transmission signal but also many reflected signal arrive at the receiver at different times in time domain is characterized by a channel impulse response, which includes the



Fig(2.1)Channel Effect on Serial and Parallel data

information about the relative time. When the delayed signal arrive at the receiver the power of the signal, and its phase are compared to power and phase of direct wave.

Figure 2.1 shows a typical impulse response of multipath fading in the time. From the time domain point of view of many signals with different arrival times, signal power, and phases are at the receiver. From the frequency two main points of view, on the other the multipath fading environment is characterized by the enhancement of some frequencies and the attenuation of others. If there is a mobile reception the relative power levels and attenuations of various reception paths will change with time. A narrow band signal will vary in quality as the peaks and the freq response move around in the frequency domain there will be a noticeable variation in phase response.

The width of frequency band that have high correlations value is called the Coherent bandwidth for a narrow band signal ,distortion is usually minimized if the bandwidth of the signal is less than the coherent bandwidth there is however a significant chance that the signal will be subjected to severe attenuation in some occasion .

A signal that occupies a bandwidth greater than the coherent bandwidth will be subjected to more distortion, but will suffer from less variation in the total received power, even if it is subjected to significance level of multipath fading .so it is necessary to use parallel transmission, in which the transmitted high speed data is converted to parallel data in several channels .these data are multiplexed using several multiplexing techniques to distinguish between the sub-channels.

For a given overall data rate, increasing the parallel transmission channels reduces the data rate, increasing the number of parallel transmission channel reduces the data rate that each individual sub-channel must conveyor in other words lengthens the symbol period.

To distinguish between the sub-channels, frequency division multiplexing (FDM) and code division multiplexing (CDM).the above is called **Multicarrier transmission** and the second method is called **multicode transmission.**

2.2 OFDM (Orthogonal Frequency Division Multiplexing):

Orthogonal Frequency Division Multiplexing (OFDM) is a multicarrier transmission technique used in applications catering to both Wired and Wireless Communications. However, in the wired case, the usage of the term Discrete Multi-Tone is more appropriate. The OFDM technique divides the frequency spectrum available into many closely spaced carriers, which are individually modulated by low-rate data streams. In this sense, OFDM is similar to FDMA (The bandwidth is divided into many channels, so that, in a multi-user environment, each channel is allocated to a user). However, the difference lies in the fact that the carriers chosen in OFDM are much more closely spaced than in FDMA (1kHz in OFDM as opposed to about 30kHz in FDMA), thereby increasing its spectral usage efficiency. The orthogonality between the carriers is what facilitates the close spacing of carriers.

The Orthogonality principle essentially implies that each carrier has a null at the center frequency of each of the other carriers in the system while also

FIG 2.2 Basic OFDM System

maintaining an integer number of cycles over a symbol period.

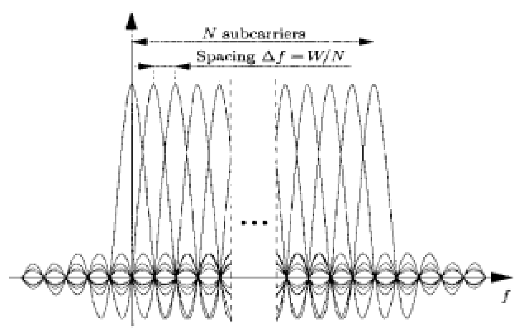
The motivation for using OFDM techniques over TDMA techniques is twofold. First, TDMA limits the total number of users that can be sent efficiently over a channel. In addition, since the symbol rate of each channel is high, problems with multipath delay spread invariably occur. In stark contrast, each carrier in an OFDM signal has a very narrow bandwidth (i.e. 1 kHz); thus the resulting symbol rate is low. This results in the signal having a high degree of tolerance to multipath delay spread, as the delay spread must be very log to cause significant inter-symbol interference.

2.3 OFDM principle “Orthogonality”:

The orthogonality of the carriers means that each carrier has an integer number of cycles over a symbol period.

Due to this, the spectrum of each carrier has a null at the centre frequency of each of the other carriers in the system. This result in no interference between the carriers, allowing them to be spaced as close as possible This overcomes the problem of overhead carrier spacing required in FDMA. To generate OFDM signals successfully the relationship between all carriers must be carefully controlled in order to maintain orthogonality. Figure below is the frequency spectrum depicting the various carriers/channels (used interchangeably). Rectangular windowing of transmitted pulses results in a sinc-shaped frequency response for each channel. As can be seen, whenever any particular carrier frequency attains peak amplitude, the remaining carriers have a null point. Frequency spectrum showing N channels for an OFDM system with N carriers over a bandwidth W.

The system operational principle is that the original bandwidth is divided in a high number of narrow bands, in which mobile channel can be considered non-dispersive. Hence no channel equalizer is required and instead of Implementing by the help of single Fourier transforms (FFT).These OFDM systems often as multitone systems.



Fig(2.3) Orthogonality of the Carriers

2.4 OFDM system model:

To generate OFDM successfully the relationship between all the carriers must be carefully controlled to maintain the orthogonality of the carriers. For this reason, OFDM is generated by firstly choosing the spectrum required, based on the input data, and modulation scheme used. Each carrier to be produced is assigned some data to transmit. The required amplitude and phase of the carrier is then calculated based on the modulation scheme (typically differential BPSK, QPSK, or QAM).

The Fast Fourier Transform (FFT) transforms a cyclic time domain signal into its equivalent frequency spectrum. This is done by finding the equivalent waveform, generated by a sum of orthogonal sinusoidal components. The amplitude and phase of the sinusoidal components represent the frequency spectrum of the time domain signal. The IFFT performs the reverse process, transforming a spectrum (amplitude and phase of each component) into a time domain signal. An IFFT converts a number of complex data points, of length which is a power of 2, into the time domain signal of the same number of points. Each data point in frequency spectrum used for an FFT or IFFT is called a bin.

The orthogonal carriers required for the OFDM signal can be easily generated by setting the amplitude and phase of each bin, then performing the IFFT.

Since each bin of an IFFT corresponds to the amplitude and phase of a set of orthogonal sinusoids, the reverse process guarantees that the carriers generated are orthogonal.

Figure 2.4 shows the setup for a basic OFDM transmitter and receiver. The signal generated is a baseband, thus the signal is filtered, then stepped up in frequency before transmitting the signal.

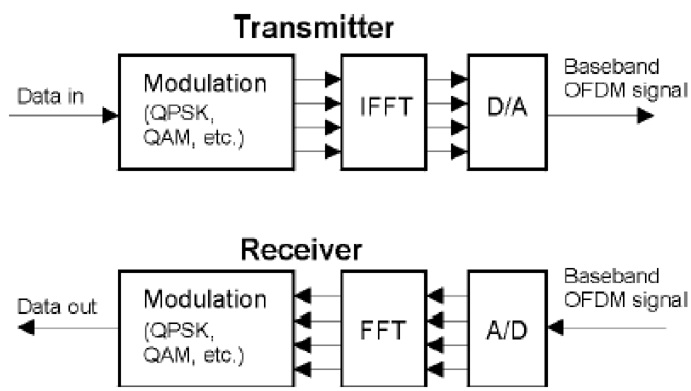


FIG 2.4 Basic OFDM Transmitter and Receiver

2.5 Preliminary Concepts:

When the DFT (Discrete Fourier Transform) of a time signal is taken, the frequency domain results are a function of the *time sampling period* and the *number of samples* as shown in F. The fundamental frequency of the DFT is equal to 1/NT (1/total sample time). Each frequency represented in the DFT is an integer multiple of the fundamental frequency. The maximum frequency that can be represented by a time signal sampled at rate 1/T is f_{max} = 1/2T as given by the Nyquist sampling theorem. This frequency is located in the center of the DFT points. All frequencies beyond that point are images of the representative frequencies. The maximum frequency bin of the DFT is equal to the sampling frequency (1/T) minus one fundamental (1/NT).

The IDFT (Inverse Discrete Fourier Transform) performs the opposite operation to the DFT. It takes a signal defined by frequency components and converts them to a time signal. The parameter mapping is the same as for the DFT. The time duration of the IDFT time signal is equal to the number of DFT bins (N) times the sampling period (T).

It is perfectly valid to generate a signal in the frequency domain, and convert it to a time domain equivalent for practical use. This is how modulation is applied in OFDM.

- The frequency domain is a mathematical tool used for analysis. Anything usable by the real world must be converted into a real, time domain signal.

In practice the Fast Fourier Transform (FFT) and IFFT are used in place of the DFT and IDFT, so all further references will be to FFT and IFFT.

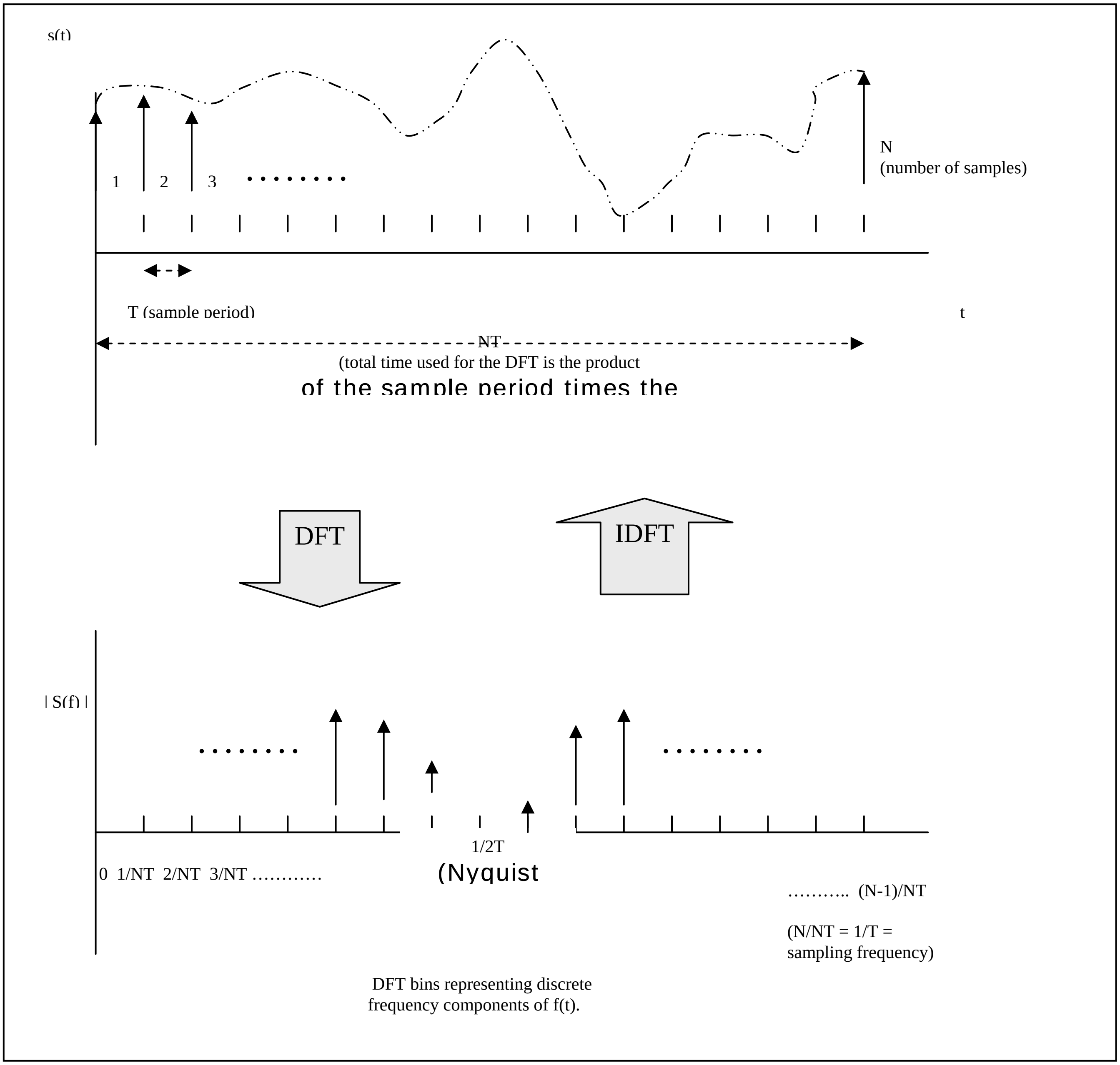


Fig 2.5 Parameter Mapping from Time to Frequency for the DFT

2.5.1 **Definition of Carriers:**

The maximum number of carriers used by OFDM is limited by the size of the IFFT. This is determined as follows:

2.5.1.1 OFDM Carrier Count:

$$N_{carriers} \leq \frac{IFFTsize}{2} - 2 \quad \text{(real - valued time signal)}$$
$$N_{carriers} \leq IFFTsize - 1 \quad \text{(complex - valued time signal)}$$

In order to generate a real-valued time signal, OFDM (frequency) carriers must be defined in complex conjugate pairs, which are symmetric about the Nyquist frequency (f_{max}). This puts the number of potential carriers equal to the IFFT size/2. The Nyquist frequency is the symmetry point, so it cannot be part of a complex conjugate pair. The DC component also has no complex conjugate. These two points cannot be used as carriers so they are subtracted from the total available.

If the carriers are not defined in conjugate pairs, then the IFFT will result in a time domain signal that has imaginary components. This must be a viable option as there are OFDM systems defined with carrier counts that exceed the limit for real-valued time signals. Here, only real-value time signals will be treated, but in order to obtain maximum bandwidth efficiency from OFDM, the complex time signal may be preferred (possibly an analagous situation to QPSK vs. BPSK).

Both IFFT size and assignment (selection) of carriers can be dynamic. The transmitter and receiver just have to use the same parameters. This is one of the advantages of OFDM. Its bandwidth usage (and bit rate) can be varied according to varying user requirements. A simple control message from a base station can change a mobile unit's IFFT size and carrier selection.

2.5.2 Modulation:

Binary data from a memory device or from a digital processing stream is used as the modulating (baseband) signal. The following steps may be carried out in order to apply modulation to the carriers in OFDM:

- combine the binary data into symbols according to the number of bits/symbol selected
- convert the serial symbol stream into parallel segments according to the number of carriers, and form carrier symbol sequences

- apply differential coding to each carrier symbol sequence
- convert each symbol into a complex phase representation
- assign each carrier sequence to the appropriate IFFT bin, including the complex conjugates
- take the IFFT of the result

Fig give an example of modulated OFDM carriers for one symbol period, prior to IFFT. For this example, there are 4 carriers, the IFFT bin size is 64, and there is only 1 bit per symbol. The magnitude of each carrier is 1, but it could be scaled to any value. The phase for each carrier is either 0 or 180 degrees, according to the symbol being sent. The phase determines the value of the symbol (binary in this case, either a 1 or a 0). In the example, the first 3 bits (the first 3 carriers) are 0, and the 4th bit (4th carrier) is a 1.

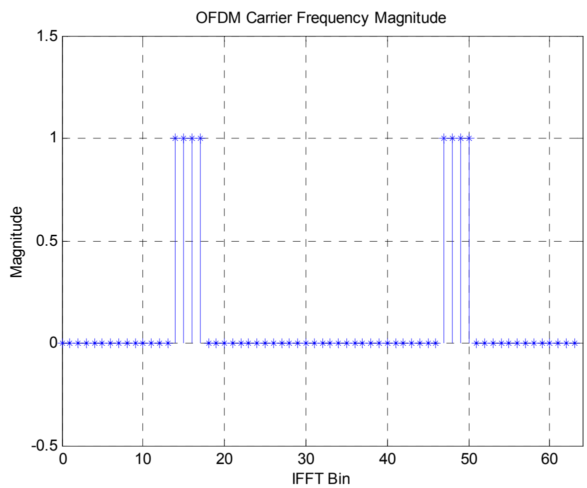
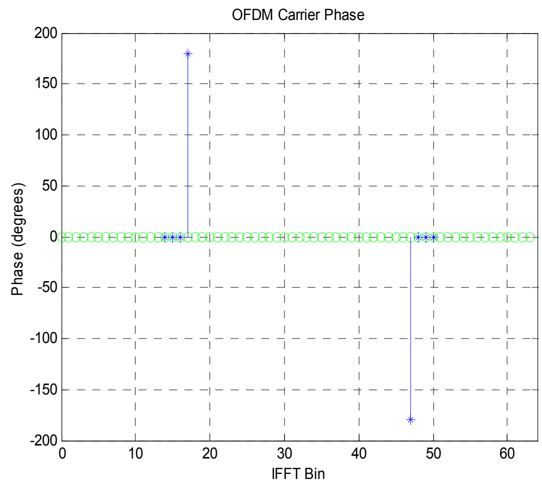


Fig (2.6) OFDM Carrier Magnitude prior to IFFT



Fig(2.7) OFDM Carrier Phase prior to IFFT

The modulated OFDM signal is nothing more than a group of delta (impulse) functions, each with a phase determined by the modulating symbol. In addition, note that the frequency separation between each delta is proportional to 1/N where N is the number of IFFT bins.

2.5.3 OFDM Frequency Domain Representation (one symbol period):

$$S(k) = e^{j\theta_m}\,\delta\left(k - m - \frac{N}{2}\right) + e^{-j\theta_m}\,\delta\left(k + m - \frac{N}{2}\right) \quad \text{single (real) OFDM modulated carrier}$$

k = frequency (0 to N - 1)
m = OFDM carrier frequency
N = IFFT bin size

$$S(k)_{ofdm} = \sum_{m=c_{first}}^{c_{last}} \left[e^{j\theta_m}\,\delta\left(k - m - \frac{N}{2}\right) + e^{-j\theta_m}\,\delta\left(k + m - \frac{N}{2}\right)\right] \quad \text{composite (real) OFDM modulated carriers}$$

c = OFDM carrier (first through last)

After the modulation is applied, an IFFT is performed to generate one symbol period in the time domain. The IFFT result is shown in

It is clear that the OFDM signal has a varying amplitude. It is very important that the amplitude variations be kept intact as they define the content of the signal. If the amplitude is clipped or modified, then an FFT of the signal would no longer result in the original frequency characteristics, and the modulation may be lost.

This is one of the drawbacks of OFDM, the fact that it requires linear amplification. In addition, very large amplitude peaks may occur depending on how the sinusoids line up, so the peak-to-average power ratio is high. This means that the linear amplifier has to have a large dynamic range to avoid distorting the peaks. The result is a linear amplifier with a constant, high bias current resulting in very poor power efficiency.

The one sinusoid with 180 phase shift is clearly visible as is the frequency difference between each of the 4 sinusoids. Note that this figure is 'zoomed' i.e. all 64 point of the IFFT are not shown. In addition, the waveform plots are not very smooth. This is because there are not many samples per cycle for any of the sinusoids.

2.5.4 OFDM Time Domain Representation (one symbol period):

$$s(n) = \sum_{m=c_{first}}^{c_{last}} \sum_{n=0}^{N-1} \cos\left(\frac{2\pi mn}{N} + \theta_m\right)$$

$\mathrm{n} = \text{time sample}$
$\mathrm{m} = \text{OFDM carrier}$
$\mathrm{N} = \text{IFFT bin size}$
$\theta_m = \text{phase modulation for OFDM carrier (m)}$
$c_{first}, c_{last} = \text{OFDM carriers (first and last)}$

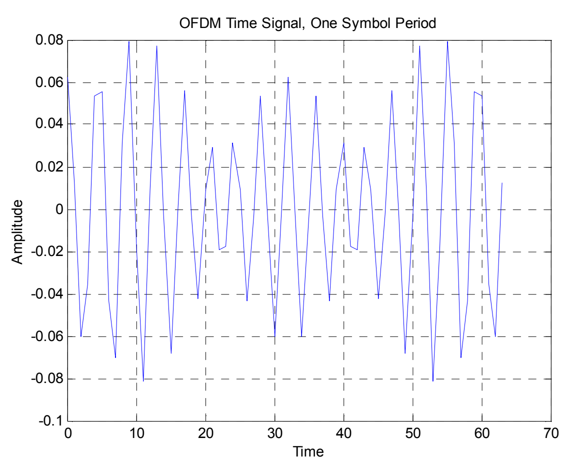


Fig 2.8 OFDM Signal, 1 Symbol Period

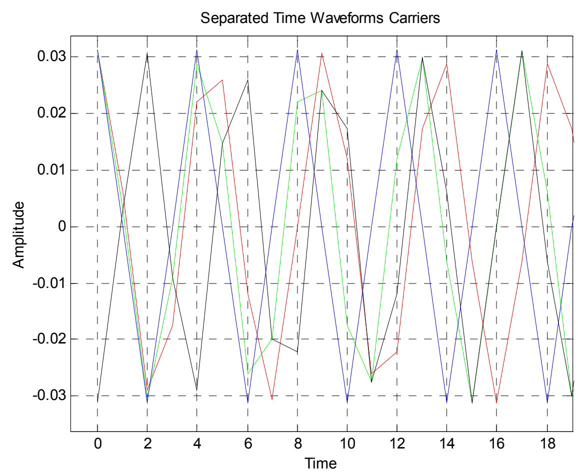


Fig 2.9 Separated Components of the OFDM Time Waveform

2.5.5 Transmission:

The key to the uniqueness and desirability of OFDM is the relationship between the carrier frequencies and the symbol rate. Each carrier frequency is separated by a multiple of 1/NT (Hz). The symbol rate (R) for each carrier is 1/NT (symbols/sec).

The effect of the symbol rate on each OFDM carrier is to add a sin(x)/x shape to each carrier's spectrum. The nulls of the sin(x)/x (for each carrier) are at integer multiples of 1/NT. The peak (for each carrier) is at the carrier frequency k/NT. Therefore, each carrier frequency is located at the nulls for all the other carriers. This means that none of the carriers will interfere with each other during transmission, although their spectrums overlap. The ability to space carriers so closely together is very bandwidth efficient.

shows the spectrum for of an OFDM signal with the following characteristics:

- 1 bit / symbol
- 100 symbols / carrier (i.e. a sequence of 100 symbol periods)
- 4 carriers
- 64 IFFT bins
- spectrum averaged for every 20 symbols (100/20 = 5 averages)

Red diamonds mark all of the available carrier frequencies. The nulls of the spectrums line up with the unused frequencies. The four active carriers each have peaks at carrier frequencies. It is clear that the active carriers have nulls in their spectrums at each of the unused frequencies (otherwise, the nulls would not exist). Although it cannot be seen in the figure, the active frequencies also have spectral nulls at the adjacent active frequencies.

There are 100 symbol periods in the signal. Each symbol period is 64 samples long (100 x 64 = 6400 total samples). Each symbol period contains 4 carriers each of which carries 1 symbol. Each symbol carries 1 bit. It is not currently practical to generate the OFDM signal directly at RF rates, so it must be upconverted for transmission. To remain in the discrete domain, the OFDM could be upsampled and added to a discrete carrier frequency. This carrier could be an intermediate frequency whose sample rate is handled by current technology. It could then be converted to analog and increased to the final transmit frequency using analog frequency conversion methods. Alternatively, the OFDM modulation could be immediately converted to analog and directly increased to the desired RF transmit frequency. Either way, the selected technique would have to involve some form of linear AM (possibly implemented with a mixer).

2.5.6 Reception and Demodulation:

The received OFDM signal is downconverted (in frequency) and taken from analog to digital. Demodulation is done in the frequency domain (just as modulation was). The following steps may be taken to demodulate the OFDM:

- partition the input stream into vectors representing each symbol period
- take the FFT of each symbol period vector
- extract the carrier FFT bins and calculate the phase of each
- calculate the phase difference, from one symbol period to the next, for each carrier
- decode each phase into binary data
- sort the data into the appropriate order

For this example, there are 4 carriers, the IFFT bin size is 64, there is 1 bit per symbol, and the signal was sent through a channel with AWGN having an SNR of 8 dB. The figures show that, under these conditions, the modulated symbols are very easy to recover. These bins are not decoded, so it does not matter, but the result is of interest. Even if the noise is removed from the channel, these phase variations still occur. It must be a result of the IFFT/FFT operations generating very small complex values (very close to 0) for the unused carriers. The phases are a result of these values.

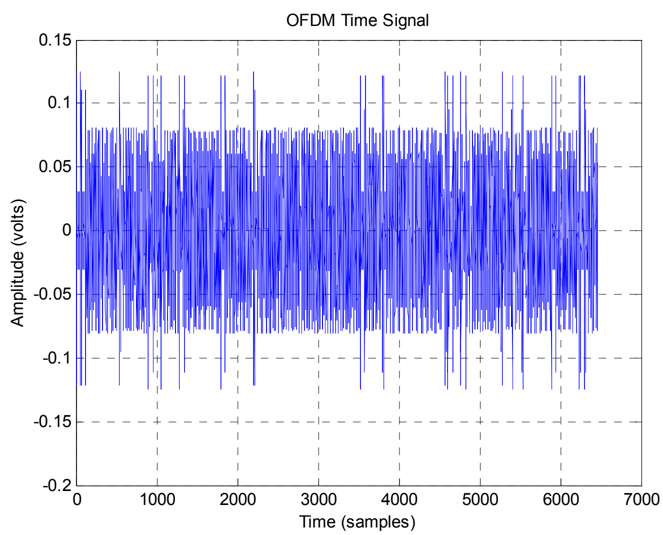


Fig 2.10 OFDM Time Waveform

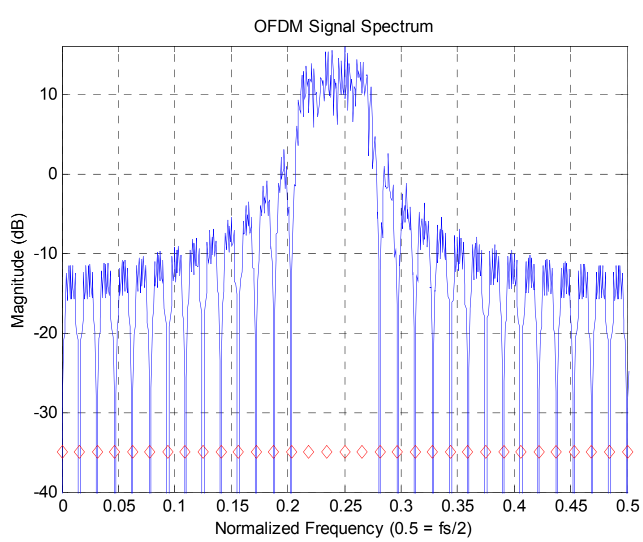


Fig 2.11 : OFDM Spectrum

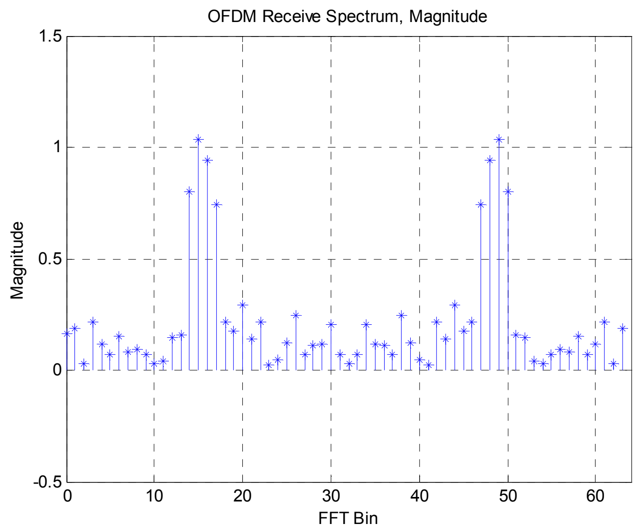


Fig 2.12 OFDM Carrier Magnitude following FFT

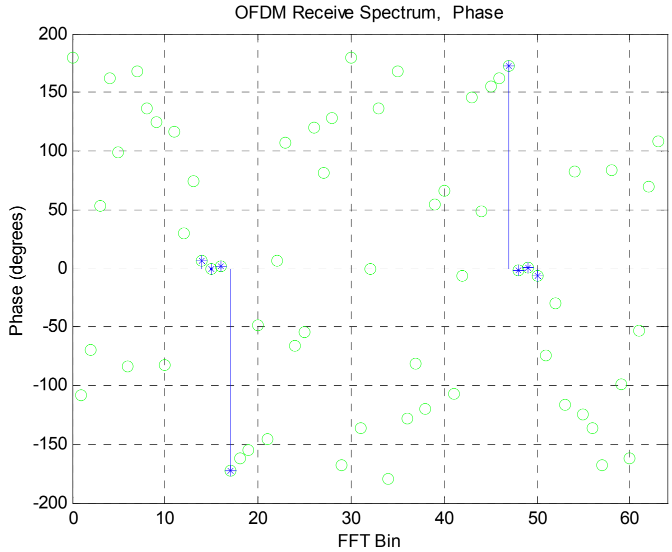


Fig 2.13 OFDM receive spectrum

2.6 Choice of modulation technique:

The first step in deciding on the coding and modulation techniques is determining the number of bits carried by an OFDM symbol. Then, a suitable combination of modulation and coding techniques can be selected to fit the input data rate into the OFDM symbols and, at the same time, satisfying the bit-error rate requirements. The choice of modulation and coding techniques are lot easier now, since each channel is assumed to almost AWGN and one doesn't need to worry about the effects of multi-path delay spread. QAM has the best results in AWGN channel so QAM is used.

2.6.1 QAM-OFDM:

In the OFDM scheme of Figure 2.5 the serial data stream of a traffic channel is passed through a serial-to-parallel converter which splits the data into a number of parallel channels. The data in each channel is applied to a modulator, such that for channels there are N modulators whose carrier frequencies are f_0, f_1, f_{N-1} the difference between adjacent channels is Δf A and the overall bandwidth W of the of the N modulated carriers is N Δf, as shown in Figure

These N modulated carriers are then combined to give an OFDM signal. We may view the serial-to-parallel converter as applying every Nth symbol to a modulator this has the effect of interleaving the symbols into each modulator, e.g. symbols S_o, S_N, S_{2N} are applied to the modulator whose carrier frequency is f_1. At the receiver the received OFDM signal is de multiplexed into N frequency bands, and the N modulated signals are demodulated. The baseband signals are then recombined using a parallel-to-serial converter.

In the more conventional approach the traffic data is applied directly to the modulator operating with a carrier frequency at the centre of the transmission band f_0,….., f_{N-1} i.e. at $(f_{N-1} + f_0)/2$, and the modulated signal occupies the entire bandwidth W . When the data is applied sequentially the effect of a deep fade in a mobile channel is to cause burst errors.

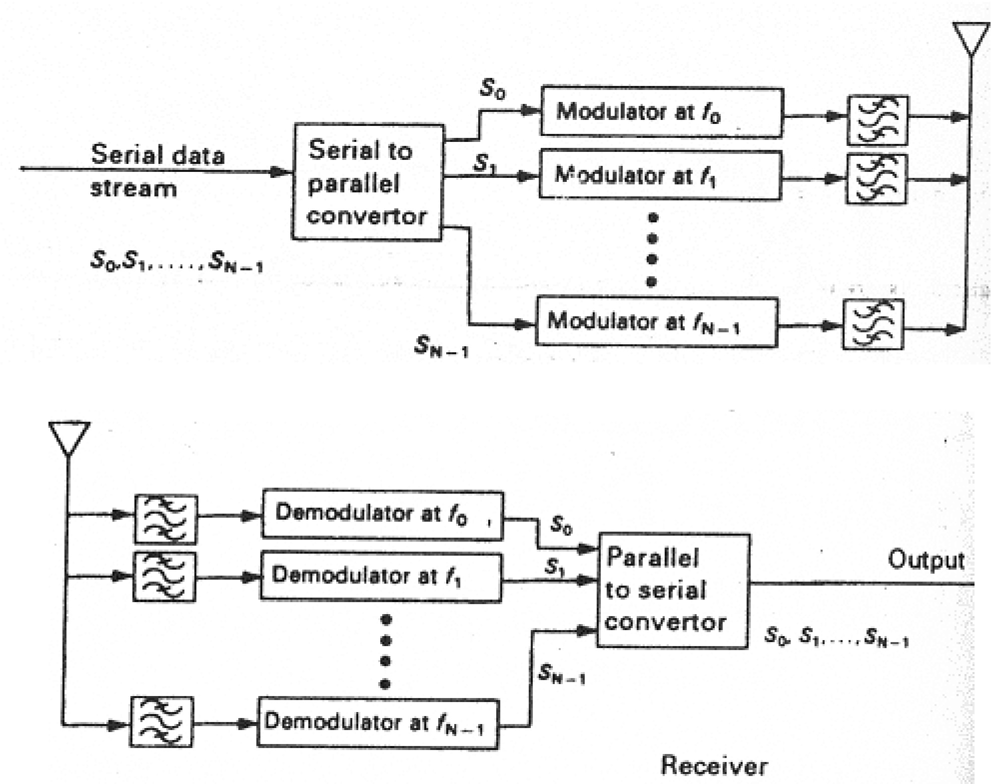


FIG 2.14 QAM-OFDM Transceiver

shows the serial transmission of symbols, S_o, S_1… S_{N-1} while the solid shaded block indicates the position of the error burst which affects k < N symbols. By contrast, during the N-symbol period of the conventional serial system, each OFDM modulator carries only one symbol, and the error burst causes severe signal degradation for the duration of k serial symbols. This degradation is shown cross-hatched. However, if the error burst is only a small fraction of the symbol period then each of the OFDM symbols may only be slightly affected by the fade and they can still be correctly demodulated. Thus while the serial system exhibits an error burst, no errors or few errors may occur using the OFDM approach.

A further advantage of OFDM is that because the symbol period has been increased, the channel delay spread is a significantly shorter fraction of a symbol period than in the serial system, potentially rendering the system less sensitive to ISI than the conventional serial system.

A disadvantage of the OFDM approach is the increased complexity over the conventional system caused by employing N modulators and filters at the transmitter and N demodulators and filters at the receiver.

In the system shown in above Figure7.5, the input serial data stream is rearranged in a sequence {dn} of N QAM symbols at baseband. Each serial QAM symbol is spaced by $\Delta t=1/Ts$ where fs is the serial signaling or symbol rate. At the nth symbol instant, the QAM symbol d(n) = a(n) +j b(n) is represented by an in-phase component a(n) and a quadrature component b(n). A block of N QAM symbols are S/P and the resulting in-phase symbols a(0), a(l). •••• a(N-1) and quadrature, symbols b(0), b(1), ..., b(N-1) are applied to N pairs of balance modulators. The quadrature components a(n) and b(n), n = 0,1, ..., N-1, modulate the quadrature carriers $\cos\omega_n t$ and $\sin\omega_n t$, respectively. Notice that the signaling interval of the sub-bands in the parallel system is N times longer than that of serial system giving $T= N\Delta t$, which corresponds to an N times lower signaling rate The sub-band carrier frequencies $\omega_n=2\pi n f_o$ are spaced apart by $\omega_n.=2\pi/T$.

The modulated carriers $a(n)\cos(\omega_n t)$ and $b(n)\sin(\omega_n t)$ when added together constitute a QAM signal, and as n = 0,1, ..., N- 1 we have N QAM symbols at RF where the nth QAM signal is given by:

$$\mathbf{X_n(t) = a(n)\cos(\omega_n t) + b(n)\sin(\omega_n t)}$$

$$\mathbf{X_n(t) = V(n)\cos(\omega_n t+\Theta_n)}$$

Where

$$V(n)=\sqrt{(a^2(n)+b^2(n))}$$

and

$$\Theta_n=\arctan(b(n)/a(n)) \quad n=0,l,\ \ldots,\ N\text{-}l.$$

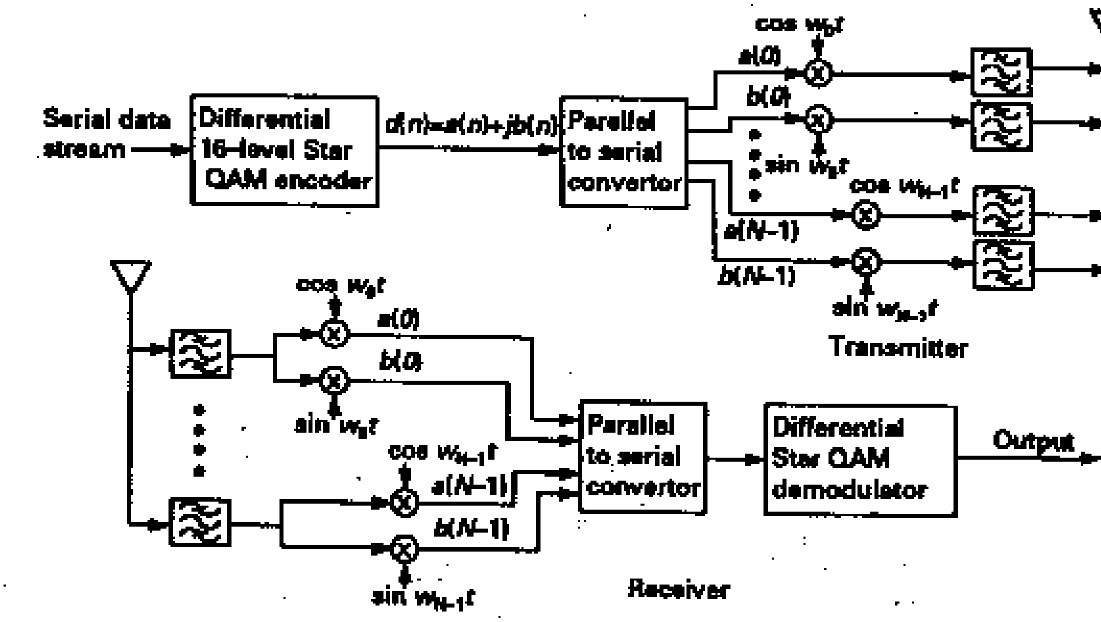


FIG 2.15 Detailed OFDM System

By adding the outputs of each sub-channel signal $X_n(t)$ $X_n(t)$ whose carriers are offset by $\omega_0=2\pi/T$ we obtain this set of N FDM / QAM signals is transmitted over the mobile radio channel. The receiver this OFDM signal is demultiplexed using a bank of N filters to regenerate the N QAM signals. The QAM baseband signals a(n) and b(n) are recovered a turned into serial form {dn}. Recovery of the data ensues using the QAM baseband demodulator and differential decoding.

2.6.2 Spectral spillage or ICI:

In theory, such a system is capable of achieving the maximum transmission of log2 Q bits/s/Hz, where Q is the number of QAM levels. In practice, there is some spectral spillage due to adjacent frequency sub-bands which reduces the efficiency. Spectral spillage due to the sub-bands at the top and bottom of the over frequency band requires a certain amount of guard space between adjacent use.

$$D(t) = \sum_{n=0}^{N-1} X_n(t).$$

Furthermore, spectral spillage between OFDM sub-bands due to the imperfections each of the sub-band niters requires that the sub-bands be spaced further apart than the theoretically required minimum amount, decreasing spectral efficiency. In order to obtain the highest efficiency, the block size should be kept high and the sub-band made to meet stringent specifications.

One of the most attractive features of this scheme is that the bandwidth of the sub-channels is very narrow when compared to the communications channel's coherent bandwidth. Therefore, flat-fading narrowband propagation conditions apply The sub-channel modems can use almost any modulation scheme, and QAM is attractive choice in some situations.

2.7 OFDM Transmitter and Reciever:

The complete system model is as shown in the figure

2.7.1 OFDM transmitter:

2.7.1.1 Modulation by Discrete Fourier Transforms:

A fundamental problem associated with the OFDM scheme described is that in on to achieve high resilience against dispersion in the channels we consider, the block size N, has to be high, requiring a large number of sub-channel modems. Fortunately it can be shown mathematically that taking the discrete Fourier transform (DFT) of the original block of N QAM symbols and then transmitting the D coefficients serially is exactly equivalent to the operations required by the OFDM transmitter of Figure 2.7. Substantial hardware simplifications can be achieved OFDM transmissions if the bank of sub-channel

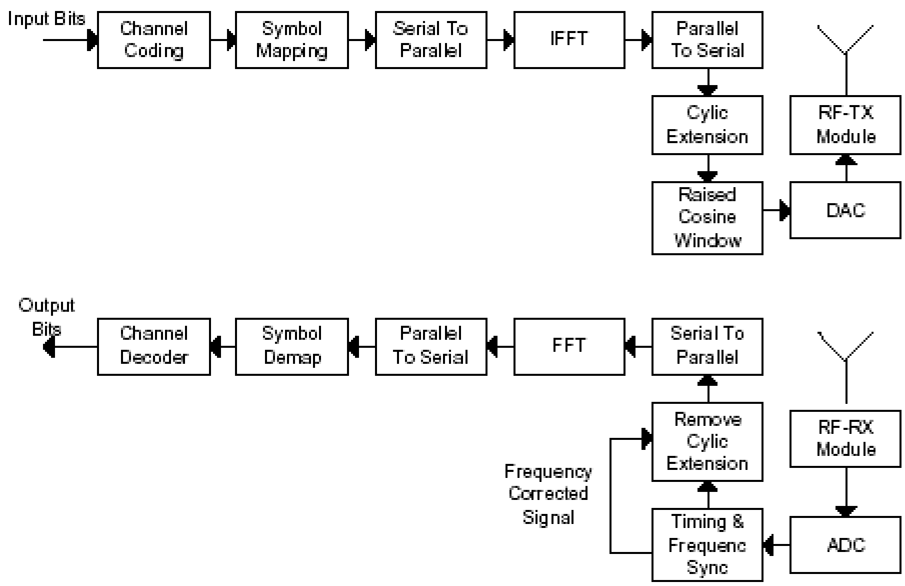


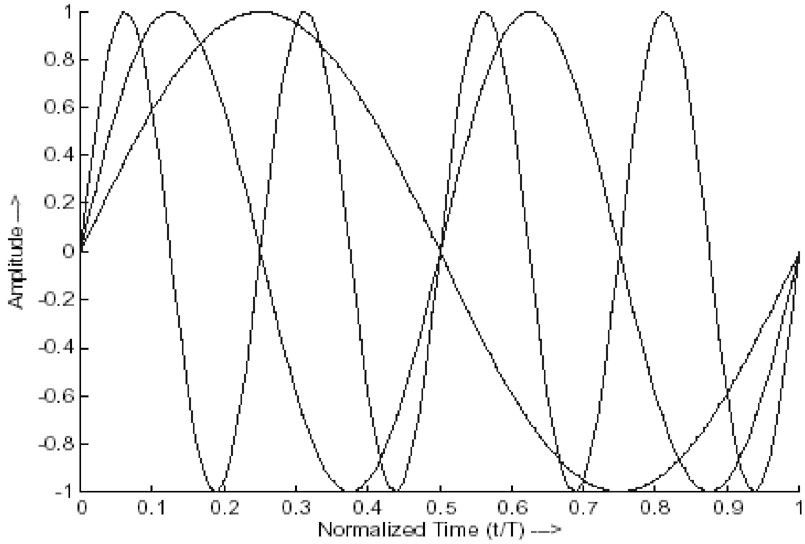
FIG 2.16 Complete System Block Diagram

modulators/demodulators is implemented using the computationally efficient pair of inverse fast Fourier transform a fast Fourier transform (IFFT/FFT).

Observe that the stream of complex baseband symbols X (k) to be transmitted c be described both in terms of in-phase I(k) and quadrature phase Q(k) component as well as by magnitude and phase- In case of a square-shaped constellation which is used above , X(k) = I(k) + j Q(k) might be a more convenient formalism X(k) = $|X(k)|^{ej\Psi k}$ to be more attractive.

Consider a data sequence d_0, d_2… d_{N-1}, where each d_n is a complex symbol. (The data sequence could be the output of a complex digital modulator, such as QAM, PSK etc).Suppose we perform an IDFT on the sequence $2d_n$ (the factor 2 is used purely for scaling purposes), we get a result of N complex numbers S_m (m = 0, 1…, N-1) as:

$S_m =$



Where, T_s represents the symbol interval of the original symbols. Passing the real part of the symbol sequence represented by equation thorough a low-pass filter with each symbol separated by a duration of T_s seconds, yields the signal,

$$y(t) = 2\,\mathrm{Re}\left\{ \sum_{n=0}^{N-1} d_n \exp(j2\pi \frac{n}{T} t) \right\}, \text{for } 0 \le t \le T$$

$$f_n = \frac{n}{NT_s} \text{ and } t = mT_s$$

FIG 2.17 Spectra Of Individual Subcarrier

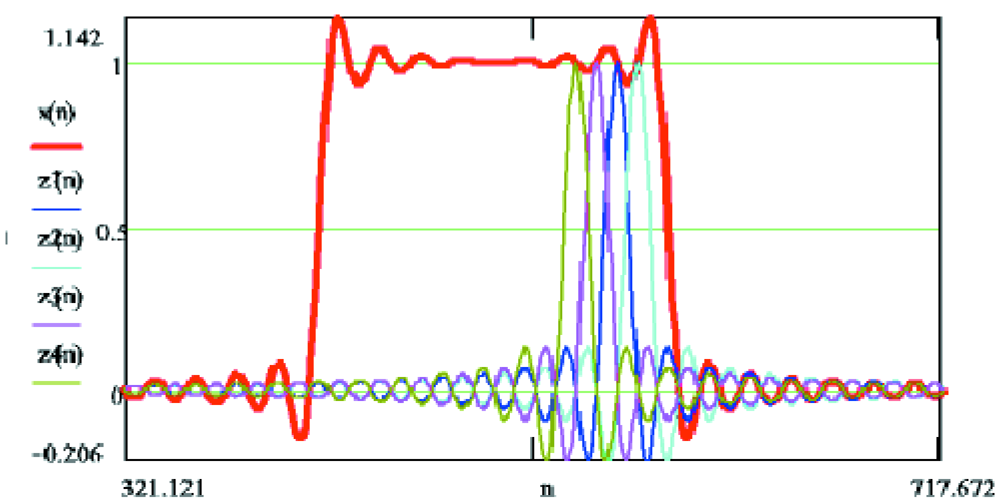


FIG 2.18 four Carriers Within OFDM Symbol

Where, T is defined as NT_s. The signal y(t) represents the baseband version of the OFDM signal.

2.7.1.2 Making a real signal:

As mentioned earlier, our design requires special attention as it relates to the data being modulated through he IFFT. This is because the output of the IFFT must be real since we are not going to modulate the signal with a carrier, since our channel is basically base-band. Furthermore, once the symbols are generated, a sequence of zeros is placed ahead of them. This gives us the advantage to be selective during transmission and completely eliminate the undesired low frequencies which microphone and speakers are filtering away. Naturally, this ability to cut off the low frequencies is fairly intuitive and comes from the fact that the number of zeros will multiply the same number of carries inside the IFFT transform. As a consequence, the signal power at those specific carriers is of zero value. A more interesting result however, is the issue of making the IFFT real. In order to eliminate the imaginary part, our design creates symmetry by concatenating a "special" sequence of data after the existing (zeros + symbols) information sequence.

This "special" sequence is obtained by flipping the (zeros + symbols) data sequence and conjugating it. The following example, clearly demonstrates the mentioned approach. Let *Symb* be a set of symbols and let *Zero* be the zeros at the undesired low frequencies:

Symb=[2 1 3 4 2 4 3 1]

Zero= [00]

Then, the zeros are placed ahead of the symbols leading the following result:

ZeroSymb=[2 1 3 4 2 4 3 1 0 0]

Next, the *ZeroSymb* sequence is flipped in the following manner:

mb FlipZeroSy=[0 0 1 3 4 2 4 3 1 2]

If the symbols are complex, the *mb FlipZeroSy* sequence is also conjugated. Finally, the *mb FlipZeroSy* sequence is attached immediately after the *ZeroSymb* sequence as follows:

SymmSeq=[0 0 1 3 4 2 4 3 1 2 2 1 3 4 2 4 3 1 0 0]

The last thing to successfully achieve the goal of making the IFFT real is to add one more zero at the beginning of *SymmSeq* which yields the following final sequence that enters the IFFT:

SeqIIFT =[0 0 1 3 4 2 4 3 1 2 2 1 3 4 2 4 3 1 0 0 0]

This zero represents the 0th frequency and therefore should not be considered for purposes of creating symmetry.

2.7.1.3 OFDM SPECTRUM SHAPE:

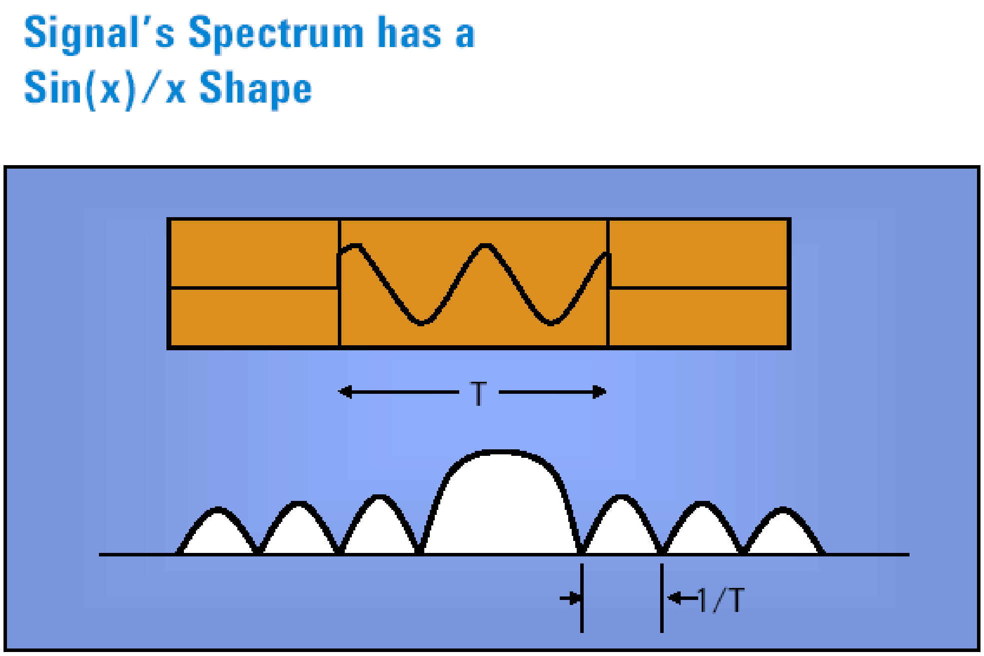


Fig 2.19 OFDM SPECTRUM SHAPE

This drawing shows the real reason for the OFDM spectral characteristics. For a single carrier, we can model the transmitted pulse as a sinusoid multiplied by a RECT function. In the frequency domain, this is simple a sin(x)/x shape convolved with an impulse at the carrier frequency.

The sin(x)/x spectrum has nulls at adjacent carrier frequencies provided the sinusoid is on frequency and has zero bandwidth, and the RECT function is the proper width.

The RECT function can have the wrong width if the ADC/DAC sample rates are incorrect in either the transmitter or receiver. The zero-width assumption can be violated by phase noise, again in either the transmitter or the receiver.

The OFDM signal is comprised of 52 carriers. The carriers at the extreme frequencies are the one that contribute the most to the sidelobe structure shown in the spectral mask plot.

2.7.1.4 Guard Period in OFDM:

The system model with guard time is as shown in the figure. One of the most important properties of OFDM transmissions is the robustness against multipath delay spread. This is achieved by having a long symbol period, which minimizes the inter-symbol interference.

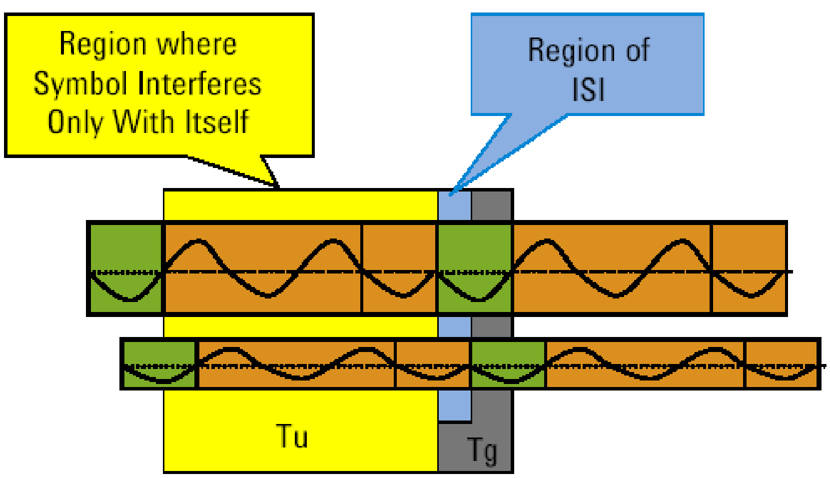
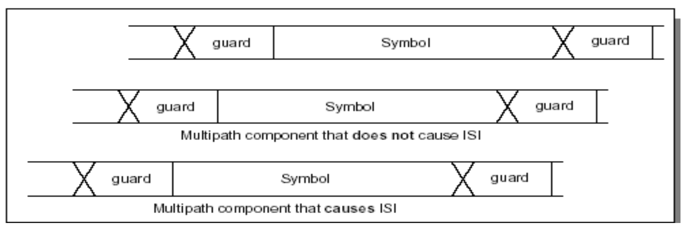


FIG 2.20 The System Model With Guard Time

This drawing illustrates how the addition of a guard interval helps with ISI. Shown are two copies of the same signal. Each copy took a different path so they will arrive at the receiver at slightly different times where they will be combined in the receiver's antenna into a single signal. In the time interval denoted by the yellow box marked Tu, the signal will only interfere with itself. This amounts to a scaling and rotation of the symbol, nothing more.

In the guard interval region(Tg), it’s easy to see that the resulting signal will have contributions from both symbols -- ISI. The guard interval is ignored in the receiver, so the ISI does not degrade receiver performance.

The level of robustness, can infact be increased even more by the addition of a guard period between transmitted symbols. The guard period allows time for multipath
signals from the pervious symbol . to die away before the information from the current



symbol is gathered .

FIG 2.21 Guard Time and Cyclic Extension - Effect of Multipath

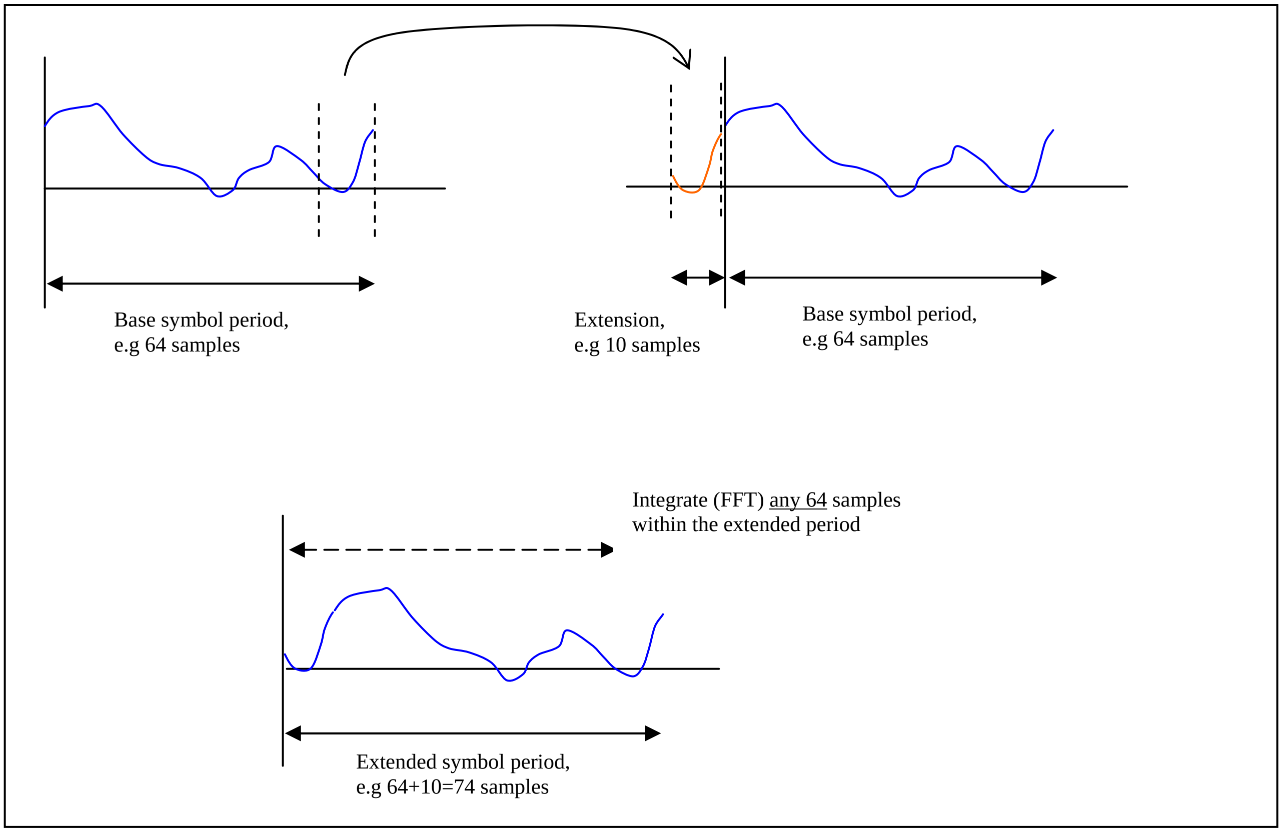


Fig 2.22 Guard Period via Cyclic Extension

The most effective guard period to use is a cyclic extension of the symbol. If a mirror in time, of the end of the symbol waveform is put at the start of the symbol as the guard period, this effectively extends the length of the symbol, while maintaining the orthogonality of the waveform. Using this cyclic extended symbol the samples required for performing the FFT (to decode the symbol), can be taken anywhere over the length of the symbol. This provides multipath immunity as well as symbol time synchronization tolerance.

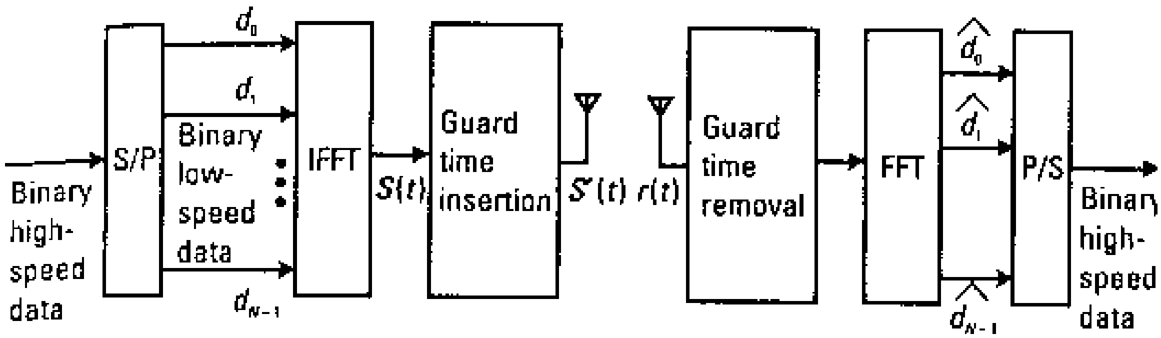


Fig 2.23 basic model with guard time

As long as the multipath delay echoes stay within the guard period duration, there is strictly no limitation regarding the signal level of the echoes:

they may even exceed the signal level of the shorter path. The signal energy from all paths just add at the input to the receiver, and since the FFT is energy conservative, the whole available power feeds the decoder. If the delay spread is longer then the guard interval then they begins to cause intersymbol interference. However, provided the echoes are sufficiently small they do not cause significant problems. This is true most of the time as multipath echoes delayed longer than the guard period will have been reflected of very distant objects. Other variations of guard periods are possible. One possible variation is to have half the guard period a cyclic extension of the symbol, as above, and the other half a zero amplitude signal. This will result in a signal as shown in Figure 15. Using this method the symbols can be easily identified. This possibly allows for symbol timing to be recovered from the signal, simply by applying envelop detection. The disadvantage of using this guard period method is that the zero period does not give any multipath tolerance, thus the effective active guard period is halved in length. It is interesting to note that this guard period method has not been mentioned in any of the research papers read, and it is still not clear whether symbol timing still needs to be recovered.

2.7.1.4.1 Guard time considerations:

Guard time in an OFDM system usually results in an SNR loss in an OFDM system, since it carries no information. The choice of the guard time is straightforward once the multi-path delay spread is known. As a rule of thumb, the guard time must be at least 2-4 times the RMS delay spread of

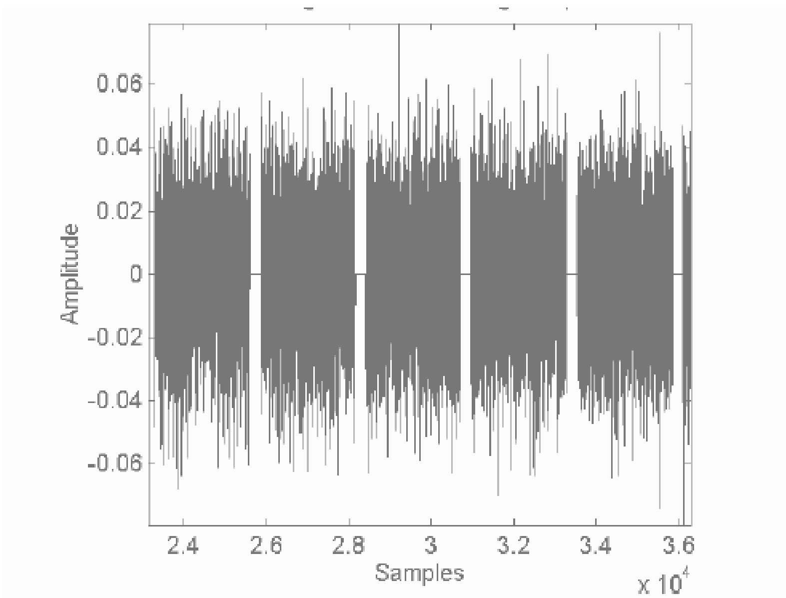


FIG 2.24 Half Zero Guard Period

the multi-path channel. Further, higher-order modulation schemes (like 32 or 64 QAM) are more sensitive to ISI and ICI (INTERCARRIER INTERFERENCE) than simple schemes like QPSK. This factor must also be taken into account while deciding on the guard-time.

2.7.1.5 Raised Cosine Windowing:

The sharp-phase transitions caused by phase modulation results in very large side-lobes in the PSD and the spectrum falls off rather slowly (according to a *sinc* function). If the number of sub-carries were increased, the spectrum roll-off will be sharper in the beginning, but gets worse at frequencies a little further away from the 3-dB cut-off frequency. To overcome this problem of slow spectrum roll-off, a windowing may be used to reduce the side-lobe level. The most commonly used window is the **Raised Cosine Window**:

$$w(t) = \begin{cases} 0.5 + 0.5\cos(\pi + \pi t/(\beta T_r)), & 0 \le t \le \beta T_r \\ 1.0, & \beta T_s \le t \le T_r \\ 0.5 + 0.5\cos((t\text{-}T_r)\pi/(\beta T_r)), & T_s \le \mathrm{t} \le (1+\beta)T_r \end{cases}$$

Here Tr is the symbol interval which is chosen to be shorter than the actual OFDM symbol duration, since the symbols are allowed to partially overlap in the roll-off region of the raised cosine window. Incorporating the windowing effect, the OFDM symbol can now be represented as:

$$y(t) = 2\,\mathrm{Re}\left\{ w(t)\sum_{n=0}^{N-1} d_n \exp(j2\pi \frac{n}{T} t) \right\}, \text{ for } 0 \le t \le T$$

It must be noted that filtering can also be used as a substitute for windowing, for tailoring the spectrum roll-off. But windowing is preferred to filtering because, it can be carefully controlled. With filtering, one must be careful to avoid rippling effects in the roll-off region of the OFDM symbol. Rippling causes distortions in the OFDM symbol, which directly leads to less-delay spread tolerance.

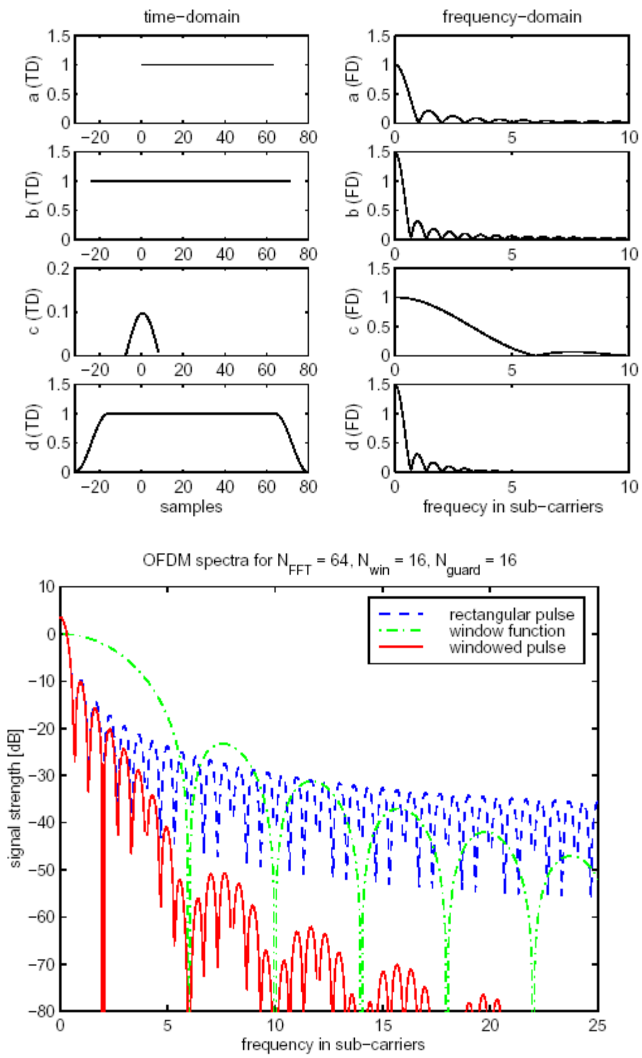


FIG 2.25 **(a): Shape and spectrum of the OFDM receive filter (realized by FFT); (b): rectangular pulse of duration *T* and its spectrum; (c): sine-half-wave used for pulseshaping and its spectrum; (d): transmitter pulse prototype *w*(*t*) and its spectrum. (e): Spectra of (b)–(d) in logarithmic scale.**

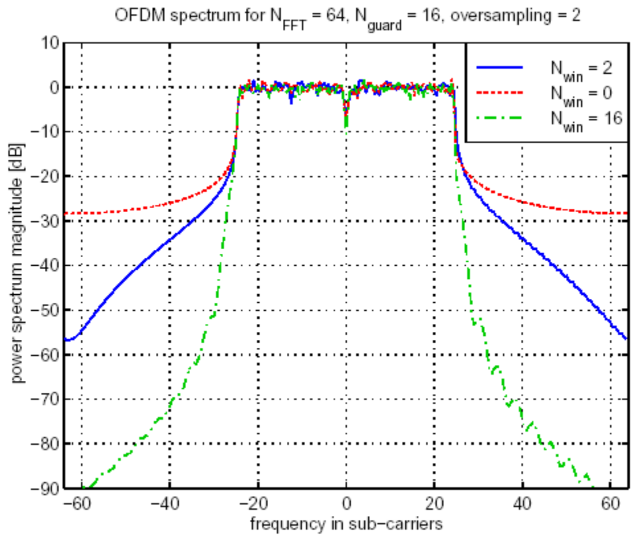


FIG 2.26

Spectrum of an OFDM signal with 64 sub-carriers and different window lengths. Two-fold oversampling has been applied in the time-domain; 48 sub-carriers are used for data.

2.7.1.6 Numbers of Sub carriers:

Once the symbol duration is determined, the number of sub-carriers required can be calculated by first calculating the sub-carrier spacing which is just the inverse of the symbol time (less the guard period). The number of sub-carriers is the available bandwidth divided by the sub-carrier spacing.

2.7.2 Channel:

Channel model is then applied to the transmitted signal. The model allows for the signal to noise ratio, multipath, and peak power clipping to be controlled. The signal to noise ratio is set by adding a known amount of white noise to the transmitted signal. Multipath delay spread then added by simulating the delay spread using an FIR filter. The length of the FIR filter represents the maximum delay spread, while the coefficient amplitude represents the reflected signal magnitude.

2.7.3 OFDM Receiver:

The receiver basically does the reverse operation to the transmitter. The guard period is removed. The FFT of each symbol is then taken to find the original transmitted spectrum. The phase angle of each transmission carrier is then evaluated and converted back to the data word by demodulating the received phase. The data words are then combined back to the same word size as the original data. The receiver has many issues Such as equalization, symbol synchronization, and removal of guard interval etc which are discussed as follows

2.7.3.1 The Peak power problem in OFDM:

One of the most serious problems with OFDM transmission is that, it exhibits a high peak-to-average ratio. In other words, there is a problem of extreme amplitude excursions of the transmitted signal. The OFDM signal is basically a sum of N complex random variables, each of which can be considered as a complex modulated signal at different frequencies. In some cases, all the signal components can add up in phase and produce a large output and in some cases, they may cancel each other producing zero output. Thus the peak-to-average ratio (PAR) of the OFDM system is very large. The problem of Peak-To-Average Ratio is more serious in the transmitter. In order to avoid clipping of the transmitted waveform, the power-amplifier at the transmitter front end must have a wide linear range to include the peaks in the transmitted waveform.

Building power amplifiers with such wide linear ranges is a costly affair. Further, this also results in high power consumption. The DAC's and the ADC's must also have a wide range to avoid clipping. There has been a lot of research put into the study of overcoming the PAR problem in OFDM. The following sections discuss some of the most common and important of those techniques as well as other issues.

2.7.3.2 Synchronization in OFDM Systems:

Another important issue in OFDM transmission is synchronization. There are basically three issues that must be addressed in synchronization. The receiver has to estimate the symbol boundaries and the optimal timing instants that minimize the effects of inter-carrier interference (ICI) and inter-symbol interference (ISI). In an OFDM system, the sub-carriers are exactly orthogonal only if the transmitter and the receiver use exactly the same frequencies. Thus receiver has to estimate and correct for the carrier frequency offset of the received signal. Further, the phase information must be recovered if coherent demodulation is employed. Another associated problem with OFDM systems is the effect of phase noise. Phase noise is present in all practical oscillators and it manifests itself in the form of random phase modulation of the carrier. Both phase-noise and frequency offset cause significant amount of ICI in an OFDM receiver. The effect these are worse in OFDM than single carrier systems. The use of efficient frequency and phase estimation schemes can help reduce these effects.Some of the common methods used to achieve synchronization in OFDM systems are:

2.7.3.2.1 Synchronization using Cyclic Extension:

Since a Cyclic extension is added to every OFDM symbol, the first T_g seconds of the OFDM symbol is identical to the last part. This property can be exploited for both timing and frequency synchronization. It correlates T_g seconds of the OFDM symbol with a part that is T seconds delayed (T – being the symbol time, less the guard period T_g). The output of the correlator can be written as:

$$y(t) = \int_0^{T_g} r(t-\tau)r(t-\tau-T)d\tau$$

The symbol timing is estimated from the correlation peaks at the output of the correlator. The characteristics of the correlation peaks (in terms of the correlation side-lobe levels and the standard deviation of the correlation magnitude) are better if the correlation is performed over a large number of independent samples. Since the number of independent samples is proportional to the number of sub-carriers, this cyclic extension correlation method is efficient only if a large number of sub-carriers are present (more than 100). In the case of less number of sub-carriers, the side-lobe to peak ratio of the correlator output will be high and sometimes this might lead to wrong timing.

Once the timing is established using the correlation output, the frequency offset can be directly estimated. The phase of the correlator output is equal to the phase drift between samples that are T seconds apart. Hence the frequency offset can be estimated as the correlation phase divided by T_g/2.

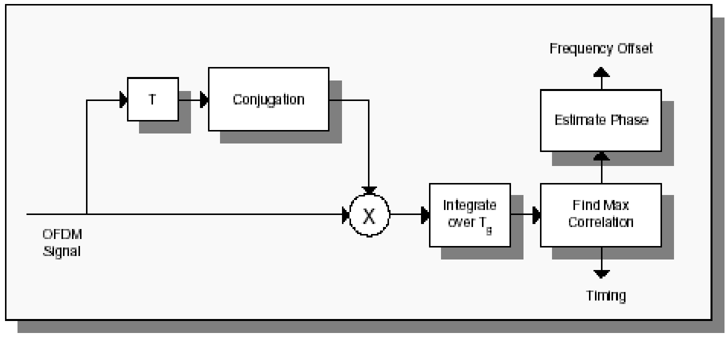


FIG 2.27 synchroniztion using cyclic extension

2.7.3.2.2 Synchronization using Training Sequences:

In cases like packet data transmission in which a training sequence is available, a much more efficient method of timing recovery is to correlate the received signal with the known training sequence and to find the peaks in the correlator output. Here T is the sampling interval and ci are the matched filter coefficients, which are in turn, the complex conjugates of the known training sequence. From the correlation peaks in the output signal, both the symbol timing and the frequency offset can be estimated.

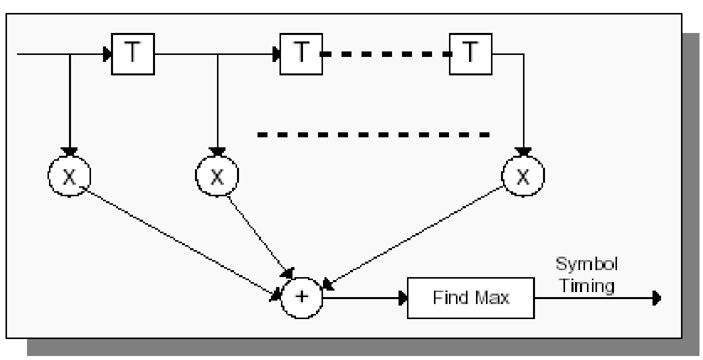


FIG 2.28 Synchronization using Training Sequences

To maximize the detection of the start of the synchronizing sequence, we choose to use a CHIRP sequence. It generates a sine wave with linearly increasing frequency:

% Matlab generation of the synchronizing sequence we use:

```
sync_seq_len= 2000;
a = 0.2;
mu = 0.1;
t = [0:sync_seq_len-1]';sync_seq=0.45*sin(2*pi*mu/sync_seq_len*t.^2);
```

The autocorrelation of this sequence is particularly interesting.

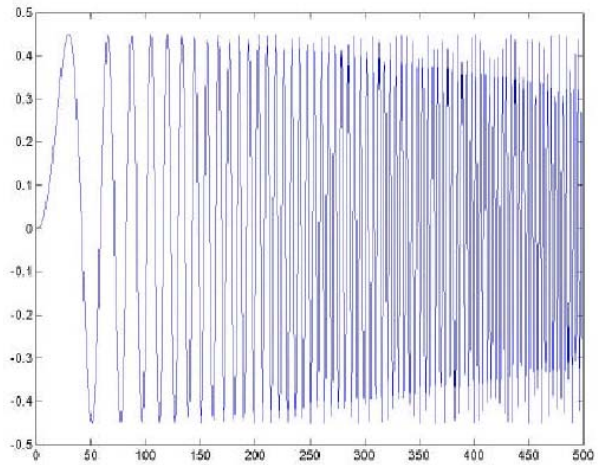


Figure 2.29: Example of synchronizing sequence

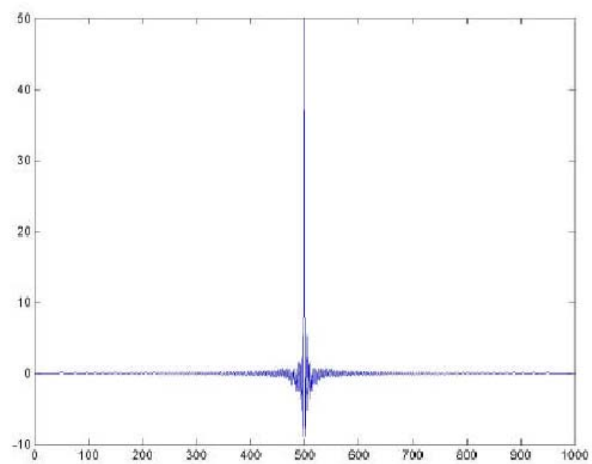


Figure 2.30a: autocorrelation of the chirp synchronizing sequence

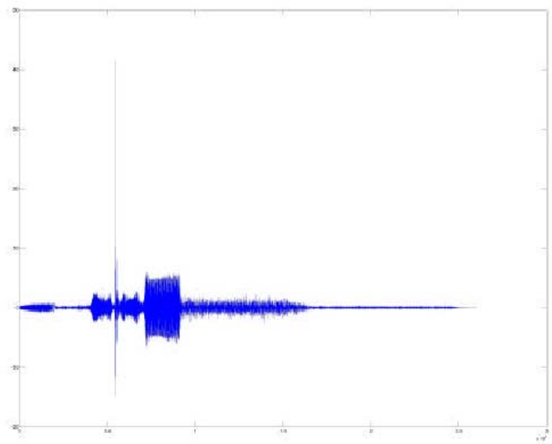


Figure 2.30b: Example of correlation for the synchronization in the experiment.

We can see that the peak is very clear: the starting point of the sequence cannot be missed much. It might not be perfect due to the channel distortion: if the channel has two close peaks, then the correlation may give us also two peaks, thus mistaking the start of the sequence.

2.7.3.2 Optimal Timing in the presence of Multi-Path:

The effect of multi-path is the introduction of ICI and ISI in the OFDM symbol. These effects are significant only if the delay spread of the channel exceeds the guard interval. ICI is caused mainly because the FFT interval is no longer flat (because the roll-off regions due the multi-path components interfere with the flat region of the FFT interval). ISI is caused mainly because of the overlap between the previous OFDM symbol and the current OFDM symbol in the FFT interval. The solution to this timing problem is to find the delay window with a width equal to the guard time-that contains the maximum signal power. The optimal FFT starting time is then equal to the starting delay of the found delay window, plus the delay that occurs between a matched filter peak output from a single OFDM pulse and the delay of the last sample from the flat part of the OFDM signal envelope, minus the length of the FFT interval.

2.8 OFDM Bandwidth Efficiency:

In the OFDM system each symbol to be transmitted modulates an assigned carrier of a set wide-sense orthogonal basis functions and these modulated sub-channel signals are superimposed for transmission via the communications channel- The received signal can be modulated for example by the correlation receiver described in the previous section or by FFT.Both their time domain waveforms and stylized spectra are shown in Figure. In a simplistic approach here we assume that the signal spectra can be band limited i.e. bandwidth of its main spectral lobe, as suggested by the figure. Using an essentially serial system with one carrier, as seen in Fig minimum bandwidth required is fB = 1/T and the bandwidth efficiency is 1Bd/Hz because the spectrum of this pulse is represented by the sinc function whose zero is at fB = 1/T. The three-carrier system of Figure expands the of the basis functions to 3T, thereby reducing the bandwidth requirement B=2/3 giving T= 1.5Bd/Hz. This is because the rectangular window\ sin and cos spectra are represented by the convolution of a tonal spectral line and \ a frequency domain sine function describing the spectrum of the rectangular time domain window. The five-carrier scheme using basis functions of 5T length further reduces the bandwidth to B == 3/5 f_B increases the spectral efficiency to 1.67 B_d/Hz.

Similarly, the approximate bandwidth of a (2M + l) carrier system using an impulse as well as M sine and M cosine carriers of length (2M + 1)T becomes hen M —> ∞, we have lim M—>∞, == 2Bd/Hz, which for a typical value of M 64 gives 1.98Bd/Hz.

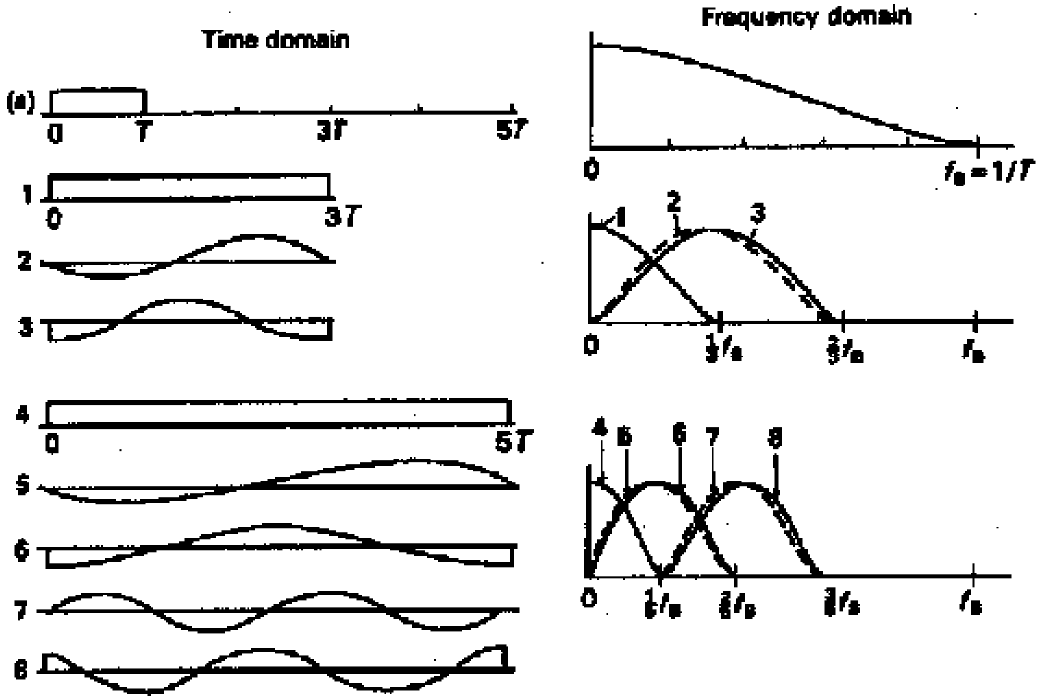


FIG 2.31 Spectral Efficiency

3.1 Advantages of OFDM:

OFDM possesses some inherent advantages for Wireless Communications. This section glances on few of the most important reasons on why OFDM is becoming more popular in the Wireless Industry today.

3.1.1 Multi-path Delay Spread Tolerance:

As discussed earlier, the increase in the symbol time of the OFDM symbol by N times (N being the number of sub-carriers), leads to a corresponding increase in the effectiveness of OFDM against the ISI caused due to multi-path delay spread. Further, using the cyclic extension process and proper design, one can completely eliminate ISI from the system.

3.1.2 Effectiveness against Channel distortion:

In addition to delay variations in the channel, the lack of amplitude flatness in the frequency response of the channel also causes ISI in digital communication systems.

A typical example would be the twister-pair used in telephone lines. These transmission lines are used to handle voice calls and have a poor frequency response when it comes to high frequency transmission. In systems that use single-carrier transmission, an equalizer might be required to mitigate the effect of channel distortion. The complexity of the equalizer depends upon the severity of the channel distortion and there are usually issues such as equalizer non-linearities and error propagation etc that cause additional trouble.

In OFDM systems on the other hand, since the bandwidth of each sub-carrier is very small, the amplitude response over this narrow bandwidth will be basically flat (of course, one can safely assume that the phase response will be linear over this narrow bandwidth). Even in the case of extreme amplitude distortion, an equalizer of very simple structure will be enough to correct the distortion in each sub-carrier.

3.1.3 Throughput Maximization (Transmission at Capacity):

The use of sub-carrier modulation improves the flexibility of OFDM to channel fading and distortion makes it possible for the system to transmit at maximum possible capacity using a technique called channel loading. Suppose the transmission channel has a fading notch in a certain frequency range corresponding to a certain sub-carrier. If we can detect the presence of this notch by using channel estimation schemes and assuming that the notch doesn't vary fast enough compared to the symbol duration of the OFDM symbol, it can be possible to change (scale down/up) the modulation and coding schemes for this particular sub-carrier (i.e., increase their robustness against noise), so that capacity as a whole is maximized over all the sub-carriers. However, this requires the data from channel-estimation algorithms. In the case of single-carrier systems, nothing can be done against such fading notches. They must somehow survive the distortion using error correction coding or equalizers.

3.1.4 Robustness against Impulse Noise:

Impulse noise is usually a burst of interference caused usually caused in channels such as the return path HFC (Hybrid-Fiber-Coaxial), twisted-pair and wireless channels affected by atmospheric phenomena such as lightning etc. It is common for the length of the interference waveform to exceed the symbol duration of a typical digital communication system. For example, in a 10 MBPS system, the symbol duration is 0.1 s m, and a impulse noise waveform, lasting for a couple of micro-seconds can cause a burst of errors that cannot be corrected using normal error-correction coding .OFDM systems are inherently robust against impulse noise, since the symbol duration of an OFDM signal is much larger than that of the corresponding single-carrier system and thus, it is less likely that impulse noise might cause (even single) symbol errors. Thus, complicated error-control coding and interleaving schemes for handling burst-type errors are not really required for OFDM Systems simplifying the transceiver design.

3.1.5 Frequency Diversity:

OFDM is the best place to employ Frequency Diversity. In fact, in a combination of OFDM and CDMA called the MC-CDMA transmission technique, frequency diversity is inherently present in the system. (i.e., it is available for free).Even though, OFDM provides a lot advantages for Wireless Transmission, it has a few serious disadvantages that must be overcome for this technology to become a success.

3.2 APPLICATIONS OF OFDM:

- Digital Television
- European and Australian standard
- Digital Video Broadcast (DVB)
- Digital Audio Broadcast (DAB)
- Wireless Local Area Networks (WLANs)
- Wireless LAN (802.11x & HiperLAN)
- ADSL (asymmetric digital subscriber loop)
- High speed data transmitted along existing telephone lines

STANDARD	MEANING	CARRIER FREQ	RATE (MBPS)	APPLICATIONS
DAB	DIGITAL AUDIO BROADCASTING	FM	.384	AUDIO BROADCASTING
DVB-T	DIGITAL VIDEO BROADCASTING	UHF	32	DIGITAL TV BROADCASTING
IEEE 802.11a	WLAN	5.2 GHz	54	WIRELESS ACCESS
IEEE 802.16.3	FIXED WIRELESS	2.1 GHz	12	INTERNET VOICE ACCESS

- Future mobile telephony

Table 4 Applications Range Of Frequencies

OFDM is ideally suited to high bandwidth broadcast applications, such as DVB, where immunity to large delay spread Multipath is essential. A secondary benefit of OFDM for broadcast is its suitability for Single Frequency Networks (SFN) where synchronized transmitters can use the same frequency. OFDM is also well suited to wireless LANs where period and a guard interval between the dynamic nature of both the traffic and the propagation path require rapid adaptation to varying Multipath.

OFDM is a physical layer technology that enables high data rate wireless applications. The IEEE 802.11a and ETSI BRAN HiperLAN 2 Wireless LAN standards are currently based on OFDM.

OFDM is also seen in Broadband Wireless Access markets in places where DSL or Cable are not cost effective or available due to infrastructure. OFDM is being strongly considered for 4G mobile networks and is under review by a number of the major telecommunications carriers globally.

OFDM is highly spectrally efficient and the best at addressing multi-path issues, which are the two largest inhibitors in the wireless arena.

3.2.1 OFDM in Broadcasting :

- Enables Single Frequency Network (SFN)
- Multiple transmit antennas geographically separated
- Enables same radio/TV channel frequency throughout a country
- Creates artificially large delay spread – OFDM has no problems

3.2.2 OFDM for high-speed Internet Access:

- High-speed data transmission
- Large bandwidths -> high rate, many computations
- Small sampling periods -> delay spread becomes a serious impairment
- Requires much lower BER than voice systems
- Takes advantage of Multipath through simple equalization
- Synchronization requirements are much more strict
- Requires more complex algorithms for time / frequency synchronization
- Peak-to-average ratio
- PAR is approximately 10 log N (dB)
- Large signal peaks require higher power amplifiers
- Amplifier cost grows nonlinearly with required power

OFDM SOLVES THE MULTIPATH PROBLEM?

- Data is transmitted in parallel

- Longer symbol period e.g. for N parallel streams, symbol period is N times as long.
- Cyclic prefix
- Trick to avoid residual ISI
- ### 3.2.4 Digital video broadcasting (DVB):
- OFDM is used in the Australian digital television system
- 2048 point IFFT
- 1705 sub carriers used
- Flexible standard
- Variable error coding
- Variable cyclic prefix
- Variable constellation
- 4-QAM, 16-QAM, 64-QAM

3.2.5 OFDM in ADSL:

- OFDM used in ADSL is usually called 'Discrete Multitone' (DMT)
- Two way transmission
- Transmission can be tailored to the particular channel
- Baseband system
- Only real (not complex signal can be transmitted)

3.2.6 HIPERLAN-2 - Wireless LAN:

- 64 point FFT
- 52 Sub carriers used
- Different modes
- Signal constellation
- Error coding
- Cyclic prefix
- Two way channel
- Feedback be used to determine transmission mode

3.2.7 OFDM for mobile Data & Spectrum Efficiency:

While projections of capital and spectrum costs for next-generation, circuit-switched mobile networks continue to swell, mobile operators are cautiously moving forward with plans to market wireless data to enterprise and consumer mass markets. If they are to succeed with mass-market adoption of wireless data services, operators must find technology solutions to overcome the legacy barriers of the circuit-switched architecture and make the move to an end-to-end packet-switched network system that allows for scalable economics in the delivery of profitable wireless data service.

Data traffic is characterized as burst, meaning that there is significant variability in when the traffic arrives, the rate at which it arrives, and the number of bits in the messages. Also, data applications require extremely reliable delivery, with virtually no tolerance for bit errors. Data cannot be efficiently carried over the hierarchical networks designed for voice traffic. Mobile data systems face additional challenges as a result of the vagaries of the wireless environment.

In order for any mobile system to create a rich user experience, it must provide ubiquitous, always-on, fast and user-friendly connectivity. Orthogonal Frequency Division Multiplexing (OFDM) has several unique properties that make it especially well-suited to handle the challenging conditions experienced by mobile wireless data applications.

OFDM is a multiple access method that segments a communications channel in such a way that many users can share it. Whereas TDMA segments according to time and CDMA segments according to spreading codes, OFDM segments according to frequency. It is a technique that divides the spectrum into a number of equally spaced tones, and carries a portion of a user's information on each tone. OFDM has a special property that each tone is orthogonal with every other tone. OFDM allows the spectrum of each tone to overlap, and since they are orthogonal, they do not interfere with each other. By allowing the tones to overlap, OFDM increases the overall spectral efficiency.

As a multiple access technique, OFDM allows an individual tone or groups of tones to be assigned to different users. Multiple users share a given bandwidth — each user can be assigned a predetermined number of tones when they have information to send, or alternatively, a user can be assigned a variable number of tones based on the amount of information they have to send. The assignments are controlled by the media access control (MAC) layer, which schedules the resource assignments based on user demand.

Conventional wireless systems have been designed primarily at the physical layer. To address the unique demands posed by mobile data applications, new air interfaces must be designed and optimized across all the layers of the protocol stack, especially the MAC and link layers. A prime example of this kind of optimization is found in flash-OFDM™ technology.

Unlike existing OFDM based systems (e.g., DSL), flash-OFDM is much more than a physical layer solution. It is a system level technology that exploits the unique physical properties of OFDM, enabling significant higher layer advantages that contribute to very efficient packet data transmission, as well as low latency, in a cellular network.

Since all conventional cellular wireless systems, including 3G, were fundamentally designed around circuit switched voice, they were designed and optimized primarily at the physical layer. Flash-OFDM, on the other hand, is a packet switched system designed for data and is optimized across the physical, MAC, link and network layers. The choice of OFDM as the multiple access technology is based not just on physical layer considerations but also on MAC, link and network layer requirements.

OFDM is combined with frequency hopping (fast hopping across all tones in a pseudorandom predetermined pattern) to create a spread spectrum system, realizing the benefits of frequency diversity and interference averaging as is also found in CDMA. In a frequency hopping spread spectrum system, each user's set of tones is changed after each time period (usually corresponding to a modulation symbol). By switching frequencies after each symbol time, the losses due to frequency selective fading are minimized. Different base stations use different hopping patterns and each uses the entire available spectrum (resulting in frequency reuse of 1). In a cellular deployment this leads to all the advantages that CDMA systems have over narrow band systems like conventional TDMA, including frequency diversity and out of cell (intercell) interference averaging.

Each user within the same cell uses different resources (tones) and hence do not interfere with each other. This is similar to TDMA where each user in a cell transmits in different time slots and do not interfere with one another. In contrast, CDMA users in a cell do interfere with each other, because the spreading codes are not orthogonal in the real world, increasing the total interference in the system. Flash-OFDM therefore has the physical layer benefits of both CDMA and TDMA and is at least three times more spectrally efficient than CDMA. In other words, at the physical layer, flash-OFDM creates the fattest pipe of all cellular technologies. Even though the 3x advantage at the physical layer is a huge advantage, the most significant advantage of flash-OFDM for data is at the MAC and link layers, where additional data capacity gains are realized.

Flash-OFDM exploits the granular nature of resources (tones) in OFDM to come up with extremely efficient control layers. In OFDM, it is possible to send as little as one bit of information from the transmitter to the receiver with virtually no overhead. This is unlike CDMA or TDMA, where the granularity is much coarser and just to initiate a transmission adds significant overhead.

Flash-OFDM takes advantage of the granularity of OFDM in its control layer design enabling the MAC layer to perform efficient packet switching over the air and at the same time providing the ability to provide Quality of Service (QoS). It also supports a link layer that uses local (as opposed to end-to-end) feedback to create a very reliable link from an unreliable wireless channel, with very low delays. The network layers traffic therefore experiences small delays and no significant delay jitter. Hence, interactive applications like (packet) voice can be supported. Moreover, Internet protocols like TCP/IP (transport control protocol) run smoothly and efficiently over a flash-OFDM airlink. 3G mobile networks, although meant to carry voice and data traffic simultaneously, retain a circuit-switched, hierarchical architecture. Consequently there is tension between the design objectives and the current environment of the wired Internet and mobile voice networks. The resulting design compromises of 3G networks impair their ability to deliver high-speed, low-latency data cost effectively. An alternative approach, focusing directly on high-speed, low-cost and low-latency wireless data delivery (such as flash-OFDM) is now in consideration with many operators.

Flash-OFDM is well positioned to meet the unique demands of mobile packet data traffic. But in order to seamlessly unwire all the IP applications inherent in the wired Internet and Intranets (including interactive data applications and peer-to-peer applications), all layers of the air interface need to be jointly designed and optimized from the ground up for the IP data world. Thus, flash-OFDM does not rely solely on OFDM's physical layer advantages, but to leverage them into all of the higher layers of the system.

3.2.8 OFDM for mobile communications :

OFDM represents a different system-design approach. It can be thought of as a combination of modulation and multiple-access schemes that segments a communications channel in such a way that many users can share it. Whereas TDMA segments are according to time and CDMA segments are according to spreading codes, OFDM segments are according to frequency. It is a technique that divides the spectrum into a number of equally spaced tones and carries a portion of a user's information on each tone. A tone can be thought of as a frequency, much in the same way that each key on a piano represents a unique frequency. OFDM can be viewed as a form of frequency division multiplexing (FDM), however, OFDM has an important special property that each tone is orthogonal with every other tone. FDM typically requires there to be frequency guard bands between the frequencies so that they do not interfere with each other. OFDM allows the spectrum of each tone to overlap, and because they are orthogonal, they do not interfere with each other. By allowing the tones to overlap, the overall amount of spectrum required is reduced.

For example, if a 100-tone system were used, a single data stream with a rate of 1 megabit per second (Mbps) would be converted into 100 streams of 10 kilobits per second (kbps). By creating slower parallel data streams, the bandwidth of the modulation symbol is effectively decreased by a factor of 100, or, equivalently, the duration of the modulation symbol is increased by a factor of 100. Proper selection of system parameters, such as the number of tones and tone spacing, can greatly reduce, or even eliminate, ISI, because typical Multipath delay spread represents a much smaller proportion of the lengthened symbol time. Viewed another way, the coherence bandwidth of the channel can be much smaller, because the symbol bandwidth has been reduced. The need for complex multi-tap time-domain equalizers can be eliminated as a result.

OFDM can also be considered a multiple-access technique, because an individual tone or groups of tones can be assigned to different users. Multiple users share a given bandwidth in this manner, yielding the system called OFDMA. Each user can be assigned a predetermined number of tones when they have information to send, or alternatively, a user can be assigned a variable number of tones based on the amount of information that they have to send. The assignments are controlled by the media access control (MAC) layer, which schedules the resource assignments based on user demand.

OFDM can be combined with frequency hopping to create a spread spectrum system, realizing the benefits of frequency diversity and interference averaging previously described for CDMA. In a frequency hopping spread spectrum system, each user's set of tones is changed after each time period (usually corresponding to a modulation symbol). By switching frequencies after each symbol time, the losses due to frequency selective fading are minimized. Although frequency hopping and CDMA are different forms of spread spectrum, they achieve comparable performance in a Multipath fading environment and provide similar interference averaging benefits.

OFDM therefore provides the best of the benefits of TDMA in that users are orthogonal to one another, and CDMA, as previously mentioned, while avoiding the limitations of each, including the need for TDMA frequency planning and equalization, and multiple access interference in the case of CDMA.

3.2.9 Leading-Edge mobile OFDM Technologies:

Unlike most existing forms of wireless access, including 3G technologies, conventional wireless systems have been designed primarily at the physical layer. To address the unique demands posed by mobile users of high-speed data applications, new air interfaces must be designed and optimized across all the layers of the protocol stack, including the networking layers. A prime example of this kind of optimization is found in flash-OFDM™ technology by Flarion. As its name suggests, the system is based on OFDM, however, flash-OFDM is much more than just a physical-layer solution. It is a system-level technology that exploits the unique physical properties of OFDM, enabling significant higher-layer advantages that contribute to very efficient packet data transmission in a cellular network.

1 Packet Switched Air Interface

The telephone network, designed basically for voice, is an example of circuit-switched systems. Circuit-switched systems exist only at the physical layer that uses the channel resource to create a bit pipe. They are conceptually simple as the bit pipe is a dedicated resource, and there is no control of the pipe required once it is created (some control may be required in setting up or bringing down the pipe). Circuit-switched systems, however, are very inefficient for burst data traffic.

Packet-switched systems, on the other hand, are very efficient for data traffic but require control layers in addition to the physical layer that creates the bit pipe. The MAC layer is required for the many data users to share the bit pipe. The link layer is needed to take the error-prone pipe and create a reliable link for the network layers to pass packet data flows over. The Internet is the best example of a packet-switched network.

Because all conventional cellular wireless systems, including 3G, were fundamentally designed for circuit -witched voice, they were designed and optimized primarily at the physical layer. The choice of CDMA[1] as the physical-layer multiple-access technology was also dictated by voice requirements. Flash-OFDM, on the other hand, is a packet-switched designed for data and is optimized across the physical, MAC, link, and network layers. The choice of OFDM as the multiple-access technology is based not just on physical-layer consideration, but also on MAC–, link-, and network-layer requirements.

3.3 Physical-Layer Advantages:

Flash-OFDM uses fast hopping across all tones in a pseudorandom predetermined pattern, making it a spread spectrum technology. With fast hopping, a user that is assigned one tone does not transmit every symbol on the same tone, but uses a hopping pattern to jump to a different tone for every symbol. Different base stations use different hopping patterns, and each uses the entire available spectrum (frequency reuse of 1). In a cellular deployment, this leads to all the advantages of CDMA systems, including frequency diversity[2] and out of cell (intercell) interference averaging—a spectral-efficiency benefit that narrowband systems such as conventional TDMA do not have.

As discussed earlier, different users within the same cell use different resources (tones) and hence do not interfere with each other. This is similar to TDMA, where different users in a cell transmit at different time slots and do not interfere with one another. In contrast, CDMA users in a cell do interfere with each other, increasing the total interference in the system. Flash-OFDM therefore has the physical-layer benefits of both CDMA and TDMA and is at least three times more efficient than CDMA. In other words, at the physical layer, flash-OFDM creates the fattest pipe of all cellular technologies. Even though the 3x advantage at the physical layer is a huge advantage, the most significant advantage of flash-OFDM for data is at the MAC and link layers.

MAC and Link-Layer Advantages

Flash-OFDM exploits the granular nature of resources in OFDM to come up with extremely efficient control layers. In OFDM, when designed appropriately, it is possible to send a very small amount (as little as one bit) of information from the transmitter to the receiver with virtually no overhead. Therefore, a transmitter that is previously not transmitting can start transmitting, transmit as little as one bit of information, and then stop, without causing any resource overhead. This is unlike CDMA or TDMA, in which the granularity is much coarser and to merely initiate a transmission wastes a significant resource. Hence, in TDMA, for example, there is a frame structure, and whenever a transmission is initiated, a minimum of one frame (a few hundred bits) of information is transmitted. The frame structure does not cause any significant inefficiency in user data transmission, as data traffic typically consists of a large number of bits. However, for transmission of control-layer information, the frame structure is extremely inefficient, as the control information typically consists of one or two bits but requires a whole frame. Not having a granular technology can therefore be very detrimental from a MAC– and link-layer point of view.

Flash-OFDM takes advantage of the granularity of OFDM in its control-layer design, enabling the MAC layer to perform efficient packet switching over the air and at the same time providing all the hooks to handle QoS. It also supports a link layer that uses local (as opposed to end-to-end) feedback to create a very reliable link from an unreliable wireless channel, with very low delays. The network layer's traffic therefore experiences small delays and no significant delay jitter. Hence, interactive applications such as (packet) voice can be supported. Moreover, Internet protocols such as TCP/IP run smoothly and efficiently over a flash-OFDM air link. TCP/IP performance on 3G networks is very inefficient because the link layer introduces significant delay jitter so that channel errors are misinterpreted by TCP as network congestion and TCP responds by backing off to the lowest rate.

Packet switching leads to efficient statistical multiplexing of data users and helps the wireless operators to support a much higher number of users for a given user experience. This, together with QoS support and a 3x fatter pipe, allows the operators to profitably scale their wireless networks to meet the burgeoning data-traffic demand in an all-you-can-eat pricing environment.

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4. SIMULATION OF OFDM MODEM IN SIMULINK:

The block diagram of the OFDM modem using simulink blocks is shown:

MODULATOR:

Bernoulli Random Binary Generator has been used as input which generates random binary ones and zeroes with a probability that can be set in the block properties. Then this data is mapped into 16 star QAM using differential PSK and ASK techniques. These complex symbols are then padded with zeroes to add inactive carriers for avoiding aliasing error and to make the IFFT block compatible. To reduce ISI and ICI effects suitable guard interval is added with cyclic extension technique. Now we get the required OFDM signal with advantages of multipath delay spread tolerance, reduced ICI & ISI effects and effective usage of system bandwidth etc.

CHANNEL:

An ideal channel is used for the system which is the AWGN block.

DEMODULATOR:

On the receiver side OFDM signal is received and then the guard interval is removed. then FFT block is used to recover the orthogonally spaced carriers and then the complex symbols are demodulated using differential star QAM technique to recover the original data

BER CALCULATION:

Overall bit error rate (BER) is calculated using the error calculation block.

The description of different blocks used in the system is given below:

4.1 Bernoulli Random Binary Generator:

Generate Bernoulli-distributed random binary numbers

Library:

Comm Sources

Description:

The Bernoulli Random Binary Generator block generates random binary numbers using a Bernoulli distribution. The Bernoulli distribution with parameter p produces zero with probability p and one with probability 1-p. The Bernoulli distribution has mean value 1-p and variance p(1-p). The Probability of a zero parameter specifies p, and can be any real number between zero and one.

4.2 Inport:

Create an input port for a subsystem or an external input.

Library:

Subsystems

Description

Inports are the links from outside a system into the system.

4.3 Demux:

Extract and output the elements of a bus or vector signal.

Library:

Signals & Systems

Description:

The Demux block extracts the components of an input signal and outputs the components as separate signals. The block accepts either vector (1-D array) signals or bus signals (see Signal Buses for more information). The Number of outputs parameter allows you to specify the number and, optionally, the dimensionality of each output port. If you do not specify the dimensionality of the outputs, the block determines the dimensionality of the outputs for you.

4.4 Frame Status Conversion:

Specify the frame status of the output, sample-based or frame-based.

Library:

Signal Management / Signal Attributes

Description:

The Frame Status Conversion block passes the input through to the output, and sets the output frame status to the Output signal parameter, which can be either Frame-based or Sample-based. The output frame status can also be inherited from the signal at the Ref (reference) input port, which is made visible by selecting the Inherit output frame status from Ref input port check box.

4.5 Bit to Integer Converter:

Map a vector of bits to a corresponding vector of integers

Library:

Utility Functions

Description:

The Bit to Integer Converter block maps groups of bits in the input vector to integers in the output vector. If M is the Number of bits per integer parameter, then the block maps each group of M bits to an integer between 0 and 2M-1. As a result, the output vector length is 1/M times the input vector length.

If the input is sample-based input, then it must be a vector whose length equals the Number of bits per integer parameter. If the input is frame-based, then it must be a column vector whose length is an integer multiple of Number of bits per integer.

4.6 M-DPSK Modulator Baseband:

Modulate using the M-ary differential phase shift keying method

Library:

PM, in Digital Baseband sublibrary of Modulation

Description:

The M-DPSK Modulator Baseband block modulates using the M-ary differential phase shift keying method. The output is a baseband representation of the modulated signal. The M-ary number parameter, M, is the number of possible output symbols that can immediately follow a given output symbol. The input must be a discrete-time signal.

4.7 Sum:

Output the sum of inputs.

Library:

Math

Description:

The Sum block adds scalar, vector, or matrix inputs or the elements of a single vector input

4.8 MATLAB Fcn:

Apply a MATLAB function or expression to the input.

Library:

Functions & Tables

Description:

The MATLAB Fcn block applies the specified MATLAB function or expression to the input. The output of the function must match the output dimensions of the block or an error occurs.

Here are some sample valid expressions for this block.

sin

atan2(u(1), u(2))

u(1)^u(2)

4.9 Integer Delay:

Delay an input by an integer number of sample periods.

Library:

Signal Operations

Description:

The Integer Delay block delays a discrete-time input by the number of sample intervals specified in the Delay parameter. Noninteger delay values are rounded to the nearest integer, and negative delays are clipped at 0.

4.10 Relay:

Switch output between two constants.

Library:

Nonlinear

Description:

The Relay block allows the output to switch between two specified values. When the relay is on, it remains on until the input drops below the value of the Switch off point parameter. When the relay is off, it remains off until the input exceeds the value of the Switch on point parameter. The block accepts one input and generates one output.

The Switch on point value must be greater than or equal to the Switch off point. Specifying a Switch on point value greater than the Switch off point value models hysteresis, whereas specifying equal values models a switch with a threshold at that value.

4.11 Product:

Generate the element-wise product, quotient, matrix product, or inverse of block inputs.

Library:

Math

Description:

The Product block outputs the element-wise or matrix product of its inputs, depending on the values of the Multiplication and Number of inputs parameters.

4.12 Outport:

Create an output port for a subsystem or an external output.

Library:

Subsystems

Description:

Outports are the links from a system to a destination outside the system.

4.13 Zero Pad:

Alter the input size by zero-padding or truncating rows and/or columns.

Library:

Signal Operations

Description:

The Zero Pad block changes the size of the input matrix from Mi-by-Ni to Mo-by-No by zero-padding or truncating along the rows, the columns, or both dimensions. The dimensions of the output, Mo and No, are specified by the Number of output rows and Number of output columns parameters, respectively. You can set Action when truncation occurs so that the block gives a warning or an error when truncation occurs.

4.14 Selector:

Select input elements from a vector or matrix signal.

Library:

Signals & Systems

Description:

The Selector block generates as output selected elements of an input vector or matrix.

A Selector block accepts either vector or matrix signals as input. Set the Input Type parameter to the type of signal (vector or matrix) that the block should accept in your model. The parameter dialog box and the block icon change to reflect the type of input that you select. The way the block determines the elements to select differs slightly, depending on the type of input.

4.15 IFFT:

Compute the IFFT of the input.

Library:

Transforms

Description:

The IFFT block computes the inverse fast Fourier transform (IFFT) of each channel in the M-by-N or length-M input, u, where M must be a power of two. To work with other input sizes, use the Zero Pad block to pad or truncate the length-M dimension to a power-of-two length.

The output is always frame-based, and each output column contains the M-point inverse discrete Fourier transform (IDFT) of the corresponding input channel.

4.16 AWGN Channel:

Add white Gaussian noise to the input signal

Library:

Channels

Description:

The AWGN Channel block adds white Gaussian noise to a real or complex input signal. When the input signal is real, this block adds real Gaussian noise and produces a real output signal. When the input signal is complex, this block adds complex Gaussian noise and produces a complex output signal. This block inherits its sample time from the input signal.

4.17 FFT:

Compute the FFT of the input.

Library:

Transforms

Description:

The FFT block computes the fast Fourier transform (FFT) of each channel in the M-by-N or length-M input, u, where M must be a power of two. To work with other input sizes, use the Zero Pad block to pad or truncate the length-M dimension to a power-of-two length. The output is always complex-valued and sample-based (it is unoriented 1-D vectors for unoriented inputs).

4.18 Abs:

Output the absolute value of the input.

Library:

Math

Description:

The Abs block generates as output the absolute value of the input.

Data Type Support:

An Abs block accepts a real- or complex-valued input of any type and outputs a real value of the same data type as the input.

4.19 Gain, Matrix Gain:

Multiply block input by a specified value.

Library:

Math

Description:

The Gain block generates its output by multiplying its input by a specified gain factor. You can enter the gain as a numeric value, or as a variable or expression in the Gain parameter field. The input and gain can be a scalar, vector, or matrix. The Multiplication parameter lets you specify whether to use element-by-element or matrix multiplication of the input by the gain.

The Gain block icon displays the value entered in the Gain parameter field if the block is large enough. If the gain is specified as a variable, the block displays the variable's name.

4.20 Relational Operator:

Perform the specified relational operation on the input.

Library:

Math

Description:

The Relational Operator block performs a relational operation on its two inputs and produces output according to the following table.

Operator	Output
==	TRUE if the first input is equal to the second input
~=	TRUE if the first input is not equal to the second input
<	TRUE if the first input is less than the second input
<=	TRUE if the first input is less than or equal to the second input
>=	TRUE if the first input is greater than or equal to the second input
>	TRUE if the first input is greater than the second input

If the result is TRUE, the output is 1; if FALSE, it is 0.

4.21 M-DPSK Demodulator Baseband:

Demodulate DPSK-modulated data

Library:

PM, in Digital Baseband sublibrary of Modulation

Description:

The M-DPSK Demo:dulator Baseband block demodulates a signal that was modulated using the M-ary differential phase shift keying method. The input is a baseband representation of the modulated signal. The input and output for this block are discrete-time signals. The input can be either a scalar or a frame-based column vector. The M-ary number parameter, M, is the number of possible output symbols that can immediately follow a given output symbol. The block compares the current symbol to the previous symbol. The block's first output is the initial condition of zero (or a group of zeros, if the Output type parameter is set to Bit) because there is no previous symbol.

4.22 Integer to Bit Converter:

Map a vector of integers to a vector of bits

Library:

Utility Functions

Description:

The Integer to Bit Converter block maps each integer in the input vector to a group of bits in the output vector. If M is the Number of bits per integer parameter, then the input integers must be between 0 and 2M-1. The block maps each integer to a group of M bits, using the first bit as the most significant bit. As a result, the output vector length is M times the input vector length. The input can be either a scalar or a frame-based column vector.

4.23 Error Rate Calculation:

Compute the bit error rate or symbol error rate of input data

Library:

Comm Sinks

Description:

The Error Rate Calculation block compares input data from a transmitter with input data from a receiver. It calculates the error rate as a running statistic, by dividing the total number of unequal pairs of data elements by the total number of input data elements from one source. You can use this block to compute either symbol or bit error rate, because it does not consider the magnitude of the difference between input data elements. If the inputs are bits, then the block computes the bit error rate. If the inputs are symbols, then it computes the symbol error rate. This block inherits the sample time of its inputs.

Input Data:

This block has between two and four input ports, depending on how you set the mask parameters. The inports marked Tx and Rx accept transmitted and received signals, respectively. The Tx and Rx signals must share the same sampling rate. The Tx and Rx inputs can be either scalars or frame-based column vectors. If Tx is a scalar and Rx is a vector, or vice-versa, then the
block compares the scalar with each element of the vector.

4.24 Display:

Show the value of the input.

Library:

Sinks

Description:

The Display block shows the value of its input. You can control the display format by selecting a Format choice.