MODULE-4

Syllabus: Analog and Digital Communication – Modern communication system scheme, Information source and input transducer, Transmitter, Channel – Hardwire and Softwire, Noise, Receiver, Multiplexing, Types of communication systems.

Text 3: 1.2, 1.2.1, 1.3, 1.4 – 1.4.1, 1.4.2, 1.5, 1.5.2, 1.6, 1.14, 1.15

From Summary portion of Chapter 1 of Text 3:

Types of modulation (only concepts) – AM (2.2), FM, Phase Modulation, Pulse Modulation, PAM (Fig.6.5b), PWM (Fig. 6.8), PPM, PCM.

Concept of Radio wave propagation (Ground, space, sky with Fig. 1.28)

From Summary portion of Chapter 6 – Digital Communication of Text 3:

Concepts of Sampling theorem, Nyquist rate, Digital Modulation Schemes (also see 6.12) – ASK, FSK, PSK

Radio signal transmission – Text 3: 6A.1.1, Fig. 6A.1, Fig. 6A.3

Multiple access techniques – Text 3: 6A.1.4, 6A.1.5

Multipath and fading – Text 3: 6A.2.1

Error Management – Text 3: 6A.3.1, 6A.3.2

Antenna, Types of antennas – Text 3: 13.1, 13.3 (only definition and antenna model, exclude radiation

patterns).

Chapter-Analog and Digital Communication

Modern Communication System Scheme

We know that communication is the science and practice of transmitting information. Further, communication engineering also deals with the techniques of transmitting information. In brief, communication engineering means *electrical communication*, in which information is transmitted through electrical signals. In this process, the information or message, *e.g.,* spoken words, live scenes, photographs, and sounds is first converted into electrical signals and then transmitted through electrical links. Thus, electrical *communication is a process by which the information/message is transmitted from one point to another, from one person to another, or from placeto another in the form of electrical signals, through some communication link.*

- A basic communication system provides a link between the information source and its destination.
- The process of electrical communication involves sending, receiving, and processing information in electrical form. A
- basic communication system consists of certain units, called constituents, *subsystems*, or *stages*.
- We may also note that the information to be transmitted passes through a number of stages of the communication system prior it reaches its destination.
- Figure 1.1 shows a block schematic diagram of the most general form of basic

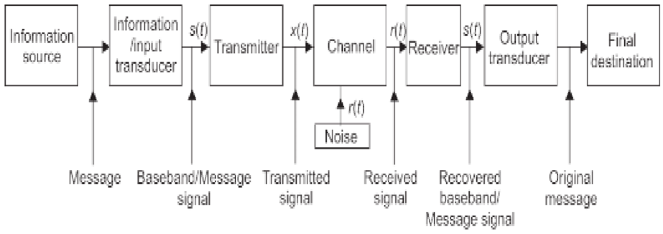


Fig. 1.1 Schematic block diagram of a basic communication system

In most general form We see from Fig. 1.1, the main constituents of basic communication system are:

(i) Information source and input transducer

(ii) Transmitter

(iii) Channel or medium

(iv) Noise

(v) Receiver

(vi) Output transducer and final destination.

- There are many types of communication systems, e.g., analog, digital, radio, and line communication systems.
- Figure 1.1 shows each type of communication system comprises the constituents. However, different communication systems apply different principles of operation and physical appearance to each constituent, in accordance with its type.

Now, we briefly describe each of the constituents or subsystems,

Information Source and Input Transducer

- A communication system transmits information from an information source to a destination and hence the first stage of a communication system is the *information source.*

- The physical form of information is represented by a message that is originated by an information source, *e.g.* a sentence or paragraph spoken by a person is a *message* that contains some information. The person,
- ***information source***. Few other familiar examples of messages are voice, live scenes, music, written text, and e-mail.
- A communication system transmits information in the form of electrical signal or signals.
- If the information produced by the source is not in an electrical form, one will have to use a device, known as *transducer, to* convert the information into electrical form.
- An example of a transducer is a *microphone.* Microphone converts sound signals into the corresponding electrical signals. Similarly, a television (TV) picture tube converts electrical signals into its corresponding pictures.

There are two *types of signals.*

(a) *analog signal,* and

(b) *digital signal.*

(a) Analog Signal

- An analog signal is a function of time, and has a continuous range of values. However, there is a definite function value of the analog signal at each point of time.
- A familiar example of analog signal or analog wave form is a *pure sine wave form.* A practical example of an analog signal is a *voice signal.*
- When a voice signal is converted to electrical form by a microphone, one gets a corresponding electrical analog signal. this wave form has definite values at all points of time.
- Analog signals are shown in Fig. 1.2

(b) Digital Signal

- A digital signal does not have continuous function values on a time scale.
- It is discrete in nature, *i.e.,* it has some values at discrete timings. In between two consecutive values, the signal values is either zero, or different value.
- A familiar example of a digital signal is the *sound signal* produced by *drumbeats.*

Figure 1.3 shows a graphical representational of a digital signal.

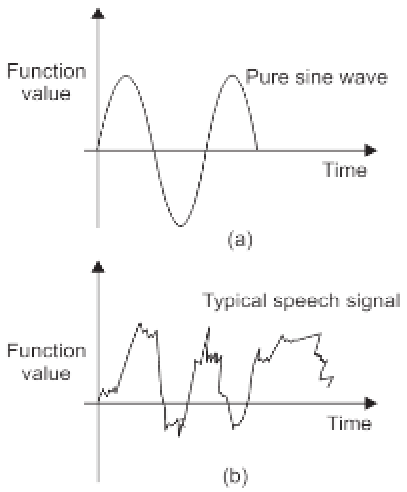


Fig. 1.2 Analog signals: (*a) Pure* sine wave *(b) Typical speech signal*

- Digital signals in their true sense correspond to a binary digital signal, where the discrete amplitude of the signal is coded into binary digits represented by '0' and '1'. The analog signal,
- Which is continuous in time, is converted to discrete time, using a procedure calling *sampling.*
- The continuous amplitude of the analog signal is converted to discrete amplitude using a process called *Quantization.*
- Sampling and quantization are together termed as *analog-to-digital* conversion (ADC) and the circuitry that performs this operation is called an *analog to-digital converter.*

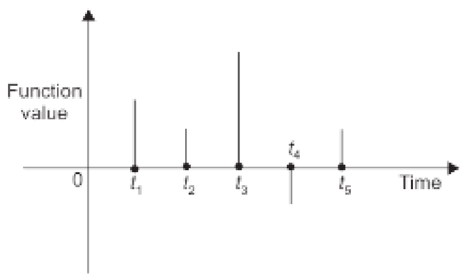


Fig. 1.3 Digital signal

Transmitter

The transmitter section processes the signal prior transmission. We may note that the nature of processing depends on the type of communication system. However, the processing carried out for signal transmission in the analog form is different from signal transmission in the digital form.

There are two following options for processing signals prior transmission:

(**i**) The baseband signal, which lies in the low frequency spectrum, is translated to a higher frequency spectrum.

(ii) The baseband signal is transmitted without translating it to a higher frequency spectrum.

- In the former case, we call the communication system as a ***carrier communication system***. In this system, the baseband signal is carried by a higher frequency signal, called the ***carrier signal***.
- In the latter case, we call the system as a ***baseband communication system***, because the baseband signal is transmitted without translating it to a higher frequency spectrum.
- However, some processing of the signal is required prior its transmission, *e.g.* a train of pulses that are to be transmitted can be replaced by a series of two sine waves of different frequencies prior to transmission.
- One of these two frequencies represent a low and the other represents a high value of the digital pulse. Therefore, the baseband signal is converted into a corresponding series of sine waves of two different frequencies prior to transmission.
- Figure 1.4 illustrates this processing.

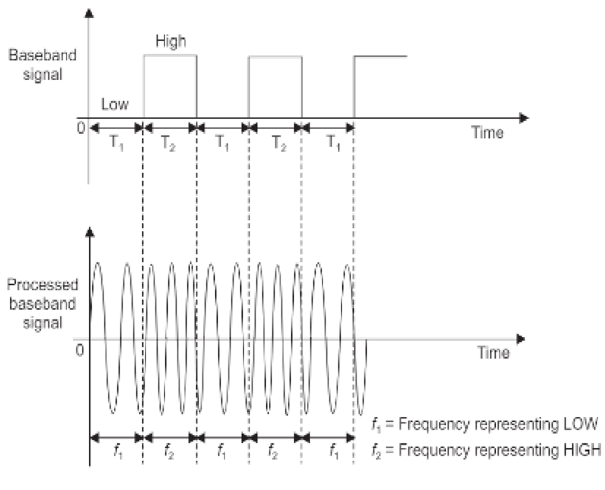


Fig. 1.4 The processing of a baseband signal

- The carrier communication system is based on the principle of translating a low frequency baseband signal to higher frequency spectrum. This process is termed as ***modulation.***
- If the baseband signal is a digital signal, the carrier communication system is called a ***digital communication*** **system**. The digital modulation methods are employed for this.
- If the baseband signal is an analog signal, the carrier communication system is called as an ***analog communication system*** and for processing the analog modulation techniques are used.
- Figure 1.5 shows the baseband signal, *s(t) applied* to the modulated stage. This stage translates the baseband signal from its low frequency spectrum to high frequency spectrum.

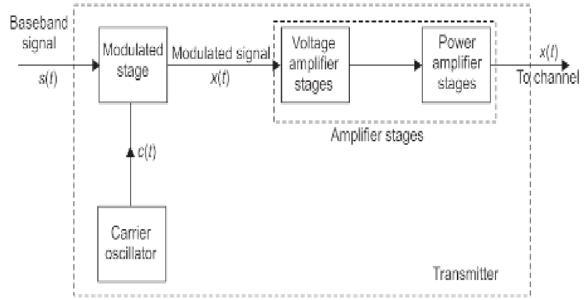


Fig. 1.5 Block diagram representing schematic of an analog transmitter section

- This stage also receives another input called the ***carrier signal, c(t),*** which is generated by a high frequency ***carrier oscillator***.
- Modulation takes place at this stage with the baseband and the carrier signals as two inputs
- After modulation, the baseband signal is translated to a high frequency spectrum and the carrier signal is said to be modulated by the baseband signal.
- The output of the modulated stage is called the ***modulated signal***, and is designated as ***x(t).***
- The voltage of the modulated signal is then amplified to drive the last stage of the transmitter, called the ***power amplifier stage*** (Fig. 1.5). This stage amplifies the power of the modulated signal and thus it carries enough power to reach the receiver stage of the communication system.
- Finally, the signal is passed to the transmission medium or channel.

Radio signals are transmitted through ***electromagnetic (em) waves***, also referred as ***radio waves***, in a radio communication system. The radio waves have a wide frequency range starting from a few ten kilo Hertz (Hz) to several thousand Mega Hertz (MHz). This wide range of frequencies is referred as the ***radio frequency*** **(RF)** ***spectrum.***

The RF spectrum is classified according to the applications of the spectrum in different service areas

Table 1.1 shows the classification of the RF spectrum along with the associated applications in communication systems.

Table 1.1: Classification of radio frequency (RF) spectrum along with the associated applications in communication systems.

Radio frequency range	Wavelength (meters)	Class	Applications
10–30 kHz	$3 \times 10^4 - 10^4$	Very Low Frequency (VLF)	Point-to-point communication (long distance)
30–300 kHz	$10^4 - 10^3$	Low Frequency (LF)	Point-to-point communication (long distance) and navigation
300–3000 kHz	$10^3 - 10^2$	Medium Frequency (MF)	Radio broadcasting
3–30 MHz	$10^2 - 10$	High Frequency (HF)	Overseas radio broadcasting, Point-to-point radio telegraphy, and telephony
30–300 MHz	10 – 1.0	Very High Frequency (VHF)	FM broadcast, television, and radar
300–3000 MHz	1.0 – 0.1	Ultra High Frequency (UHF)	Television and navigation
3000–30,000 MHz	0.1 – 0.01	Super High Frequency (SHF)	Radar navigation and radio relays

Channel Or Medium

After the required processing, the transmitter section passes the signal to the transmission medium. The signal propagates through the transmission medium and is received at the other side by the receiver section. The transmission medium between the transmitter and the receiver is called a ***channel***.

- ***Channel*** is a very important part of a communication system as its characteristics add many constraints to the design of the communication system, e.g. most of the ***noise*** is added to the signal during its transmission through the channel.
- The transmitted signal should have adequate power to withstand the channel noise. Further, the channel characteristics also impose constraints on the ***bandwidth***.
- The bandwidth is the frequency range that can be transmitted by a communication system. Moreover, the channel characteristics are also taken into consideration as a *design parameter* while designing the transmitting and receiving equipment.

Depending on the physical implementations, one can classify the channels in the following two groups:

Hardware Channels

These channels are manmade structure which can be used as transmission medium. There are following three possible implementations of the hardware channels

- **Transmission lines**
- **Waveguides**
- **Optical Fiber Cables (OFC)**

The examples of transmission lines are ***twisted-pair cables*** used in landline telephony and ***coaxial cables*** used for cable TV transmission.

- However, transmission lines are not suitable for ultra-high frequency (UHF) transmission. To transmit signals at UHF range, *waveguides* are employed as medium.
- *Optical fiber cables* are highly sophisticated transmission media, in the form of extremely thin circular pipes. Signals are transmitted in the form of light energy in optical fiber cables.
- In general, there is always a physical link between the transmitter and receiver in hardware channels. A communication system that makes use of a hardware channel is called as a *line communication* system, *e.g.* landline telephony and cable TV network.

Software Channels

- There are certain natural resources which can be used as the transmission medium for signals. Such transmission media are called *software channels*. The possible natural resources that can be used as software channels are: *air or open space* and *sea water*.
- Note that in communication systems that use software channels there is no physical link between the transmitter and the receiver.
- The transmitter passes the signals in the required form to the software channel. The signals propage through the natural resource and reach the receiver.
- The signals are transmitted in the form of ***electromagnetic* (em) *waves'***, also called ***radio waves***.
- Radio waves travel through open space at a speed equal to that of light (c = 3 x 1O ms').
- The transmitter section converts the electrical signal into em waves or radiation by using a *transmitting antenna*. These waves are radiated into the open space by the transmitting antenna.
- At the receiver side, another antenna, called the receiving antenna, is used to pick up these radio waves and convert them into corresponding electrical signals.
- Systems that us radio waves to transmit signals through open space are called *radio communication*

systems, e.g., radio broad cast, television transmission, satellite communication, and cellular mobile communication.

Noise

- In electronics and communication engineering, noise is defined as *unwanted electrical energy of random and unpredictable nature present in the system due to any cause.*
- Obviously, *noise is an electrical disturbance, which does not contain any useful information.* Thus, noise is a highly undesirable part of a communication system, and have to be minimized.
- Noise cannot be eliminated once it is mixed with the signal. When noise is mixed with the transmitted signal, it rides over it and deteriorates it waveform. This results in the alteration of the original information so that wrong information is received.
- The receiver processes the signal to cover the original information produced by the information source at the transmitter end.
- If the amplitude of the noise is comparable with that of the signal, then the noise may render the transmitted signal unintelligible, and the receiver recovers nothing but the noise. In order to avoid this underiable situation, the system designer can make the signal adequately powerful prior to transmitting it.
- This enables the signal to withstand the noise. In fact, the system designer increases the power of the signal in comparison with that of the noise. This increases the ratio of the signal power to the noise power, *i.e.,* **SNR** ***(signal to noise ratio).***
- The designer provides adequate signal strength at the time of transmission so that a high SNR is available at the receiver.
- The noise block in Fig. 1.1 represents the total noise present in the system, contributed by all the sources.
- The noise signal n*(t)*, is applied to the channel block. However, this does not mean that the noise is intermingled with the signal only during its propagation through the channel.
- In fact, the channel contributes the major part of the noise. However, other noise sources along the communication chain can also add noise to the signal.
- The noise may also be mixed with the signal from within the transmitting and receiving equipment.

Since it is not possible to show all the individual sources of noise along the communication chain.

- The noise introduced by the transmission medium is called ***extraneous or external noise.*** The main cause of the ***internal noise*** is the thermal agitation of atoms and electrons of electronic components used in the equipment.

SNR and Noise Figure (F)

One can define the SNR as the ratio of the signal power to the noise power at a point in the circuit. Note that SNR is the measure of the signal power relative to the noise power at a particular point in a circuit. Now, if P_s is signal power and P_n is noise power, then SNR expressed as S/N, is given as

$$\frac{S}{N} = \frac{P_s}{P_n}$$

If $P_s = V_s^2 R$ and $P_n = V_n^2 R$, then

$$\frac{S}{R} = \frac{P_s}{P_n} = \frac{V_s^2 R}{V_n^2 R} \qquad ...(2)$$

where V_s is signal voltage and V_n is noise voltage.

In addition, it is assumed that both the signal and noise powers are dissipated in the same resistor R. Therefore, SNR can be expressed in terms of decibels (*dB*) as

$$\left(\frac{S}{N}\right)_{dB} = 10 \log_{10}\left(\frac{V_s^2}{V_n^2}\right)$$

$$\left(\frac{S}{N}\right)_{dB} = 20 \log_{10}\left(\frac{V_s}{V_n}\right) \qquad ...(3)$$

For example, if, at a particular point in a circuit, the signal and noise voltages are given as 3.5 mV and 0.75 mV, respectively, SNR in *dB* is calculated as:

$$\left(\frac{S}{N}\right)_{dB} = 20 \log_{10}\left(\frac{3.5}{0.75}\right)$$

or

$$\text{SNR} = 20 \log_{10}(4.66)$$

$$= 13.38\ dB$$

Clearly, the SNR of the circuit at the point is 13.38 *db.*

The *noise figure* (F) is the measure of the noise introduced by the circuit. It is defined as the ratio of the signal-to-noise power at the input of the circuit and the signal-to-noise power at the output of the circuit. Noise figure (F) can be expressed as

$$F = \frac{\frac{S}{N} \text{ Power at the input terminals of the circuit}}{\frac{S}{N} \text{ Power at the output terminals of the circuit}} \qquad ...(4)$$

We can see that if F is unity, the noise power introduced by the circuit is zero, as both the input and output S/N powers are the same.

Receiver

- The task of the receiver is to provide the original information to the user. This information is altered due to the processing at the transmitter side. The signal received by the receiver, thus, does not contain information in its original form.
- The receiver system receives the transmitted signal and performs some processing on it to recover the original baseband signal.
- Marked the signal received by the receiver by *r(t)* in Fig. 1.1. This signal contains both the transmitted signal, x(t), and the noise, *n(t)*, added to it during transmission.
- The function of the receiver section is to separate the noise from the received signal, and then recover the original baseband signal by performing some processing on it.
- The receiver receives a weak signal because the transmitted signal losses its strength during its propagation through the channel. This occurs due to the attenuation of the signal.
- A voltage amplifier first amplifies the received signal so that it becomes strong enough for further processing, and then recovers the original information.
- The original baseband signal is recovered by performing an operation opposite to the one performed by the transmitter section. The transmitter performs ***modulation*** on the baseband signal to translate it to a higher spectrum from its low frequency spectrum. The receiver, in turn, performs an operation known as ***demodulation***, which brings the baseband signal from the higher frequency spectrum to its original low-frequency spectrum.
- The demodulation process removes the high frequency carrier from the received signal and retrieves the original baseband. This occurs in a carrier communication system.
- The baseband communication systems, assume that the transmitter replaces the digital baseband signal with a series of two sinusoidal wave forms of different frequencies as shown in Fig. 1.4. When the receiver receives this signal, it recovers the original baseband pulse by replacing the two sinusoidal waveforms with the corresponding original levels.
- The recovered baseband signal is then handed over to the final destination, which uses a transducer to convert this electrical signal to its original form. It is essential that enough signal power is given to the transducer so that it satisfactorily reproduces the message. Therefore, prior to handing over the recovered baseband signal to its final destination,
- The voltage and power are amplified by the amplifier stages.

The detailed block diagram of a typical receiver section is shown in Fig. 1.7.

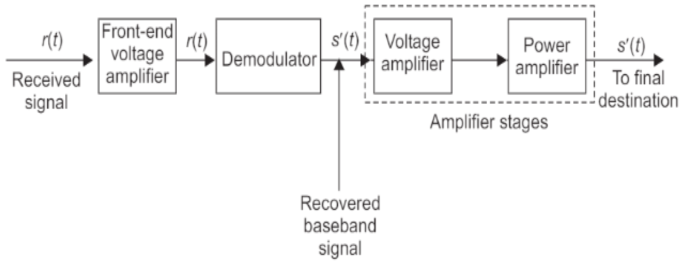


Fig. 1.7 Detailed block diagram of a typical receive section

- From Fig. 1.7 it is evident that the received signal, *r(t)*, is first amplified by the front-end voltage amplifier. This is done to strengthen the received signal, which is weak and to facilitate easy processing. Next, this signal is given to the demodulator, which in turn, demodulates the received signal to recover the original baseband signal.
- Interestingly, the type of demodulation is based on the type of modulation employed at the transmitter. After recovering the original baseband signal, its voltage and power is amplified prior it to final destination block.

Multiplexing

This is a technique that is most widely used in nearly all types of communication systems, radio and line communication systems. Basically, multiplexing is a process which allows more than one signal to transmit through a single channel. Clearly, multiplexing facilitates the simultaneous transmission of multiple messages over a single transmission channel

- Multiplexing allows the maximum possible utilization of the available bandwidth of the system. Bandwidth is an important entity in any communication system.
- The use of multiplexing also makes the communication system economical because more than one signal can be transmitted through a single channel.
- Multiplexing is possible in communication system only through modulation.

If all these baseband audio signals are simultaneously transmitted through a single channel, then they will be mixed together. The transmitter will transmit these mixed signals, and the receiver will receive them.

The purpose of the receiver is to deliver the audio signals in their original form. However, all the received signals lie within the same audio range, and the receiver is not capable of separating them into individual signals, similar to the case with human ears. In order to avoid this difficulty, each signal can be translated to a different frequency

spectrum, such that every signal differs in its transmitted frequency. This is done through modulation. Therefore, if all the baseband signals are modulated, i.e., *translated* to higher frequencies by using different carrier frequencies, then each signal is easily distinguishable from the other although they all lie within the same audio band. At the transmitter they can be mixed and transmitted. At the receiver, the different signals can be easily separated because they are at different frequencies, and these can be delivered to the next stages of the receiver for further processing. Obviously, multiplexing becomes possible only because of modulation.

Types Of Communication Systems

One may categorize communication systems based on their ***physical infrastructure*** and the ***specifications of the signals they transmit***. The physical infrastructure pertains to the type of the channel used and the hardware design of the transmitting and receiving equipment.

Communication Systems based on Physical Infrastructure

There are two types of communication systems based on the physical infrastructure

(i) Line Communication Systems

There is a physical link, called the hardware channel, between the transmitter and the receiver in the line communication systems. In a radio communication system, there is no such link and natural resources, such as space and water are used as software ***channels***. A particular communication system can be one of these two types, e.g., *radio broad cast is* a purely radio communication system and cannot be categorized as a line communication system. on the other hand, landline *telephony* is purely a line communication system and cannot be typed as a radio communication system.

Let us consider a TV system in which a user can only receive the signals and view available channels. A television receiver cannot transmit the signals. In another example, we consider telephony. In this case, one can simultaneously send and receive signals. ***TV transmission* is a *one-way transmission***. This is called as *simplex*, while a two-way transmission is called ***duplex***. A derivative of duplex is ***half duplex***, *in* which two-way transmission is carried out, but not simultaneously. In this system, the signal can either be sent or received at a time.

The one-way or two-way transmission feature of a communication system depends on the design of the equipment used on the two sides of the communication system, and is therefore included in the physical structure specifications of the system. As a rule, a communication system can be ***simplex* or a *duplex***, but not both.

Obviously, based on the physical structure of a communication system, one can define two groups, and only one specification from each group is required to decide the type of communication system. These groups are:

- **Line/radio comuncation**
- **Simplex/ duplex communication**

For example, a TV communication system is a combination of the radio and simplex communication system and landline telephony is a combination of the line and duplex communication systems. A particular communication system can be implemented as both line and radio communication system, *e.g.* landline telephony is a line communication system. However, overseas or long-distance telephony is carried out through satellites and the system is called radio telephony as it makes use of radio waves for transmission and reception. This system is then categorised as radio communication system

(ii) Communication Systems Based on Signal Specifications

The signal specifications used to decide the type of communication include:

- Nature of *baseband* or informal signal
- Nature of the *transmitted* signal.

Based on the nature of the baseband signal, there are two types of communication systems:

- Analog communication systems

- Digital communication systems.

Based on the nature of the transmitted signal, the baseband signal can either be transmitted as it is, without modulation, or through a carrier signal with modulation. The two systems can then be put under following categories:

- Baseband communication system
- Carrier communication system

Thus, there are four types of communication system categories based on *signal* specification. These are:

- Analog communication system
- Digital communication system
- Baseband communication system
- Carrier communication system

Of the four, at least two types are required to specify a particular communication system. Thus, one can form two groups consisting each of two types such that at least one of the types from each group is necessarily required to specify a communication system. These groups can be put as:

- Analog/ digital communication system
- Baseband/ carrier Communication system

We may note that a particular communication system is either an ***analog communication*** system or a ***digital communication*** system at a time. For example, TV transmission is an analog communication system while *high-definition television* (HDTV) is a digital communication system. Another example of a digital communication system is *internet.*

Similarly, a particular communication system is either a ***baseband*** communication system or a carrier communication system. Examples of ***baseband*** communication systems are ***landline telephony*** and *Fax.* Examples of carrier communication systems are TV transmission, radio broadcast, and able TV

Obviously, to completely describe a particular communication system four out of the eight types of communication systems are required. If any one type is missing, then the description of the communication system will be incomplete. In order to make things simple as well as clear, *four groups* containing these *eight types* are formed, such that only one choice is possible from each group and the four choices together describe the communication system. These four are listed below:

- **Group I**: Line/radio communication system
- **Group II**: Simplex/ duplex communication system
- **Group III**: Analog/ digital communication system
- **Group IV**: Baseband/ carrier communication system

We consider the ***television communication*** *system.* It is described as a radio communication system from group I, simplex communication system from group II, analog communication system from group III, and carrier communication system from group IV. This means, a ***television communication system* is a *radio-simplex-analog-carrier*** communication system. Obviously, this is the description of a television communication system in its totality.

Now, we consider another example of ***landline telephony***. This is described as line communication system from group I, duplex communication system from group II, analog communication system from group III, and baseband communication system from group IV. Obviously, *telephony* through landlines is a ***line-duplex-analog-baseband*** communication system. Clearly, one can

completely describe any communication system with these four groups.

Types of modulation

The *amplitude modulation* is produced by varying the amplitude of the carrier waves in accordance with the amplitude of the modulated wave (audio signal) which as the frequency is kept fixed fixed equal to the frequency of the carrier wave. The modulation index (or modulation factor) *m*, for amplitude modulation is defined as

$$m_a = \frac{E_{max} - E_{min}}{E_{max} + E_{min}}$$

where $E_{max} = E_c + E_m$ and $E_{min} = E_c - E_m$ are maximum and minimum values of amplitude and E_c is the amplitude of the carrier wave and E_m is the amplitude of modulating audio frequency signal. In amplitude modulation, bandwidth = 2 × frequency of audio signal

- If $m(t) = A_m \sin \omega_m t$ represents modulated signal and $m_c(t) = A_c \sin \omega_c t$ represents carrier signal, than the instantaneous voltage of the A.M. wave is obtained as

$$C_m(t) = A_c \sin \omega_c t + \frac{\mu A_c}{2}$$

$$[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t] \qquad ...(i)$$

This is the expression for *amplitude modulated wave.*

Eq. (i) reveals that the A.M. wave is the sum of *three sinusoidal waves, i.e.*

(i) The *original carrier wave i.e.* first wave having amplitude A_c and frequency $f_c = \omega_c/2$

(ii) The second wave having amplitude $\mu A_c/2$ and frequency *(fc - fm)*. The sum of carrier frequency and modulating frequency, i.e. *(fc + fm)* is termed as *upper side band frequency* (USB).

(iii) The third wave have the amplitude $\mu A_c/2$ and frequency $(f_c - f_m)$. This difference of carrier wave frequency and modulating frequency *[i.e. (fc - fm)]* is termed as *lower sideband frequency (LSB)*

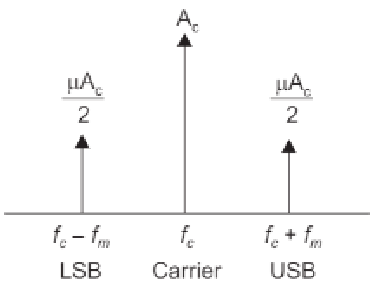


Fig. 1.19 Frequency spectrum of A.M. wave

- One can represent the frequency spectrum of A.M. wave is as shown in Fig. 1.19 below

$$\text{Bandwidth} = (f_c + f_m) - (f_c - f_m) = 2f_m$$

- Obviously, *bandwidth* of A.M. wave is the difference between the highest and lowest frequencies present in the A.M. wave and is twice the frequency of the modulated signal.
- A modulating signal may vary the frequency of the carrier keeping the amplitude and phase constant. This type of modulation is called *frequency modulation.* Broadly speaking, the frequency modulation is the process of changing the frequency of the carrier voltage in accordance with the instantaneous value of the modulating voltage. The original frequency of the carrier signal is called centre or resting frequency and denoted by f. The amount by which the frequency of the carrier wave changes or shifts above or below the resting frequency is termed as frequency deviation (Δf). This means $\Delta f \propto m(t)$.
- The total variation is frequency of F.M. wave from the lowest to the highest is termed as carrier saving (CS), *i.e., CS* = 2 x frequency deviation in centre frequency or $CS = 2\ \Delta f$
- Modulation index in F.M. is the ratio of frequency deviation to the modulating frequency, *i.e.*

$$\mu_f = \frac{\text{Frequency deviation}}{\text{Modulating frequency}} = \frac{\Delta f}{f_m}$$

- When the various sidebands are separated by the modulating frequency *fm*, then bandwidth $= 2\, n \times fm$, where *n* is the number of significant bands present in the F.M. wave and f, is the modulating frequency.
- *Phase modulation* is another form of angle modulation. Phase modulation (PM) is the process in which amplitudes of the regularly spaced rectangular rules vary in direct proportion to the instantaneous values of continuous message signal. Phase modulation and frequency modulation are closely related to each other. In both the cases, the total phase angle ϕ of the modulated signal varies.
- *Phase modulation* is the process in which the instantaneous phase of the carrier signal is varied in accordance with the instantaneous amplitude of the modulating signal. In this type of modulation, the amplitude and frequency of the carrier signal remains unaltered after pulse modulation. The modulating signal is mapped to the carrier signal in the form of variations in the instantaneous phase of the carrier signal.
- *Pulse modulation* may be used to transmit analog information, such as continuous speech or data. It is system in which continuous waveforms are sampled at regular intervals.
- *Pulse modulation* may be subdivided into two categories, *analog* and *digital.* In the former, the indication of sample amplitude may be indefinitely variable while in the latter a code which indicates the sample amplitude to the nearest predetermined level is sent. *Pulse-amplitude* and *pulse-time* modulation are both analog, while the *pulse code* and *delta modulation* systems are both digital.

(i) Phase-amplitude modulation (PAM):

PAM is the simplest form of pulse modulation. PAM is a pulse modulation system in which the signal is sampled at regular intervals, and each sample is made proportional to the amplitude of the signal at the instant of sampling. The pulses are then sent by either wire or cable, or else are used to modulate a carrier. The ability to use constant-amplitude pulses is a major advantage of pulse modulation, and since PAM does not use constant• amplitude pulses, it is infrequently used. When it is used, the pulses frequency• modulate the carrier.

(ii) **Pulse width or pulse-duration modulation (PWM or PDM):**

In this system, the starting time and amplitude of each pulse are constant but the width or duration of each pulse is made proportional to the instantaneous value of analog signal.

PDM has the disadvantage, when compared with pulse-position modulation (PPM), that its pulses are of varying width and therefore of varying power content. This means that the transmitter must be powerful enough to handle the maximum• width pulses, although the average power transmitted is perhaps only half of the peak power. PWM still works if synchronization between transmitter and receiver fails, whereas PPM does not.

(iii) **Pulse position-modulation (PPM):**

In this system, the amplitude and width of the pulses is kept constant, while the position of each pulse, in relation to the position of a recurrent reference pulse is varied by instantaneous sampled value of the modulating wave. As compared to PWM, PPM has the advantage of requiring constant transmitter power output, but the disadvantage of depending on transmitter receiver synchronization.

In this system, the position of pulse relative to its unmodulated time of occurrence is varied in accordance with the instantaneous value of the message signal.

(iv) **Pulse-code modulation (PCM):**

PCM is a digital process in which the message signal is sampled is rounded off to the nearest value of a finite set of allowable values and rounded values are coded. PCM generator produces a series of numbers or digits. Each one of these digits, almost always in binary code, represents the approximate amplitude of the signal sample at that instant. Obviously, signals are transmitted as binary code.

Depending primarily on the frequency a radio wave travels from he transmitting to the receiving antenna in several ways. On the basis of the mode of propagation, radio waves can be broadly classified as:

(i) **ground or surface wave** *(ii)* **space or tropospheric wave, and** *(iii)* **sky way Accordingly,**

we have three types of propagation:

(i) **Ground wave propagation**:

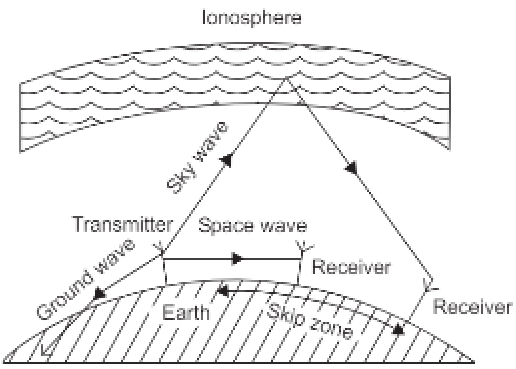
In ground wave propagation, radio waves are guided by the earth and move along its curved surface from the transmitter to the receiver. As the waves moves over the ground, they are strongly influenced by the electrical properties of the ground. As high frequency waves are strongly absorbed by ground; ground wave propagation is useful only at low frequencies. Below 500 kHz, ground waves can be used for communication within distances of about 1500 km from the transmitter. AM radio broadcast in the medium frequency band cover local areas and take place primarily by the ground wave. The ground waves at higher frequencies employed by frequency modulation (FM) and television (TV) are increasing absorbed and therefore become very weak beyond a distance of several kilometres from the transmitter. Ground wave transmission is very reliable whatever the atmospheric conditions be.

(ii) **Space or tropospheric wave propagation**:

When a radio wave transmitted from an antenna, travelling in a straight line directly reaches the receiving antenna, it is termed as space or tropospheric wave. In space wave or line of sight propagation, radio waves move in the earth's troposphere within about 15 km over the surface of the earth. The space wave is made up of two components: *(a) a direct or line-of• sight wave* form the transmitting to the receiving antenna and *(b)* the *ground-reflected wave* traversing form the transmitting antenna to ground and reflected to the receiving antenna. Television frequencies in the range 100-220 MHz are transmitted through this mode.

(iii) **Sky wave propagation**:

In this mode of propagation, radio waves transmitted from the transmitting antenna reach the receiving antenna after reflection form the ionosphere, i.e. the ionized layers lying in the earth's upper atmosphere. Short wave transmission around the globe is possible through sky wave via successive reflections at the ionosphere and the earth's surface.



- The *ionized region* of the earth's upper atmosphere extending from about 40 km to the height of a few earth radii above the earth, is referred to as the *ionosphere.* The ionosphere is made up of electrons, and positive and negative ions in the background of neutral particles of the atmosphere.

- *The propagation of radio wave through the ionosphere* is affected by the electrons and ions in the ionosphere. The effect of the electrons on the propagation is much greater than that of the ions since the electronic mass is much less than the ionic mass.

Chapter- Digital Communication

Digital Modulation Techniques

When it is required to transmit digital signals on a bandpass channel, the amplitude, frequency or phase of the sinusoidal carrier is varied in accordance with the incoming digital data. And for this purpose, digital modulation techniques are used.

Digital modulation techniques may be classified into coherent and non-coherent techniques, depending upon whether the receiver consists of a phase-recovery circuit or not. The phase recovery circuit ensures that the oscillator supplying the locally generated carrier signal in the receiver is synchronized to the oscillator supplying the carrier signal used to originally modulate the incoming data stream in the transmitter. These techniques are discussed as follows:

Coherent Digital Modulation Technique

These techniques employ coherent detection in which the local carrier generated at the receiver is phase locked with the carrier at the transmitter. Hence, the detection is done y correlating the received noisy signal and locally generated carrier. It is also known as synchronous detection.

Non-Coherent Digital Modulation Techniques

In these techniques, the detection process does not require receiver carrier to be phase locked with the transmitter carrier. So, the system becomes simple in design. But, on the other hand, in such systems, the probability of error increases.

There are two types of signals, continuous-time and discrete-time signals.

- The processing of discrete-time signals is more flexible and is also preferable than the continuous-time signals.
- The sampling theorem governs the conversion of continuous-time signal into discrete time signal.
- The concept of sampling provides a widely used method for using discrete-time system technology to implement continuous-time systems and process the continuous-time signals.
- A continuous-time signal may be completely represented in its samples and recovered back if the sampling frequency is f, > 2f, here, *f*, is the sampling frequency and *f*, is the maximum frequency present in the signal.
- When the sampling rate becomes exactly equal to 2f, samples per second, then it is called Nyquist rate. Nyquist rate is the minimum sampling rate.
- A low pass filter is used to recover the original signal from its samples.
- The process of reconstructing the continuous-time signal from its samples is known as interpolation.
- When the sampling frequency is less than the Nyquist rate, aliasing problem is said to occur.
- Aliasing is the phenomenon in which a high frequency component in the frequency spectrum of the signal takes the identity of a lower frequency component in the spectrum of the sampled signal.
- To avoid aliasing:
  - Prealias filter must be used to limit the band of frequencies of the signal to f_m Hz
  - Sampling frequency must be selected such that $f_s \geq 2f$,
- There are two types of pulse modulation schemes as follows
  - Pulse Amplitude Modulation (PAM)
  - Pulse Time Modulation (PTM)
- In PAM the amplitude of the pulses of the carrier pulse train is varied in accordance with the modulating signal whereas, in PTM, the timing of the pulses of the carrier pulse train is varied
- There are two types of PTM:
  - Pulse Width Modulation (PWM)
  - Pulse Position Modulation (PPM)
- In PWM, the width of the pulses of the carrier pulse train is varied in accordance with the modulating signal whereas, in PPM, the position of the pulses of the carrier pulse train is varied.
- All of the above pulse modulation methods (*i.e.,* PAM, PWM and PPM) are called analog pulse modulation methods because the modulating signal is analog in nature.

- In digital communications, the modulating signal consists of binary data or an M-ary version of it.
- When it is required to transmit digital signals on a bandpass channel the amplitude, frequency or phase of the sinusoidal carrier is varied in accordance with the incoming digital data.
- Since, the digital data is in discrete steps, the modulation of the bandpass sinusoidal carrier is also done in discrete steps. Due to this reason, this type of modulation is known as digital modulation.
- Digital modulation schemes as classified as under:
  - **Amplitude Shift Keying (ASK)**
  - **Frequency Shift Keying (FSK)**
  - **Phase Shift Keying (PSK)**
- Because of constant amplitude of FSK or PSK, the effect of non-linearities, noise interference is minimum on signal detection. However, these effects are more pronounced on ASK. Therefore, FSK and PSK are preferred over ASK.
- Coherent digital modulation techniques are those techniques which employ coherent detection. In coherent detection, the local carrier generated at the receiver is phase locked with the carrier at the transmitter. Thus, the detection is done by correlating the received noisy signal and locally generated carrier. The coherent detection is also called synchronous detection
- ASK signal may be generated by simply applying the incoming binary data and the sinusoidal carrier to the two inputs of a product modulator.
- The demodulation of binary ASK waveform can be achieved with the help of coherent detector.

6A.1.1 Signal Transmission

Figure 6A.1 shows the most important components of a wireless transmission system. In the figure, the transmitter accepts a stream of bits from the application software. It then encodes these bits onto a radio wave, known as a *carrier*, by adjusting parameters of the wave such as its amplitude or phase.

As shown in the figure, the transmitter usually processes the information in two stages. In the first stage, a *modulator* accepts the incoming bits, and computes *symbols* that represent the amplitude and phase of the outgoing wave. It then passes these to the analogue transmitter, which generates the radio wave itself.

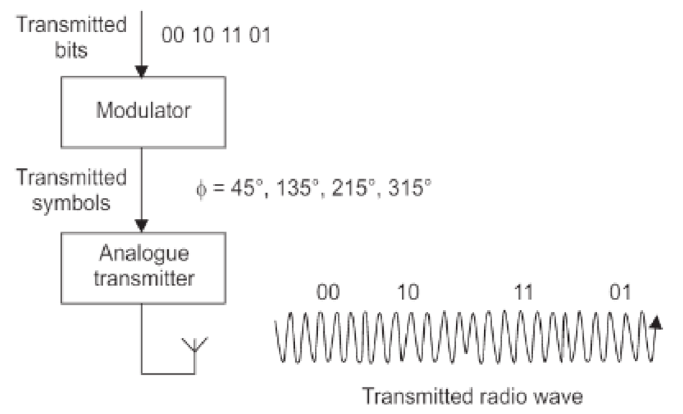


Fig. 6A.1 Architecture of a wireless communication transmitter

The modulation scheme used in Fig. 6A.1 is known as *quadrature phase shift keying* (QPSK). A QPSK modulator takes the incoming bits two at a time and transmits them using a radio wave that can have four different states. These have phases of 45, 135, 225° and 315°

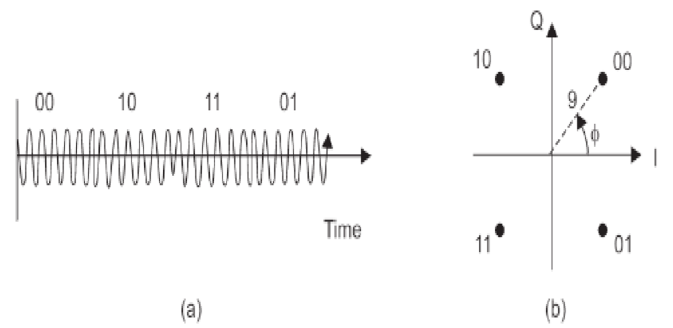


Fig. 6A.2 Quadrature phase shift keying. *(a)* Example QPSK waveform. *(b)* QPSK constellation diagram

(Fig. 6A.2a), which correspond to bit combinations of 00, 10, 11 and 01 respectively. We can represent the four states of QPSK using the *constellation diagram* shown in Fig. 6A.2b. In this diagram, the distance of each state from the origin represents the amplitude of the transmitted wave, while the angle (measured anti-clockwise from the x-axis) represents its phase.

Usually, it is more convenient to represent each symbol using two other numbers, which are known as the *in-phase* (I) and quadrature (Q) components. These are computed as follows:

$$I = a \cos \varphi$$

$$Q = a \sin \varphi \qquad \ldots\ldots \quad (1)$$

where a *is* the amplitude of the transmitted wave and φ is its phase. Mathematicians will recognize the in-phase and quadrature components as the real and imaginary parts of a complex number.

As shown in Fig. 6A.3, **LTE** uses four modulation schemes altogether. *Binary phase shift keying* (BPSK) sends bits one at a time, using two states that can be interpreted as starting phases of 0° and 180°, or as signal amplitudes of +1 and –1. LTE uses this scheme for a limited number of control streams, but does not use it for normal data transmissions. *16 quadrature amplitude modulation* (16-QAM) sends bits four at a time, using 16 states that have different amplitudes and phases. Similarly, 64-QAM sends bits six at a time using 64 different states, so it has a data rate six times greater than that of BPSK.

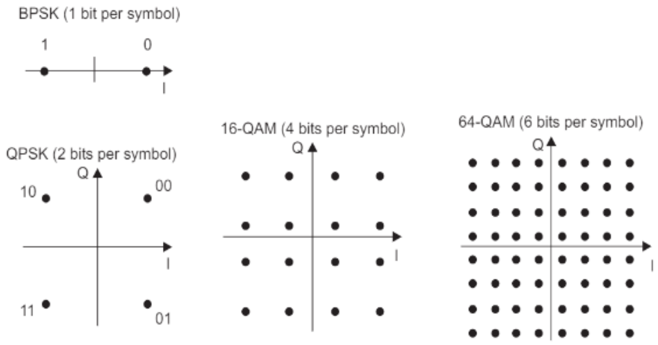


Fig. 6A.3 Modulation schemes used by LTE

6A.1.4 Multiple Access Techniques

The techniques described so far work well for one-to-one communications. In a cellular network, however, a base station has to transmit to many different mobiles at once. It does this by sharing the resources of the air interface, in a technique known as *multiple access*

Mobile communication systems use a few different multiple access techniques, two of which are shown in Fig. 6A.6. *Frequency division multiple access* (POMA) was used y the first-generation analogue systems. In this technique, each mobile receives on its own carrier frequency, which it distinguishes from the others y the use of analogue filters. The carriers are separated by unused *guard bands*, which minimizes the interference between them. In *time division multiple access* (TOMA), mobiles receive information on the same carrier frequency but at different times.

GSM uses a mix of frequency and time division multiple access, in which every cell has several carrier frequencies that are each shared amongst eight different mobiles. LTE uses another mixed technique known as *orthogonal frequency division multiple access* (OFDMA).

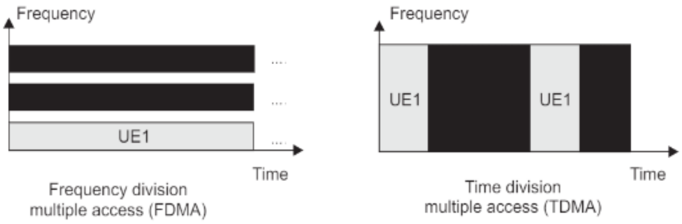


Fig. 6A.6 Example multiple access techniques

Third generation communication systems used a different technique altogether, known as *code division multiple access* (CDMA). In this technique, mobiles receive on the same carrier frequency and at the same time, but the signals are labelled by the use of codes, which allow a mobile to separate its own signal from those of the others. LTE uses a few of the concepts from CDMA for some of its control signals, but does not implement the technique otherwise

Multiple access is actually a generalization of a simpler technique known as *multiplexing*. The difference between the two is that a multiple access system can dynamically change the allocation of resources to different mobiles, while in a multiplexing system the resource allocation is fixed.

6A.1.5 FDD and TDD Modes

By using the multiple access techniques described above, a base station can distinguish the transmissions to and from the individual mobiles in the cell. However, we still need a way to distinguish the mobiles' transmissions from those of the base stations themselves.

To do this, a mobile communication system can operate in the transmission modes that we introduced in Fig. 6A.7. When using frequency division duplex (FOO), the base station and mobile transmit and receive at the same time, but using different carrier frequencies. Using time division duplex (TDD), they transmit and receive on the same carrier frequency but at different times.

FOO and TDD modes have different advantages and disadvantages. In FOO mode, the bandwidths of the uplink and downlink are fixed and are usually the same. This makes it suitable for voice communications, in which the uplink and downlink data rates are very similar. In TDD mode, the system can adjust how much time is allocated to the uplink and downlink. This makes it suitable for applications such as web browsing, in which the downlink data rate can be much greater than the rate on the uplink. TDD mode can be badly affected by interference if, for example, one base station is transmitting while a nearby base station is receiving. To avoid this, nearby base stations must be carefully time synchronized and must use the same allocations for the uplink and downlink, so that they all transmit and receive at the same time. This makes TDD suitable for networks that are made from isolated hotspots, because each hotspot can have a different timing and resource allocation. In contrast, FDD is often preferred for wide-area networks that have no isolated regions.

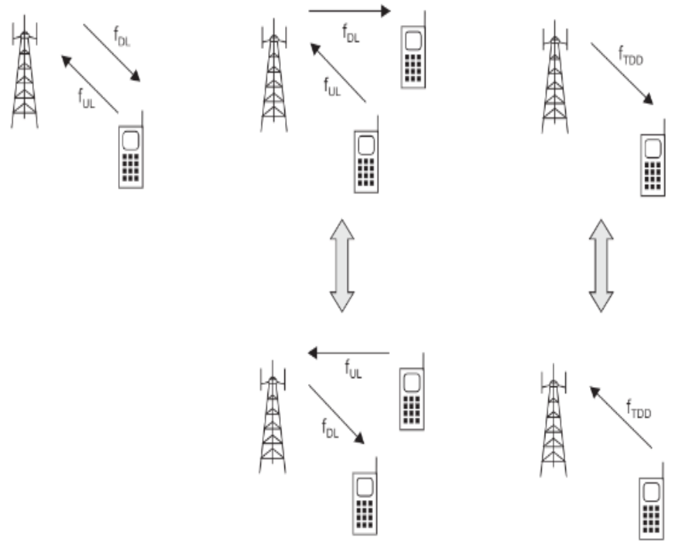


Fig. 6A.7 Operation of FDD and TDD modes

When operating in FDD mode, the mobile usually has to contain a high attenuation duplex filter that isolates the uplink transmitter from the downlink receiver. In a variation known as *half duplex FOO mode*, a base station can still transmit and receive at the same time, but a mobile can only do one or the other. This means that the mobile does not have to isolate the transmitter and receiver to the same extent, which eases the design of its radio hardware.

LTE supports each of the modes described above. A cell can use either FDD or TDD mode. A mobile can support any combination of full duplex FDD, half duplex FDD and TDD, although it will only use one of these at a time.

6A.2.1 Multipath and Fading

Propagation loss and noise are not the only problem. As a result of reflections, rays can take several different paths from the transmitter to the receiver. This phenomenon is known as *multipath*. At the receiver, the incoming rays can add together in different ways, which are shown in Fig. 6A.8. If the peaks of the incoming rays coincide then they reinforce each other, a situation known as *constructive interference*. If, however, the peaks of one ray coincide with the troughs of another, then the result is *destructive interference*, in which the rays cancel. Destructive interference can make the received signal power drop to a very low level, a situation known as *fading*. The resulting increase in the error rate makes fading a serious problem for any mobile communication system.

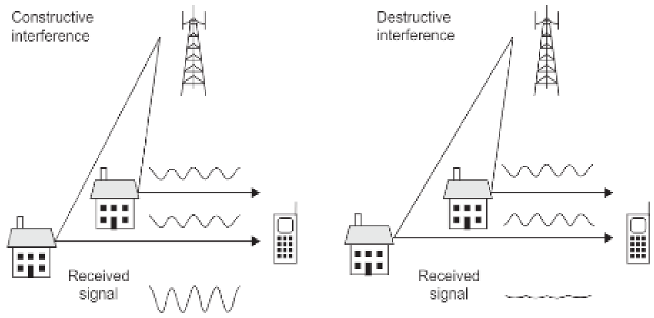


Fig. 6A.8 Generation of constructive interference, destructive interference and fading in a multipath environment

If the mobile moves from one place to another, then the ray geometry changes, so the interference pattern changes between constructive and destructive. Fading is therefore a function of time, as shown in Fig. *6A.9a*. The amplitude and phase of the received signal vary over a timescale called the *coherence time*, *T*, which can be estimated as follows:

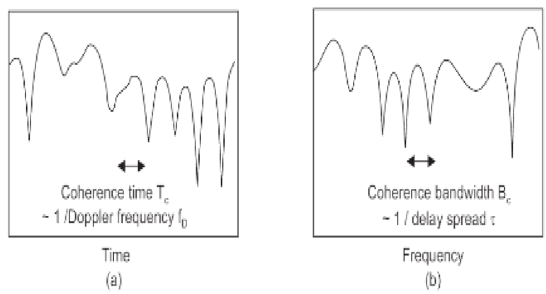


Fig. 6A.9 Fading as a function of *(a)* time and *(b)* frequency

$$T_c \approx \frac{1}{f_D} \qquad ...(3)$$

Here f_D is the mobile's Doppler frequency:

$$f_D = \frac{v}{c} f_C \qquad ...(4)$$

where fc is the carrier frequency, v is the speed of the mobile and c is the speed of light (3×10 ms^{-1}). For example, a pedestrian might walk with a speed of 1 ms^{-1} (3.6 km hr^{-1}). At a carrier frequency of 1500 MHz, the resulting Doppler shift is 5 Hz, giving a coherence time of about 200 milliseconds. Faster mobiles move through the interference pattern more quickly, so their coherence time is correspondingly less.

If the carrier frequency changes, then the wavelength of the radio signal changes. This also makes the interference pattern change between constructive and destructive, so fading is a function of frequency as well (Fig. 6A.9b). The amplitude and phase of the received signal vary over a frequency scale called the *coherence bandwidth, B,* which can be estimated as follows:

$$B_C \approx \frac{1}{\tau} \qquad ...(5)$$

Here, τ is the *delay spread* of the radio channel, which is the difference between the arrival times of the earliest and latest rays. It can be calculated as follows:

$$\tau = \frac{\Delta L}{c} \qquad ...(6)$$

where ΔL is the difference between the path lengths of the longest and shortest rays. In a macrocell, a typical path difference might be around 300 metres, giving a delay spread of 1 µs and a coherence bandwidth of around 1 MHz Smaller cells have a smaller delay spread, so have a larger coherence bandwidth.

6A.3 Error Management

6A.3.1 Forward Error Correction

We have read that noise and interference lead to errors in a wireless communication receiver. These are bad enough during voice calls, but are even more damaging to important information such as web pages and emails. Fortunately, there are several ways to solve the problem.

The most important technique is *forward error correction.* In this technique, the transmitted information is represented using a *codeword that* is typically two or three times as long. The extra bits supply additional, redundant data that allow the receiver to recover the original information sequence. For example, a transmitter might represent the information sequence 101 using the codeword 110010111. After an error in the second bit, the receiver might recover the codeword 100010111. If the coding scheme has been well designed, then the receiver can conclude that this is not a valid codeword, and that the most likely transmitted codeword was 110010111. The receiver has therefore corrected the bit error and can recover the original information. The effect is very like written English, which contains redundant letters that allow the reader to understand the underlying information, even in the presence of spelling mistakes.

The *coding rate* is the number of information bits divided by the number of transmitted bits (1/3 in the example above). Usually, forward error correction algorithms operate with a fixed coding rate. Despite this, a wireless transmitter can still adjust the coding rate using the two-stage process shown in Fig. 6A.11. In the first stage, the information bits are passed through a fixed-rate coder. The main algorithm used by LTE is known as *turbo coding* and has a fixed coding rate of 1/3. In the second stage, called *rate matching,* some of the coded bits are selected for transmission, while the others are discarded in a process known as *puncturing.* The receiver has a copy of the puncturing algorithm, so it can insert dummy bits at the points where information was discarded. It can then pass the result through a turbo decoder for error correction.

Changes in the coding rate have a similar effect to changes in the modulation scheme. If the coding rate is low, then the transmitted data contain many redundant bits. This allows the receiver to correct a large number of errors and to operate successfully at a low SINR, but at the expense of a low information rate. If the coding rate is close to 1, then the information rate is higher but the system is more vulnerable to errors.

LTE exploits this with a similar trade-off to the one we saw earlier, by transmitting with a high coding rate if the received SINR is high and vice versa.

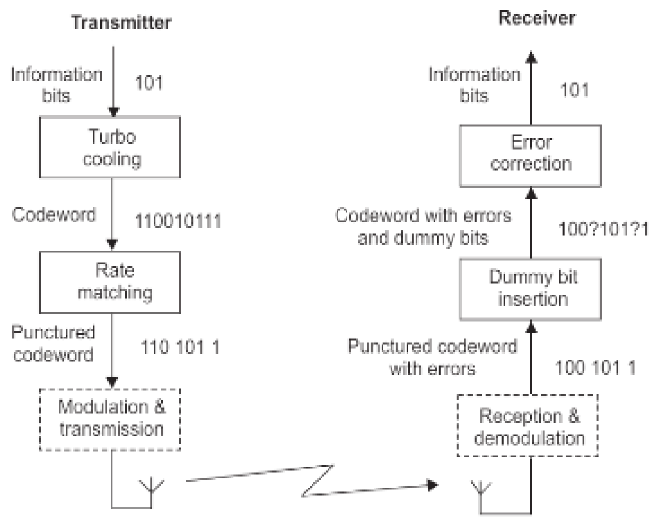


Fig. 6A.11 Block diagram of a transmitter and receiver using forward error correction and rate matching

6A.3.2 Automatic Repeat Request

Automatic repeat request (ARQ) is another error management technique, which is illustrated in Fig. 6A.12. Here, the transmitter takes a block of information bits and uses them to compute some extra bits that are known as a *cyclic redundancy check* (CRC).

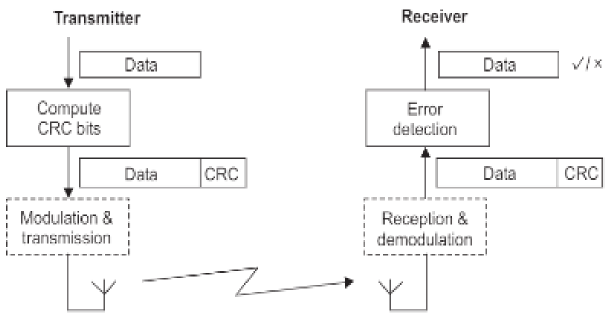


Fig. 6A.12 Block diagram of a transmitter and receiver using automatic repeat request

It appends these to the information block and then transmits the two sets of data in the usual way. The receiver separates the two fields and uses the information bits to compute the expected CRC bits. If the observed and expected CRC bits are the same, then it concludes that the information has been received correctly and sends a positive acknowledgement back to the transmitter. If the CRC bits are different, it concludes that an error has occurred and sends a negative acknowledgement to request a re-transmission. Positive and negative

acknowledgements are often abbreviated to ACK and NACK respectively

A wireless communication system often combines the two error management techniques that we have been describing. Such a system corrects most of the bit errors by the use of forward error correction and then uses automatic repeat requests to handle the remaining errors that leak through.

Normally, ARQ uses a technique called selective *re-transmission* (Fig. 6A.13), in which the receiver waits for several blocks of data to arrive before acknowledging them all. This allows the transmitter to continue sending data without waiting for an acknowledgement, but it means that any re-transmitted data can take a long time to arrive. Consequently, this technique is only suitable for non-real-time streams such as web pages and emails.

Chapter-Antenna

13.1 INTRODUCTION

An ***antenna*** is a device for converting electromagnetic radiation in space into electrical currents in conductors or vice-versa, depending on whether it is being used for receiving or for transmitting, respectively. Antennas transform wire propagated waves into space propagated waves. They receive electromagnetic waves and pass them onto a receiver or they transmit electromagnetic waves which have been produced by a transmitter. As a matter of principle all the features of passive antennas can be applied for reception and transmission alike (reciprocity). From a connection point of view the antenna appears to be a dual gate, although in reality it is a quad gate. The connection which is not made to a RF-cable is connected to the environment, therefore one must always note, that the surroundings of the antenna have a strong influence on the antenna's electrical features.

Passive radio telescopes are receiving antennas. It is usually easier to calculate the properties of transmitting antennas. Fortunately, most characteristics of a transmitting antenna (*e.g.,* its radiation pattern) are unchanged when the antenna is used for receiving, so we often use the analysis of a transmitting antenna to understand a receiving antenna used in radio astronomy.

The requirements on the antennas needed for the ever-expanding networks are becoming continually higher. An antenna must have the following features:

- Strictly defined radiation patterns for a most accurate network planning.
- Growing concern for the level of intermodulation due to the radiation of many HF- carriers via one antenna.
- Dual polarization.
- Electrical down-tilting of the vertical diagram.
- Unobtrusive design.

13.3 Common Antennas

In this section, some common antennas are described along with details about typical patterns that can be expected from these common antennas. Described here are a dipole, a collinear array, a single patch antenna, a patch array, a Yagi and even a sector antenna. The patterns from each antenna are shown and explained in detail, including a 3D radiation pattern.

13.3.1 Omnidirectional Antenna

An *omnidirectional antenna* is an antenna that has a non-directional pattern (circular pattern) in a given plane with a directional pattern in any orthogonal plane. Examples of omnidirectional antennas are dipoles and collinear antennas.

13.3.2 Dipole Antennas

A dipole antenna most commonly refers to a half-wavelength ($\lambda/2$) dipole. The physical antenna is constructed of conductive elements whose combined length is about half of a wavelength at its intended frequency of operation. This is a simple antenna that radiates its energy out toward the horizon

Fig: Dipole antenna model

The patterns shown in Fig. 13.4 are those resulting from a perfect dipole formed with two thin wires oriented vertically along the z-axis.

13.3.3 Collinear Omni Antennas

In order to create an omnidirectional antenna with higher gain, multiple omnidirectional structures (either wires or elements on a circuit board) can be arranged in a vertical, linear fashion to retain the same omnidirectional pattern in the azimuth plane but a more focused elevation plane beam which then has

higher gain. This is frequently referred to as a *collinear array*. Note that the higher gain doesn't imply that the antenna creates more power. It means that the same amount of power is radiated in a more focused way.

13.3.4 Directional Antennas

A *directional antenna* is one that radiates its energy more effectively in one (or some) direction than others. Typically, these antennas have one main lobe and several minor lobes. Examples of directional antennas are patches and dishes.

Directional antennas are used for coverage as well as point-to-point links. They can be patch antennas, dishes, horns or a whole host of other varieties. They all accomplish the same goal: radiating their energy out in a particular direction.

13.3.5 Patch Antennas

A patch antenna, in its simplest form, is just a single rectangular (or circular) conductive plate that is spaced above a ground plane. Patch antennas are attractive due to their low profile and ease of fabrication.

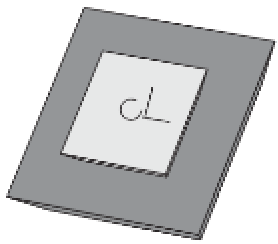


Fig: Patch Antennas model

The radiation pattern of a single patch is characterized by a single main lobe of moderate beamwidth. Frequently, the beamwidths in the azimuth and elevation planes are similar, resulting in a fairly circular beam, although this is by no means universal. The beamwidths can be manipulated to produce an antenna with higher or lower gain, depending on the requirements. An antenna built with a single patch will have a maximum gain of about 9 dBi or a bit less.

The patch antenna in Fig. 13.6 shows how simple these antennas can be. This is a simple rectangular patch built over a rectangular ground plane. The radiation patterns exhibit typical patch antenna characteristics. There is a single main lobe with a fairly wide beamwidth with shallow nulls pointing up and down from the antenna. Other than that, there aren't many features to the pattern. The one shown in Fig. 13.6 is designed to have higher gain rather than symmetrical plane patterns.

13.3.6 Patch Array Antennas

A patch array antenna is, in general, some arrangement of multiple patch antennas that are all driven by the same source.

Frequently, this arrangement consists of patches arranged in orderly rows and columns (a rectangular array) as shown in Fig. 13.7. The reason for these types of arrangements is higher gain. Higher gain commonly implies a narrower beamwidth and that is, indeed, the case with patch arrays. The array shown here has a gain of about 18 dBi with an azimuth and elevation plane beamwidth of about 20 degrees. Notice that the back lobes are very small and that the front-to-back ratio is about 30 db. The first sidelobes are down from the peak about 14 dB.

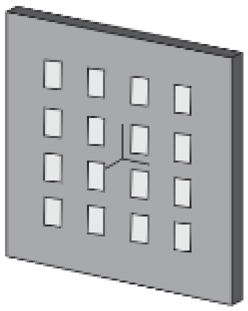


Fig: 4 x 4 patch array antenna model

13.3.7 Yagi Antennas

A Yagi antenna is formed by driving a simple antenna, typically a dipole or dipole-like antenna, and shaping the beam using a well-chosen series of non-driven elements whose length and spacing are tightly controlled. The Yagi shown here in Fig. 13.7 is built with one reflector (the bar behind the driven antenna) and 14 directors (the bars in front of the driven antenna). This configuration yields a gain of about 15 dBi with azimuth and elevation plane beamwidths that are basically the same, around 36 degrees. That is a common feature of Yagi antennas. Many times, these antennas are designed so that they can be rotated for either horizontal or vertical polarization, so having the same 3-dB beamwidth in each plane is a nice feature in those instances.

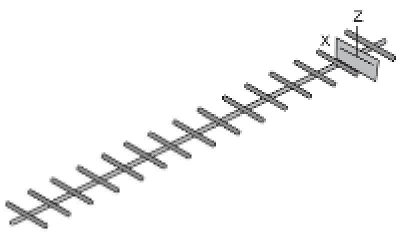


Fig: Yagi antenna model