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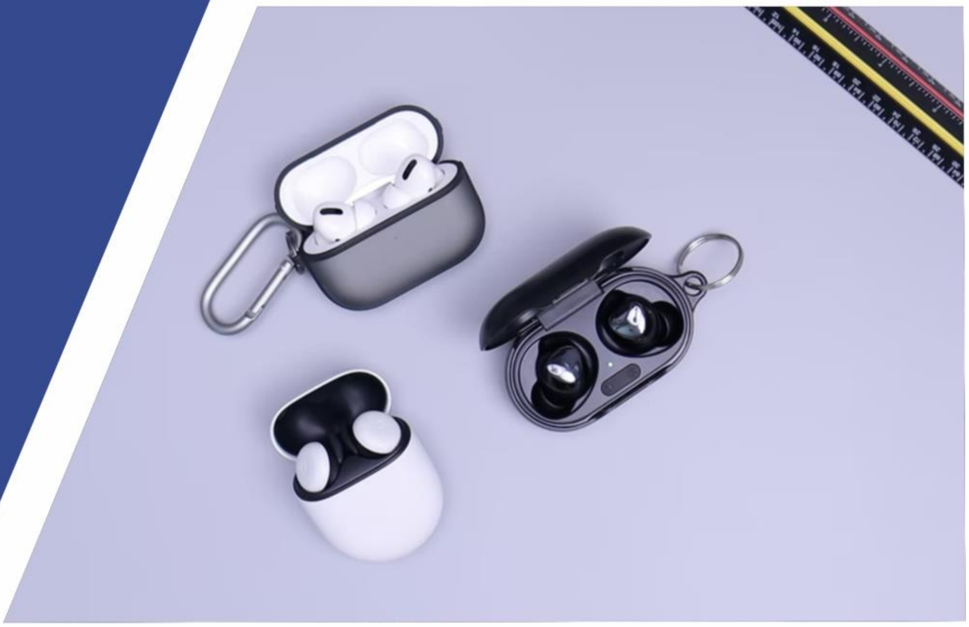

Wireless Communication I

Module 4

CIT 855 Wireless Communications I Module 4

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Unit 1 Introduction to Modulation

1.0 Introduction

The simple transmission scheme outlined in the previous units cannot be used for commercial broadcasting. If a dozen of stations all transmitted sounds by the mechanism described in the previous units, a receiving station would pick up a garbled combination of all transmissions. To prevent interference from a number of transmitting stations, all broadcast radio waves are first modulated.

2.0 Objectives

At the end of this unit, you should be able to:

- define a modulation
- state the reason for modulation
- explain how a sound wave is modulated
- list clearly the four types of modulation techniques.

3.0 Main Content

3.1 Definition of Modulation

Modulation is the process of encoding information from a message source into manner suitable for transmission. It generally involves translating a baseband message signal (called the source) to a **bandpass** signal at frequencies that are very high when compared to the baseband frequency. The **bandpass** signal is called the modulated signal and the **baseband** signal is called the modulating signal. Modulation may be done by varying the amplitude, phase or frequency of a high frequency carrier in accordance with the amplitude of the message signal.

Modulation may be defined as a process by which some characteristic of a signal known as carrier is varied according to the instantaneous value of another signal known as modulating signal. The signals containing intelligence or information to be transmitted are called modulating signals. These modulating signals containing information are also called baseband signals. Also the carrier frequency is greater than the modulating frequencies and the signal which results from the process of modulation is known as modulated signal.

Demodulation is the process of extracting the baseband message from the carrier so that it may be processed and interpreted by the intended receiver (also called sink). It is the process of extracting a modulating or baseband signal from the modulated signal. In other word, demodulation is the process by which the message is recovered from the modulated signal at the receiver.

A device that performs modulation is known as a modulator and a device that performs the inverse operation of modulation is known as a demodulator (sometimes detector *or* demod*). A* device that can do both operations is a modem (short for "Modulator-Demodulator").

3.1.1Why Modulate?

- For allowing multiple signals to share a single physical channel by Frequency Division Multiplexing.
- Necessary for wireless communication where the antenna diameter must be at least equal to the wavelength of the carrier signal.

This means, for a 3000 Hz signal through space, the antenna diameter must be at least 60 miles.

3.1.2 Modulation Choices

- The **simplest modulation** method is also the first used to transmit messages. The signal is turned on and off to transmit the characters of an agreed code. Text messages can be carried by the signal modulated in this way. Unique patterns stand for letters of the alphabet, numerals, and punctuation marks.
- Amplitude modulation is the least complicated modulation method capable of transmitting speech or music by varying the carrier signal's instantaneous power. It means that the amplitude (or size) of the wave of the original sound wave has been changed by adding it to the carrier wave.
- Sound waves can be modulated in such a way that their frequency is altered. For example, a sound wave can be added to a carrier signal to produce a signal with the same amplitude, but a different frequency. The sound wave has, in this case, undergone **frequency modulation (FM).**
- Sound can be converted to digital data, transmitted, then used to reconstruct the original waveform in the receiver. This sound wave is a form of **digital modulation**.

3.1.3Modulating a Sound Wave

After the sound wave has been modulated at the transmitting station, both AM and FM signals must be decoded at the receiving station. In either case, the carrier wave is electronically subtracted from the radio wave that is picked up by the receiving antenna. What remains after this process is the original sound wave, encoded, of course, as an electrical signal.

All broadcasting stations are assigned characteristic carrier frequencies by the Federal Communications Commission (FCC). This system allows a number of stations to operate in the same area without overlapping.

Thus, two stations a few kilometers apart could both be sending out exactly the same program, but they would sound different (and have different electric signals) because each had been overlaid on a different carrier signal.

Receiving stations can detect the difference between these two transmissions because they can tune their equipment to pick up only one or the other carrier frequency. When you turn the tuning knob on your own radio, for example, you are adjusting the receiver to pick up carrier waves from station A, station B, or some other station. Your radio then decodes the signal it has received by subtracting the carrier wave and converting the remaining electric signal to a sound wave.

The identifying characteristics by which you recognise a radio station reflect its two important transmitting features. The frequency, such as 101.5 megahertz (or simply "101.5 on your dial") identifies the carrier wave frequency, as described above. The power rating ("operating with 50,000 watts of power") describes the power available to transmit its signal. The higher the power of the station, the greater the distance at which its signal can be picked up.

3.1.4Types of Modulation Techniques

- **Analog modulation:** The aim of analog modulation is to transfer an analog baseband (or lowpass) signal, for example an audio signal or TV signal, over an analog passband channel, for example a limited radio frequency band or a cable TV network channel.
- **BandPass digital modulation:** The aim of digital modulation is to transfer a digital bit stream over an analog passband channel, for example over the public switched telephone network (where a bandpass filter limits the frequency range to between 300 and 3400 Hz), or over a limited radio frequency band.
- **Digital baseband modulation or line coding modulation:** The aim of digital baseband modulation methods, also known as line coding, is to transfer a digital bit stream over a baseband channel, typically a non-filtered copper wire such as a serial bus or a wired local area network.
- **Pulse shaping modulation:** The aim of pulse modulation methods is to transfer a narrowband analog signal, for example a phone call over a wideband baseband channel or, in some of the schemes, as a bit stream over another digital transmission system

Self-Assessment Exercise

What do you understand by the term modulation?

4.0 Conclusion

In this unit, the term "modulation", the reason for modulation, the choices of modulation, how a sound wave is being modulated and the four modulation techniques were discussed.

5.0 Summary

In this unit you have learnt that:

- modulation is a fundamental requirement of a communication system
- modulation may be done by varying the amplitude, phase or frequency of a high frequency carrier in accordance with the amplitude of the message signal
- the four types of modulation techniques are analog modulation, digital bandpass modulation, digital baseband modulation and pulse shaping modulation.

6.0 Tutor-Marked Assignment

i. Briefly describe how sound wave can be modulated?
ii. Mention four types of modulation techniques.
iii. State two major reasons for modulation.

7.0 References/Further Reading

Bloomfield, L. A. (2000). *How Things Work: The Physics of Everyday Life.* (Second Edition). New York: John Wiley & Sons.

Davidovits, P. (1972). *Communication.* New York: Holt Rinehart and Winston, Inc.

Sharma, S. (2006). *Wireless & Cellular Communications*. New Delhi: S. K. Kataria & Sons

Unit 2 Analog Modulation Techniques

1.0 Introduction

In analog modulation, the carrier waveform is continuous in nature. The two families of continuous wave modulation systems are amplitude modulation and angle modulation.

2.0 Objectives

At the end of this unit, you should be able to:

- differentiate between the two types of analog modulation
- explain the meaning of angle modulation
- state the types of angle modulation
- mention the merit and demerit of FM
- compute the modulation index, frequency, deviation, carrier frequency and power of a signal.

3.0 Main Content

3.1 Concept of Analog Modulation Techniques

In analog modulation, the modulation is applied continuously in response to the analog information signal. The aim of **analog modulation** is to transfer an analog baseband (or lowpass) signal, for example an audio signal or TV signal, over an analog passband channel, for example a limited radio frequency band or a cable TV network channel.

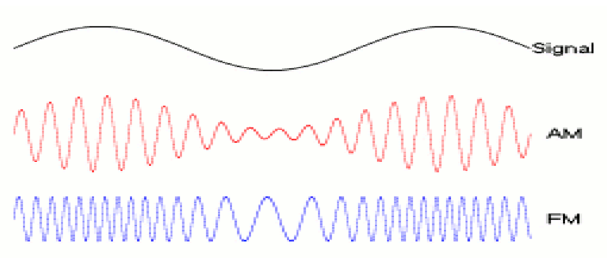

Fig. 2.1: Low-Frequency Message Signal (top) may be Carried by an AM or FM Radio Wave

Common analog modulation techniques are:

- Amplitude Modulation and
- Angle Modulation

3.1.1 Amplitude Modulation (AM)

AM may be defined as a system in which the maximum amplitude of the carrier wave is made proportional to the instantaneous value (amplitude) of the modulating or baseband signal. **AM** is a technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. AM works by varying the strength of the transmitted signal in relation to the information being sent. For example, changes in the signal strength can be used to reflect the sounds to be reproduced by a speaker, or to specify the light intensity of television pixels.

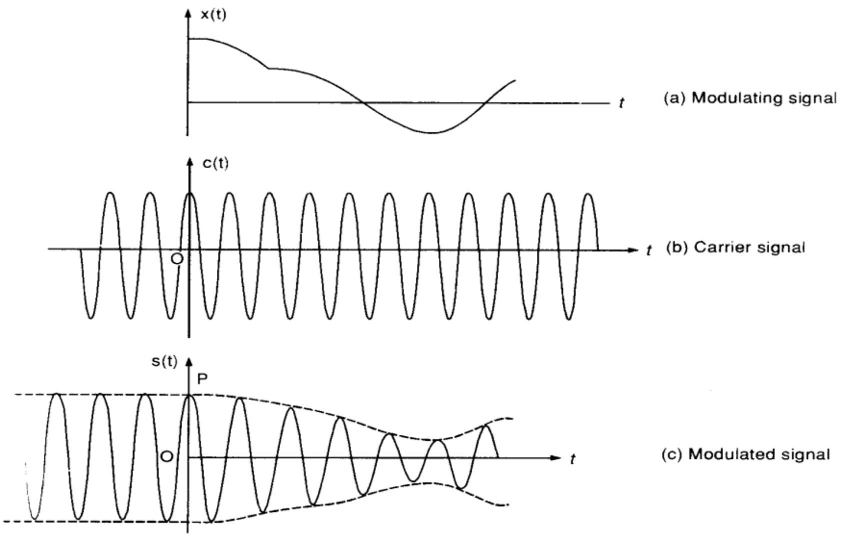

Fig. 2.2:Illustration of Amplitude Modulation

***Source:* Sharma S. (Wireless & Cellular Communication)**

Conventional AM

This is how a typical AM system transmitter works:

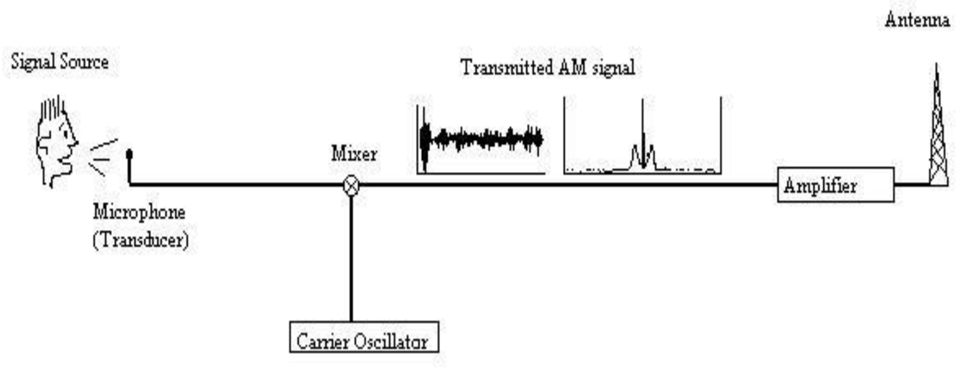

Fig.2.3: Conventional Amplitude Modulation

The information signal is mixed with the carrier signal and produces the full AM signal to be transmitted.

Transmission Efficiency of Amplitude Modulated Signal

The total modulated power of an AM signal is expressed as

$$P_t = P_c + P_s = \frac{1}{2}\left[A^2 + \overline{x^2(t)}\right]$$

Out of this total power P_t, the useful message or baseband power is the power carried by the sidebands, i.e. P_s. The large carrier power P_c is a waste from the transmission point of view because it does not carry any information or message. This large carrier power P_c is transmitted along with the sideband power only for the convenient and cheap detection.

Hence, Ps is the only useful message power present in the AM wave.

In AM wave, the amount of useful message power P_s may be expressed by a term known as **transmission efficiency** η.

Hence, transmission efficiency of AM wave may be defined as the percentage of total power contributed by the sidebands.

Mathematically,

Transmission Efficiency, $$\eta = \frac{P_s}{P_t} \times 100$$

Or

$$\eta = \frac{\frac{1}{2}\overline{x^2(t)}}{\frac{1}{2}\left[A^2 + \overline{x^2(t)}\right]} \times 100 \quad = \frac{100\overline{x^2(t)}}{A^2 + \overline{x^2(t)}}$$

The maximum transmission efficiency of the AM is only 33.33%. This implies that only one-third of the total power is carried by the sidebands and the rest two-third is wasted.

3.1.1.1 Types of Amplitude Modulation Techniques

(a) **Single-Sideband Modulation** (SSB, or SSB-AM): is a refinement of amplitude modulation that uses electrical power and bandwidth. It is closely related to vestigial sideband modulation (VSB). Amplitude modulation produces a modulated output signal that has twice the bandwidth of the original baseband signal. Single-sideband modulation avoids this bandwidth doubling, and the power wasted on a carrier, at the cost of somewhat increased device complexity. The SSB Schemes are:

- SSB with carrier (SSB-WC)
- SSB suppressed carrier modulation (SSB-SC): is a modulation which provides a single sideband with suppressed carrier. In SSB-SC the carrier power level is suppressed to the point where it is insufficient to demodulate the signal. The information represented by the modulating signal is contained in both the upper and the lower sidebands. Since each modulating frequency f_c produces corresponding upper and lower side-frequencies

$f_c + f_i$ and $f_c - f_i$

it is not necessary to transmit both side-bands. Either one can be suppressed at the transmitter without any loss of information.

Advantages of SSB-WC

- Less transmitter power.

- Less bandwidth, one-half that of Double-Sideband (DSB).
- Less noise at the receiver.
- Size, weight and peak antenna voltage of a single-sideband (SSB) transmitters is significantly less than that of a standard AM transmitter

The single side-band (SSB) is very simple: if you do not need two side-bands, you can get rid of one of the band. To do this, you add a band pass filter component to your system that removes the extra side-band.

This is how the SSB transmitter looks like:

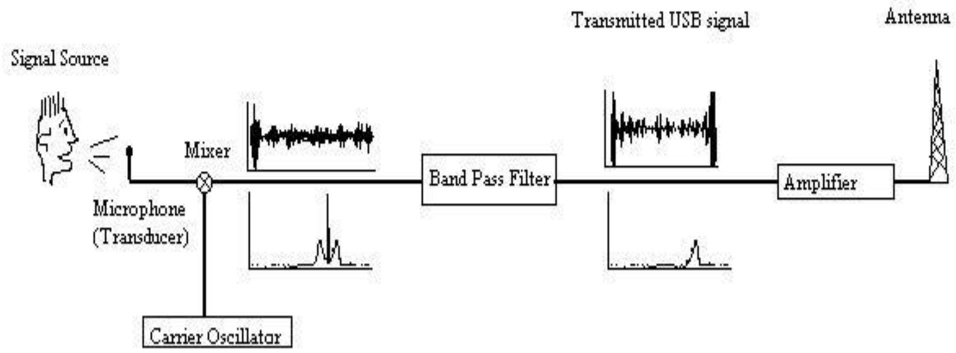

Fig.2.4: SSB Transmitter

Note that the band pass filter has removed the lower side-band (LSB) and the carrier from the spectrum. The remainder is transmitted.

The receiver cannot output the signal as it is, it must first restore the signal to what it should be before demodulation. The receiver in a SSB system has its own carrier signal (from a local oscillator) that is puts back in. The receiver looks like the diagram in figure 2.5 below:

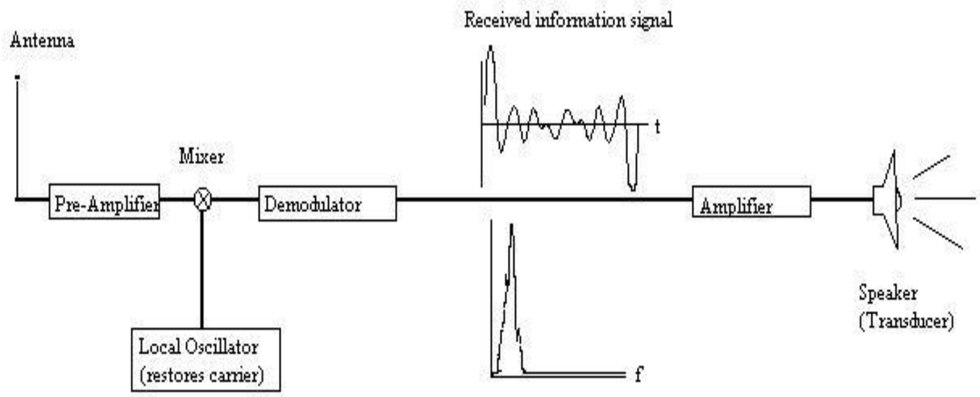

Fig.2.5: SSB Receiver

By having its own carrier signal, the receiver makes the signal back into what would be sent by a conventional AM system. The remaining signal is processed normally.

So, there are two modifications to make it work.

- The transmitter adds a band pass filter before amplification for transmission and
- The receiver adds a local carrier signal back into the signal prior to processing.

Power of SSB-AM Signal

The Total Power of a SSB-AM can be expressed as $P_t = P_c\left(1+\frac{m_a^2}{2}\right)$

Self-Assessment Exercise

A 400 watts carrier is modulated to a depth of 75 percent. Find the total power in the amplitude-modulated wave. Assume the modulating signal to be a sinusoidal one.

Solution: We know that for a sinusoidal modulating signal, the total power is expressed as

$$P_t = P_c\left(1+\frac{m_a^2}{2}\right)$$

where

Pt = total power or modulated power

Pc = carrier power or unmodulated power

ma = modulation index

Given that, P_c = 400watts m_a = 75 percent = 0.75

Therefore, $P_t = P_c\left(1+\frac{m_a^2}{2}\right)$ = $400\left(1+\frac{0.75^2}{2}\right) = 512.5 watts$

Self-Assessment Exercise

An AM broadcast radio transfer radiates 10K watts of power, if modulation percentage is 60. Calculate how much of this the carrier power is.

Solution: We known that the total power is expressed as

$$P_t = P_c\left(1+\frac{m_a^2}{2}\right)$$

where P_t = total power or modulated power

Pc = carrier power or unmodulated power

ma = modulation index

Given that, P_t = 10 K watts m_a =60 percent = 0.6

Therefore $$P_c = \frac{P_t}{1+\frac{m_a^2}{2}} = \frac{10}{1+\frac{0.6^2}{2}}$$

or $$P_c = \frac{10}{1.18} = 8.47\ KW$$

Current Calculation for SSB AM

In AM, it is generally more convenient to measure the AM transmitter current than the power. In this case, the modulation index may be calculated from the values of unmodulated and modulated currents in the AM transmitter. Let I_c be the r.m.s value of the carrier or unmodulated current and I_t be the r.m.s value of the total or modulated current of an AM transmitter. Let R be the antenna resistance through which these currents flow.

We know that for an SSB AM the power relation is expressed as

$$P_t = P_c\left(1+\frac{m_a^2}{2}\right)$$

where

Pt = total power or modulated power

Pc = carrier power or unmodulated power

ma = modulation index

From the above equation, we may write

$$\frac{P_t}{P_c} = 1+\frac{m_a^2}{2}$$

or $$\frac{I_t^2 . R}{I_c^2 . R} = 1+\frac{m_a^2}{2}$$

or $$\frac{I_t}{I_c} = \sqrt{1+\frac{m_a^2}{2}}$$

or $$I_t = I_c\sqrt{1+\frac{m_a^2}{2}}$$

Self-Assessment Exercise

The antenna current of an AM transmitter is 8 A if only the carrier is sent, but it increases to 8.93 A if the carrier is modulated by a single sinusoidal wave. Determine the percentage modulation. Also find the antenna current if the percent of modulation changes to 0.8.

Solution: (i) The current relation for an SSB-AM is expressed as

$$I_t = I_c\sqrt{1+\frac{m_a^2}{2}}$$

where

I_t = total or modulated current

I_c = carrier or unmodulated current

m_a = modulation index

Using the above equation, we have

$$\frac{I_t}{I_c} = \sqrt{1+\frac{m_a^2}{2}}$$

or $$\left(\frac{I_t}{I_c}\right)^2 = 1+\frac{m_a^2}{2}$$

or $$\frac{m_a^2}{2} = \left(\frac{I_t}{I_c}\right)^2 - 1$$

or $$m_a^2 = 2\left[\left(\frac{I_t}{I_c}\right)^2 - 1\right]$$

or $$m_a = \sqrt{2\left[\left(\frac{I_t}{I_c}\right)^2 - 1\right]}$$

Putting all the given values, we have

$$m_a = \sqrt{2\left[\left(\frac{8.93}{8}\right)^2 - 1\right]} = \sqrt{2\left[(1.116)^2 - 1\right]}$$

$$m_a = \sqrt{2(1.246-1)} = \sqrt{0.492} = 0.701 = 70.1\%$$

(ii) Since $I_t = I_c \sqrt{1 + \frac{m_a^2}{2}}$

Here, I_c = 8A and m_a = 0.8

Therefore, $I_t = 8 \times \sqrt{1 + \frac{0.8^2}{2}} = 8\sqrt{1 + \frac{0.64}{2}}$

or $I_t = 8\sqrt{1.32} = 8 \times 1.149 = 9.19A$

(b)Double-Sideband Modulation (DSB)

- Double-sideband modulation with unsuppressed carrier (DSB-WC) is used on the AM radio broadcasting band.
- Double-sideband suppressed-carrier transmission (DSB-SC): is a modulated signal, which contains no carrier but two sidebands. It is a transmission in which (a) frequencies produced by amplitude modulation are symmetrically spaced above and below the carrier frequency and (b) the carrier level is reduced to the lowest practical level, ideally completely suppressed. In the double-sideband suppressed-carrier transmission (DSB-SC) modulation, unlike AM, the wave carrier is not transmitted; thus, a great percentage of power that is dedicated to it is distributed between the sidebands, which imply an increase of the cover in DSB-SC, compared to AM, for the same power used. DSB-SC transmission is a special case of Double-sideband reduced carrier transmission. This is used for RDS (Radio Data System) because it is difficult to decouple.
- Double-sideband reduced carrier transmission (DSB-RC): transmission in which (a) the frequencies produced by amplitude modulation are symmetrically spaced above and below the carrier and (b) the carrier level is reduced for transmission at a fixed level below that which is provided to the modulator. In DSB-RC transmission, the carrier is usually transmitted at a level suitable for use as a reference by the receiver, except for the case in which it is reduced to the minimum practical level, i.e. the carrier is suppressed.

(c) **Vestigial Sideband Modulation** (VSB, or VSB-AM) is a sideband that has been only partly cut off or suppressed. In VSB modulation instead of rejecting one sideband completely as in SSB modulation scheme, a gradual cut-off of one sideband is allowed. This gradual cut is compensated by a vestige or portion of the other sideband. Television broadcasts (in analog video formats) use this method if the video is transmitted in AM, due to the large bandwidth used. It may also be used in digital transmission, such as the ATSC standardised 8-VSB.

(d) **Quadrature Amplitude Modulation** (QAM) is also called Quadrature Carrier Multiplexing. This modulation scheme enables two DSC-SC modulated signals to occupy the same transmission bandwidth and therefore it allows for the separation of the two message signals at the receiver output. It is, therefore, known as a bandwidth-conservation scheme.

Problems with Conventional AM

Conventional AM transmission has several problems:

- bandwidth is wasted by having two identical side-bands on either side of the carrier
- the efficiency is limited to 33% to prevent distortion in the receiver when demodulating

- the carrier signal is present even if nothing is being transmitted.

3.1.2 Angle Modulation

Angle modulation is a class of analog modulation. Angle modulation may be defined as the process in which the total phase angle of a carrier wave is varied in accordance with the instantaneous value of the modulating or message signal while keeping the amplitude of the carrier constant. These techniques are based on altering the angle (or *phase*) of a sinusoidal carrier wave to transmit data, as opposed to varying the amplitude, such as in AM transmission. The two main types of angle modulation are: (i) Frequency modulation (FM) and (ii)Phase modulation (PM), with its digital correspondence phase-shift keying (PSK).

3.1.2.1 Types of Angle Modulation

3.1.2.1.1 Frequency Modulation (FM)

Frequency modulation conveys information over a carrier wave by varying its frequency. Here the frequency of the modulated signal is varied. The two types of frequency modulation are Single Tone Frequency Modulation and Multiple Frequency Modulation.

(a) **Single Tone Frequency Modulation** is a type of frequency modulation (FM) in which the modulating or baseband signal contains a single frequency.

- The total variation in frequency from the lowest to the highest point is called carrier swing. Obviously,

$$\text{the Carrier Swing} = 2 \text{ x } \text{ frequency deviation } = 2 \text{ x } \Delta_{\omega}$$

- For FM, the modulation index is defined as the ratio of frequency deviation to the modulating frequency.

Mathematically,

Modulation index, $$m_f = \frac{Frequency\, deviation}{Modulaion\ frequency}$$

$$or \ \ m_f = \frac{\Delta\omega}{\omega_m}$$

This modulation index may be greater than unity.

- The term "percentage modulation" as it is used in reference to FM refers to the ratio of actual frequency deviation to the maximum allowable frequency deviation. Thus 100% modulation corresponds to 75 kHz for the commercial FM broadcast band and 25 kHz for television.

Percent modulation $$M = \frac{\Delta f_{actual}}{\Delta f_{\max}}$$

- The expression for single-tone FM wave is

$$s(t) = A\cos\phi_i = A\cos(\omega_c t + m_f \sin\omega_m t)$$

Self-Assessment Exercise

A single-tone FM is represented by the voltage equation as:

- v(t) = 12 cos(6 x 10^8 t + 5 sin1250 t)

Determine the following:

i. carrier frequency
ii. modulating frequency
iii. the modulation index
iv. maximum deviation
v. what power will this FM wave dissipate in 10Ω resistor

Solution: We know that the standard expression for a single-tone FM wave is given as v (t) = $A\cos(\omega_c t + m_f \sin\omega_m t)$ ………………………..(i)

The given expression is v(t) = 12 cos(6 x 10^8 t + 5 sin1250 t) …….(ii)

Comparing equation (i) and (ii), we get

i. Carrier frequency

$\omega_c = 6\times10^8$ rad/sec or $f_c = \dfrac{6\times10^8}{2\pi} = 95.5MHz$

ii. modulating frequency

$\omega_m = 1250\,rad/\sec \quad or \quad f_m = \dfrac{1250}{2\pi} = 199Hz$

iii. $m_f = 5$

iv. maximum frequency deviation is given as

$$m_f = \frac{\Delta\omega}{\omega_m} = \frac{\Delta_f}{f_m}$$

or $\Delta_f = m_f \times f_m = 5\times199 = 995$

v. the power dissipated is

$$P = \frac{V_{rms}^2}{R} = \frac{(12/\sqrt{2})^2}{10} = \frac{72}{10} = 7.2watts$$

Self-Assessment Exercise

What is the modulation index of an FM signal having a carrier swing of 100 kHz when the modulating signal has a frequency of 8 kHz?

Solution: Given that

carrier swing = 100 kHz

modulating frequency f_m = 8 kHz

modulating index is given as

$$m_f = \frac{Frequency\,deviation}{Modulaion\,frequency} = \frac{\Delta f}{f_m} \quad \text{...........(i)}$$

but we know that

carrier swing = $2 \times \Delta_f$

$$\Delta_f = \frac{carrier\,swing}{2} = \frac{100}{2} = 50\,kHz$$

using equation (i) we get

$$m_f = \frac{50}{8} = 6.25$$

Self-Assessment Exercise

An FM transmission has a frequency deviation of 20 kHz.

i. Determine the percent modulation of this signal if it is broadcasted in the 88 – 08 MHz band.
ii. Calculate the percent modulation if this signal is broadcasted as the audio portion of a television broadcast.

Solution: Given that

$$\Delta_f = 20\,kHz$$

i. Percent modulation for an FM wave is defined as

$$M = \frac{\Delta f_{actual}}{\Delta f_{\max}} \times 100$$

Δf_{actual} is given as 20 kHz The maximum frequency deviation

$\Delta f_{\max}$ permitted in the FM broadcast band is 75 kHz.

Thus,

$$M = \frac{20\times10^3}{75\times10^3}\times100 = 26.67\%$$

(ii)

$$M = \frac{\Delta f_{actual}}{\Delta f_{\max}}\times100$$

$$\Delta_f = 20\,kHz$$

The maximum frequency deviation $\Delta f_{\max}$ permitted for FM audio portion of a TV broadcast is 25 kHz.

Thus, $$M = \frac{20\times10^3}{25\times10^3}\times100 = 80\%$$

Types of Frequency Modulation (FM)

Depending on the value of frequency sensitivity k_f, FM may be divided as:

(i) Narrowband FM: in this case k_f is small and hence the bandwidth of FM is narrow.

(ii) Wideband FM: in this case k_f is large and hence the FM signal has a wide bandwidth.

Application of FM

Broadcasting: FM is commonly used at VHF radio frequencies for high-fidelity broadcasts of music and speech and also for broadcasting Normal (analog) TV sound. A narrow band form is used for voice communications in commercial and amateur radio settings. The type of FM used in broadcast is generally called wide-FM, or W-FM. In two-way radio, narrowband narrow-fm (N-FM) is used to conserve bandwidth and also to send signals into space.

Sound: FM is used at audio frequencies to synthesise sound. This technique, known as FM synthesis, was popularised by early digital synthesisers and became a standard feature for several generations of personal computer sound cards

Hardware: FM is used at intermediate frequencies by all analog VCR systems, including VHS, to record both the luminance (black and white) and the chrominance portions of the video signal. FM is the only feasible method of recording video to and retrieving video from magnetic tape without extreme distortion, as video signals have a very large range of frequency components - from a few hertz to several megahertz, too wide for equalisers to work with due to electronic noise below -60 dB. FM also keeps the tape at saturation level, and therefore acts as a form of noise reduction, and a simple limiter can mask variations in the playback output, and the FM capture effect removes print-through and pre-echo.

(b)Multiple Frequency Modulations: is the type of frequency modulation (FM) in which the modulating or baseband signal contains multiple frequency.

3.1.2.1.2 Phase Modulation (PM)

Phase modulation is a form of modulation that represents information as variations in the instantaneous phase of a carrier wave. In phase modulation, the phase shift of the modulated signal is varied.

3.1.3Comparison of Angle Modulated Wave and Amplitude Modulated Wave

Figure 2.3 shows a single tone modulating signal, a carrier signal, amplitude-modulated (AM) wave and angle-modulated (i.e. FM and PM) waves.

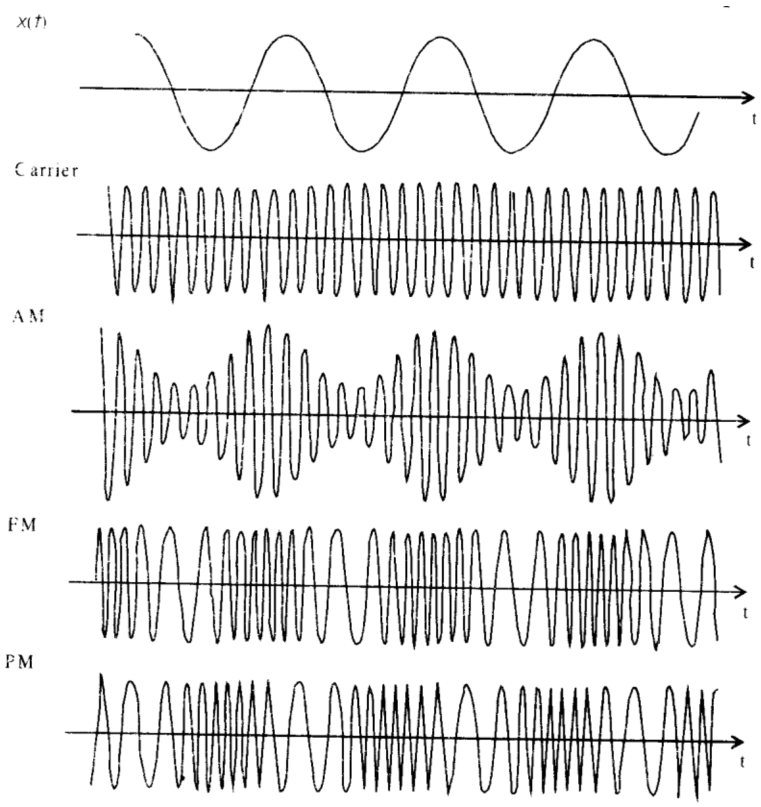

Fig. 2.6:AM, FM, and PM Waveforms

***Source*: Sharma S. (Wireless & Cellular Communication)**

On comparison, the following differences between the two techniques are:

- the envelope of FM wave or PM wave is constant and is equal to the unmodulated carrier amplitude. On the other hand, the envelope of AM wave is dependent on the modulating signal x(t)
- the zero crossings (i.e. the instants of time at which the waveform changes from negative to a positive value or vice-versa) of a FM wave or a PM wave no longer exhibit a perfect regularity in their spacing like AM wave. Thus this makes the instantaneous frequency of the angle modulated wave depend upon time.

3.1.4 Comparison of Frequency Modulation and Amplitude Modulation

(a)Advantages of FM over AM

- FM receivers may be fitted with amplitude limiters to remove the amplitude variations caused by noise. This makes FM reception a good deal more immune to noise than AM reception.
- It is possible to reduce noise still further by increasing the frequency-deviation. This is a feature which AM does not have because it is not possible to exceed 100 percent modulation without causing severe distortion.
- Standard Frequency Allocations provide a guard band between commercial FM stations. Due to this, there is less adjacent-channel interference than in AM.
- FM broadcast operates in the upper VHF and UHF frequency ranges at which there happens to be less noise than in the MF and HF ranges occupied by AM broadcasts.
- The amplitude of the FM wave is constant. It is thus independent of the modulation depth, whereas in AM, modulation depth governs the transmitted power.

(b) Disadvantages of FM over AM

- A much wider channel typically 200 KHz is required in FM as against only 10 KHz in AM broadcast. This forms serious limitation of FM.

- FM transmitting and receiving equipment particularly used for modulation and demodulation tend to be more complex and hence more costly.

Self-Assessment Exercise

i. What is angle modulation?

ii. What are the types of angle modulation?

4.0 Conclusion

You have been taken through the analog modulation concepts and its various techniques. Also, the comparison between angle modulated wave and amplitude modulated wave as well as the comparison between FM and AM was discussed.

5.0 Summary

In this unit, you have learnt that:

- in analog modulation, the modulation is applied continuously in response to the analog information signal
- the aim of **analog modulation** is to transfer an analog baseband signal over an analog pass band channel
- the two types of analog techniques are amplitude modulation and angle modulation
- amplitude modulation is a system in which the maximum amplitude of the carrier wave is made proportional to the instantaneous value (amplitude) of the modulating or baseband signal

- the amplitude modulation schemes are single sideband modulation (SSB), double-side band modulation (DSB), vestigial sideband modulation (VSB), and quadrature amplitude modulation (QAM)
- angle modulation is based on altering the angle (or *phase*) of a sinusoidal carrier wave to transmit data, as opposed to varying the amplitude, such as in AM transmission
- angle modulation schemes are frequency modulation and phase modulation.

6.0 Self-Assessment Exercise

i. Differentiate between the two types of analog modulation.
ii. Mention the application areas of Frequency Modulation
iii. A single-tone FM signal is given by $v(t) = 10\sin\left(16\pi\times10^{6}t + 20\sin 2\pi\times10^{3t}\right)volts$.

Determine the modulation index, modulating frequency, frequency deviation, carrier frequency and the power of the FM signal

iv.What are the principal merits and limitations of FM?

7.0 References/Further Reading

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Unit 3 Digital (Bandpass) Modulation Techniques

1.0 Introduction

Modern mobile communication systems use digital modulation techniques. Advancements in very large-scale integration (VLSI) and digital signal processing (DSP) technology have made digital modulation more cost effective than analog transmission systems.

2.0 Objectives

At the end of this unit, you should be able to:

- State the factors that govern the choice of digital modulation
- Explain the digital modulation techniques.

3.0 Main Content

3.1 An Overview of Digital Modulation Techniques

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

The aim of digital modulation is to transfer a digital bit stream over an analog passband channel, for example over the public switched telephone network (where a bandpass filter limits the frequency range to between 300 and 3400 Hz), or over a limited radio frequency band.

For example, a telephone line is designed for transferring audible sounds, for example tones, and not digital bits (zeros and ones).

Computers may however communicate over a telephone line by means of modems, which are representing the digital bits by tones, called symbols. If there are four alternative symbols (corresponding to a musical instrument that can generate four different tones, one at a time), the first symbol may represent the bit sequence 00, the second 01, the third 10 and the fourth 11. If the modem plays a melody consisting of 1000 tons per second, the symbol rate is 1000 symbols/second, or baud.

Since each tone represents a message consisting of two digital bits in this example, the bit rate is twice the symbol rate, i.e. 2000 bit per second.

3.2 Factors that Influence the Choice of Digital Modulation

The performance of a modulation scheme is often measured in terms of its power efficiency and bandwidth efficiency.

- **Power efficiency** describes the ability of a modulation technique to preserve the fidelity of the digital message at low power levels. In a digital communication system, in order to increase noise immunity, it is necessary to increase the signal power. However, the amount by which the signal power should be increased to obtain a certain level of fidelity (i.e., an acceptable bit error probability) depends on the particular type of modulation employed.

- **Bandwidth efficiency** describes the ability of a modulation scheme to accommodate data within a limited bandwidth. In general, increasing the data rate implies decreasing the pulse width of a digital symbol, which increases the bandwidth of the signal. If R is the data rate in bits per second, and B is the bandwidth occupied by the modulated RF signal, then bandwidth efficiency η_B is expressed as $\eta_B = \frac{R}{B} pss/Hz$

Shannon's channel coding theorem states that for an arbitrarily small probability error, the maximum possible bandwidth efficiency is limited by the noise in the channel, and is given by the channel capacity formula. Note that Shannon's bound applies for AWGN non-fading channels $\eta_B = \frac{C}{B} = \log_2\left(1 + \frac{S}{N}\right)$

Where C is the channel capacity (in bps), B is the RF bandwidth and

$\frac{S}{N}$ is the signal-to-noise ratio.

Self-Assessment Exercise

If the SNR of a wireless communication link is 20 dB and the RF bandwidth is 30 kHz. Calculate the maximum theoretical data rate which may be transmitted. Compare this rate to the US Digital Cellular Standard.

Solution

Given that $\frac{S}{N}$ = 20 dB = 100

RF Bandwidth B = 30000 Hz

Making use of Shannon's channel capacity expression the maximum possible data rate will be given by

$$C = B\log_2\left(1 + \frac{S}{N}\right) = 30000\log_2(1 + 100) = 199.75kbps$$

The USDC data rate is 48.6 kbps, which is only about the one-fourth the theoretical limit under 20 dB SNR conditions.

Self-Assessment Exercise

What is the theoretical maximum data rate which can be supported in a 200 kHz channel for SNR = 10 dB, 30 dB. How can this be compared with the GSM standard.

Solution

For SNR = 10 dB = 10, B = 200 kHz.

Making use of Shannon's channel capacity theorem, the maximum possible data rate, will be given by

$$C = B\log_2\left(1+\frac{S}{N}\right) = 200000\log_2(1+10) = 691.886kbps$$

The GSM data rate is 270.833 kbps, which is only about 40% of the theoretical limit for 10 dB SNR conditions.

For SNR = 30 dB = 1000,B = 200 kHz

The maximum possible data rate will be

$$C = B\log_2\left(1+\frac{S}{N}\right) = 200000\log_2(1+1000) = 1.99Mbps$$

3.3 List of Common Digital Modulation Techniques

Any digital modulation scheme uses a finite number of distinct signals to represent digital data. The most common digital modulation techniques are:

- Phase Shift Keying
- Frequency shift Keying
- Amplitude Shift Keying
- On-Off Keying
- Quadrature Amplitude Modulation
- Continuous Phase Modulation
- Spread-Spectrum Techniques

3.3.1 Phase-Shift Keying (PSK)

PSK is a digital modulation scheme that conveys data by changing, or modulating, the phase of a reference signal (the carrier wave). PSK uses a finite number of phases; each assigned a unique pattern of binary bits.

3.3.1.1Binary Phase Shift Keying (BPSK), Using M=2 symbols

In Binary phase shift keying, the binary symbols '1' and '0'modulate the phase of the carrier. With Binary Phase Shift Keying (BPSK), the binary digits 1 and 0 may be represented by the analog levels $+\sqrt{E_b}$ and $-\sqrt{E_b}$ respectively. The system model is as shown in the figure below.

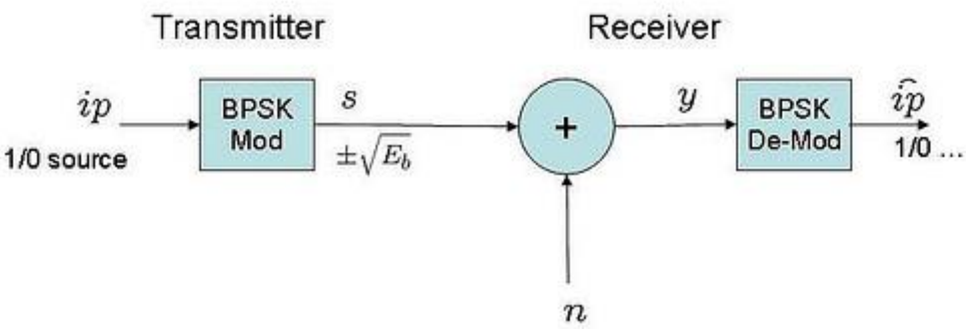

Fig. 3.1:Simplified Block Diagram with BPSK Transmitter-Receiver

The Bandwidth of BPSK signal will be,

BW = Highest frequency – Lowest frequency

BW = $f_c + f_b - (f_c - f_b)$

or $BW = 2f_b$

Hence the minimum bandwidth of BPSK signal is equal to twice of the highest frequency contained in baseband signal.

Probability of Bit Error Rate (BER) for BPSK modulation

The received signal, $y = s_1 + n$ when bit 1 is transmitted and $y = s_0 + n$ when bit 0 is transmitted.

The conditional probability distribution function (PDF) of y for the two cases are:

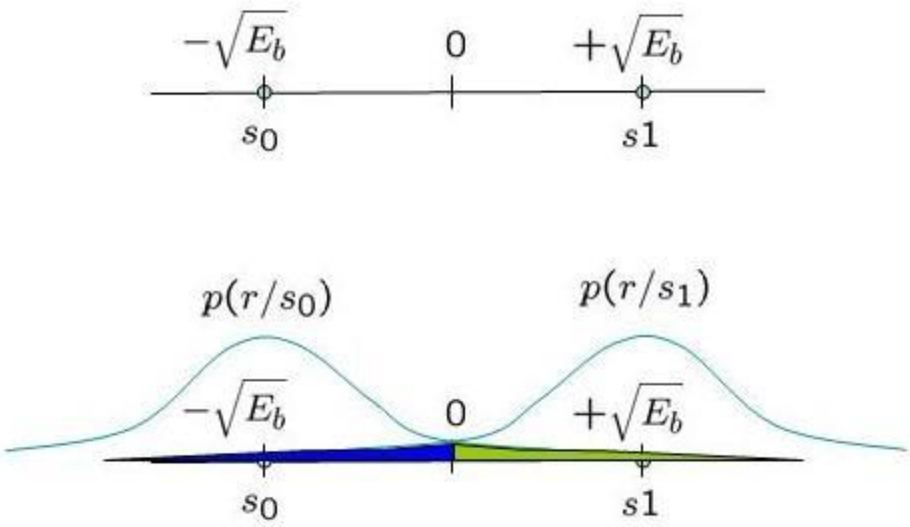

Fig. 3.2:Conditional Probability Density Function with BPSK Modulation

***Source*Krishna Sankar: 2007**

Probability of error given s_1 was transmitted

With this threshold, the probability of error given that s_1 was transmitted is (the area in blue region):

$$p(e \mid s_1) = \frac{1}{\sqrt{\pi N_0}} \int_{-\infty}^{0} e^{\frac{-(y-\sqrt{E_b})^2}{N_0}} dy \quad = \quad \frac{1}{\sqrt{N}} \int_{\sqrt{\frac{E_b}{N_0}}}^{\infty} e^{-z^2} dz = \frac{1}{2} erfc\left(\sqrt{\frac{E_b}{N_0}}\right),$$

where,

$$erfc(x) = \frac{2}{\sqrt{\pi}} \int_{x}^{\infty} e^{-x^2} dx$$

is the complementary error function.

Probability of error given s_0 was transmitted

Similarly, the probability of error given that s_0 was transmitted is (the area in green region):

$$p(e \mid s_0) = \frac{1}{\sqrt{\pi N_0}} \int_{-\infty}^{0} e^{\frac{-(y-\sqrt{E_b})^2}{N_0}} dy \quad = \quad \frac{1}{\sqrt{N}} \int_{\sqrt{\frac{E_b}{N_0}}}^{\infty} e^{-z^2} dz = \frac{1}{2} erfc\left(\sqrt{\frac{E_b}{N_0}}\right)$$

Total probability of bit error

$$P(e) = p(s_1)p(e \mid s1) + p(s_0)p(e \mid s_0)$$

Given that we assumed that s_1 and s_0 are equally probable i.e. $p(s_1)$ = $p(s_0)$ = $^1/_2$, the bit error probability is,

$$P(e) = \frac{1}{2} erfc\left(\sqrt{\frac{E_b}{N_0}}\right)$$

The error probability of BPSK reception using coherent/matched filter detection can be expressed as

$$P(e) = \frac{1}{2} erfc\sqrt{\frac{E_b}{N_0}}$$

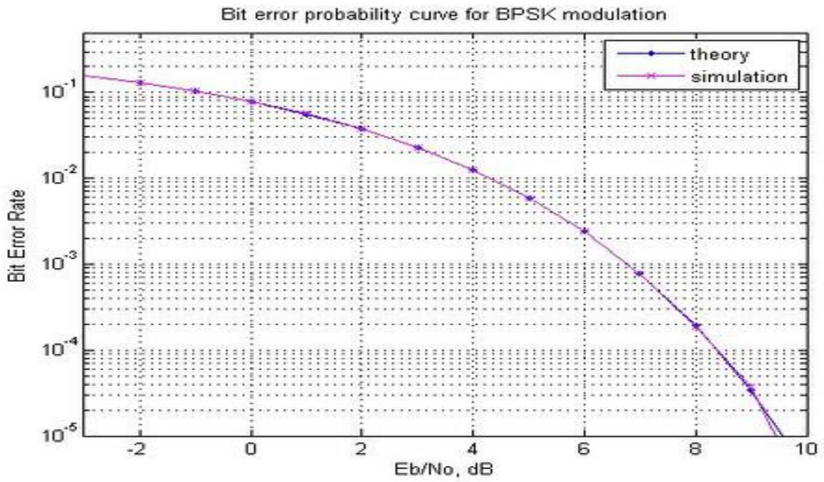

Fig. 3.3:Bit Error Rate (BER) Curve for BPSK Modulation – Theory, Simulation

***Source*Krishna Sankar: 2007**

3.3.1.2 Quadrature Phase Shift Keying (QPSK),

Using M=4 symbols

In communication systems, there are two main resources which are transmission power and the channel bandwidth. The channel bandwidth depends on the bit rate or signaling rate f_b. In digital bandpass transmission, carrier is used for transmission. This carrier is transmitted over a channel. If two or more bits are combined in some symbols, then the signaling rate will be reduced. Thus, the frequency of the carrier needed is also reduced. This reduces the transmission channel bandwidth. Hence, because of grouping of bits in symbols, the transmission channel bandwidth can be reduced. In quadrature phase shift keying (QPSK), two successive bits in the data sequence are grouped together. This reduces the bits rate or signaling rate (i.e. f_b) and thus reduces the bandwidth of the channel.

The bandwidth of QPSK signal will be $BW = 2 \times \frac{1}{2T_b} = f_b$ and the probability of error is

$$P_1^{'}(e) = P_2^{'}(e) = \frac{1}{2} erfc\sqrt{\frac{2E_b}{2N_0}} = \frac{1}{2} erfc\sqrt{\frac{E_b}{N_0}}$$

Thus, bit error probability of QPSK and BPSK are the same.

3.3.1.3 Differential Phase Shift Keying (DPSK)

DPSK does not need a synchronous (coherent) carrier at the demodulator. The input sequence of binary bits is modified such that the next bit depends upon the previous bit.

The Bandwidth is expressed as $BW = \frac{2}{T} = \frac{1}{T_b} = f_b$

and the average probability of error or bit error rate (BER) of DPSK can be expressed as

$$P(e) = \frac{1}{2} e^{-E_b / N_0}$$

3.3.2 Frequency-Shift Keying (FSK)

FSK is a frequency modulation scheme in which digital information is transmitted through discrete frequency changes of a carrier wave. The simplest FSK is binary FSK (BFSK). BFSK literally implies using a couple of discrete frequencies to transmit binary (0s and 1s) information.

In BFSK, the frequency of the carrier is shifted according to the binary symbol. However, the phase of the carrier is unaffected. This means that we have two different frequency signals according to binary symbols.

With this scheme, the "1" is called the mark frequency and the "0" is called the space frequency. In the case of FSK, a finite number of frequencies are used.

The bandwidth of $\mathrm{BFSK} = 2f_b + 2f_b$

or BW = $4f_b$

Now, if we compare this bandwidth with that of BPSK, we note that

BW (BFSK) = 2 x BW(BPSK)

The probability of error, of BFSK signal can be expressed as

$$P(e) = \frac{1}{2} erfc\left(\frac{\sqrt{2E_b}}{2\sqrt{N_0}}\right) \quad \text{or} \quad P(e) = \frac{1}{2} erfc\left(\sqrt{\frac{E_b}{2N_0}}\right)$$

This equation shows that error probability of BFSK is higher compared to that of BPSK

The various frequency-shift keying scheme are

3.3.2.1 Audio Frequency-Shift Keying (AFSK)

AFSK is a modulation technique by which digital data is represented by changes in the frequency (pitch) of an audio tone, yielding an encoded signal suitable for transmission via radio or telephone. Normally, the transmitted audio alternates between two tones: one, the "mark", represents a binary one; the other, the "space", represents a binary zero.

AFSK differs from regular frequency-shift keying in performing the modulation at baseband frequencies. In radio applications, the AFSK-modulated signal normally is being used to modulate an RF carrier (using a conventional technique, such as AM or FM) for transmission.

AFSK is not always used for high-speed data communications, since it is far less efficient in both power and bandwidth than the other modulation modes. In addition to its simplicity, however, AFSK has the advantage that encoded signals will pass through AC-coupled links, including most equipment originally designed to carry music or speech.

3.3.2.2 Continuous-Phase Frequency-Shift Keying (CPFSK)

CPFSK is a commonly-used variation of frequency-shift keying (FSK), which is itself a special case of analog frequency modulation. FSK is a method of modulating digital data onto a sinusoidal carrier wave, encoding the information present in the data to variations in the carrier's instantaneous frequency between one of two frequencies (referred to as the space frequency and mark frequency). In general, a standard FSK signal does not have continuous phase, as the modulated waveform switches instantaneously between two sinusoids with different frequencies.

As the name suggests, the phase of a CPFSK is in fact continuous; this attribute is desirable for signals that are to be transmitted over a band limited channel, as discontinuities in a signal introduce wideband frequency components. In addition, some classes of amplifiers exhibit nonlinear behavior when driven with nearly-discontinuous signals; this could have undesired effects on the shape of the transmitted signal.

3.3.2.3 Multi-Frequency Shift Keying (M-ary FSK or MFSK)

MFSK is a variation of frequency-shift keying (FSK) that uses more than two frequencies. MFSK is a form of M-ary orthogonal modulation, where each symbol consists of one element from an alphabet of orthogonal waveforms. M, the size of the alphabet, is usually a power of two so that each symbol represents $\log_2 M$ bits. M is usually between 2 and 64

3.3.2.4 Dual-Tone Multi-Frequency (DTMF)

DTMF signaling is used for telecommunication signaling over analog telephone lines in the voice-frequency band between telephone handsets and other communications devices and the switching center. As a method of in-band signaling, DTMF tones were also used by cable television broadcasters to indicate the start and stop times of local commercial insertion points during station breaks for the benefit of cable companies.

3.3.3 Amplitude-Shift Keying (ASK)

ASK is a form of modulation that represents digital data as variations in the amplitude of a carrier wave. The amplitude of an analog carrier signal varies in accordance with the bit stream (modulating signal), keeping frequency and phase constant. In the case of ASK, a finite number of amplitudes are used.

3.3.4 On-Off Keying (OOK)

OOK is the simplest form of amplitude-shift keying (ASK) modulation that represents digital data as the presence or absence of a carrier wave. In its simplest form, the presence of a carrier for a specific duration represents a binary one, while its absence for the same duration represents a binary zero. Some more sophisticated schemes vary these durations to convey additional information. It is analogous to unipolar encoding line code.

i. M-ary vestigial sideband modulation, for example 8VSB: A vestigial sideband (in radio communication) is a sideband that has been only partly cut off or suppressed.

3.3.5 Quadrature Amplitude Modulation (QAM)

QAM is a combination of PSK and ASK. QAM is both an analog and a digital modulation scheme. It conveys two analog message signals, or two digital bit streams, by changing (modulating) the amplitudes of two carrier waves, using the amplitude-shift keying (ASK) digital modulation scheme or amplitude modulation (AM) analog modulation scheme. These two waves, usually sinusoids, are out of phase with each other by 90° and are thus called quadrature carriers or quadrature components — hence the name of the scheme. The modulated waves are summed, and the resulting waveform is a combination of both phase-shift keying (PSK) and amplitude-shift keying, or in the analog case of phase modulation (PM) and amplitude modulation. In the digital QAM case, a finite number of at least two phases and at least two amplitudes are used.

i. Polar modulation like QAM is a combination of PSK and ASK: It makes use of polar coordinates, *r* (amplitude) and *Θ* (phase).

3.3.6 Continuous Phase Modulation (CPM) Method

CPM is a method for modulation of data commonly used in wireless modems. In contrast to other coherent digital phase modulation techniques where the carrier phase abruptly resets to zero at the start of every symbol (e.g. M-PSK), with CPM the carrier phase is modulated in a continuous manner.

(i) Minimum-Shift Keying (MSK): In MSK, the output waveform is continuous in phase hence there are no abrupt changes in amplitude. The sidelobes of MSK are very small hence bandpass filtering is not required to avoid interchannel interference.

The Bandwidth of MSK is BW = 1.5 f_b

(ii) Gaussian Minimum-Shift Keying (GMSK): It is similar to standard minimum-shift keying (MSK); however the digital data stream is first shaped with a Gaussian filter before being applied to a frequency modulator. This has the advantage of reducing sideband power, which in turn reduces out-of-band interference between signal carriers in adjacent frequency channels. However, the Gaussian filter increases the modulation memory in the system and causes intersymbol interference, making it more difficult to discriminate between different transmitted data values and requiring more complex channel equalization algorithms such as an adaptive equalizer at the receiver. GMSK is most notably used in the Global System for Mobile Communications (GSM).

3.3.7 Orthogonal Frequency Division Multiplexing (OFDM) Modulation

OFDM is essentially identical to Coded OFDM (COFDM) and Discrete multi-tone modulation (DMT). It is a frequency-division multiplexing (FDM) scheme utilised as a digital multi-carrier modulation method. A large number of closely-spaced orthogonal sub-carriers are used to carry data. The data is divided into several parallel data streams or channels, one for each sub-carrier. Each sub-carrier is modulated with a conventional modulation scheme (such as quadrature amplitude modulation or phase shift keying) at a low symbol rate, maintaining total data rates similar to conventional single-carrier modulation schemes in the same bandwidth. The primary advantage of OFDM over single-carrier schemes is its ability to cope with severe channel conditions — for example, attenuation of high frequencies in a long copper wire, narrowband interference and frequency-selective fading due to multipath — without complex equalisation filters.

3.3.8 Wavelet Modulation

Wavelet modulation, also known as fractal modulation, is a modulation technique that makes use of wavelet transformations to represent the data being transmitted. One of the objectives of this type of modulation is to send data at multiple rates over a channel that is unknown. If the channel is not clear for one specific bit rate, meaning that the signal will not be received, the signal can be sent at a different bit where the signal to noise ratio is higher.

3.3.9 Trellis Coded Modulation (TCM)

Trellis coded modulation, also known as trellis modulation is a modulation scheme which allows highly efficient transmission of information over band-limited channels such as telephone lines.

3.3.10 Spread-Spectrum Techniques

Spread-spectrum techniques are methods by which electromagnetic energy generated in a particular bandwidth is deliberately spread in the frequency domain, resulting in a signal with a wider bandwidth. These techniques are used for a variety of reasons, including the establishment of secure communications, increasing resistance to natural interference and jamming, to prevent detection, and to limit the power flux density on satellite downlinks.

i. **Direct-sequence spread spectrum (DSSS):** as with other spread spectrum technologies, the transmitted signal takes up more bandwidth than the information signal that is being modulated. The name 'spread spectrum' comes from the fact that the carrier signals occur over the full bandwidth (spectrum) of a device's transmitting frequency.

ii. **Chirp spread spectrum (CSS):** uses pseudo-stochastic coding. It is a spread spectrum technique that uses wideband linear frequency modulated chirp pulses to encode information. A chirp is a sinusoidal signal whose frequency increases or decreases over a certain amount of time.

iii. **Frequency-hopping spread spectrum (FHSS):** it applies a special scheme for channel release. It is a method of transmitting radio signals by rapidly switching a carrier among many frequency channels, using a pseudorandom sequence known to both transmitter and receiver. It is utilised as a multiple access method in the frequency-hopping code division multiple access (FH-CDMA) scheme.

3.4 Modulator and Detector Principles of Operation

PSK and ASK, and sometimes also FSK, are often generated and detected using the principle of QAM. The I and Q signals can be combined into a complex-valued signal I+jQ (where j is the imaginary unit). The resulting so called equivalent lowpass signal or equivalent baseband signal is a complex-valued representation of the real-valued modulated physical signal (the so called passband signal or RF signal).

Below are the general steps used by the modulator to transmit data.

i. Group the incoming data bits into codewords, one for each symbol that will be transmitted.
ii. Map the codewords to attributes, for example amplitudes of the I and Q signals (the equivalent low pass signal), or frequency or phase values.
iii. Adapt pulse shaping or some other filtering to limit the bandwidth and form the spectrum of the equivalent low pass signal, typically using digital signal processing.
iv. Perform digital-to-analog conversion (DAC) of the I and Q signals (since today all of the above is normally achieved using digital signal processing, DSP).
v. Generate a high-frequency sine wave carrier waveform, and perhaps also a cosine quadrature component. Carry out the modulation, for example by multiplying the sine and cosine wave form with the I and Q signals, resulting in that the equivalent low pass signal is frequency shifted into a modulated passband signal or RF signal. Sometimes this is achieved using DSP technology, for example direct digital synthesis using a waveform table, instead of analog signal processing. In that case the above DAC step should be done after this step.
vi. Amplification and analog bandpass filtering to avoid harmonic distortion and periodic spectrum.

At the receiver side, the demodulator typically performs:
i. bandpass filtering.
ii. automatic gain control, AGC (to compensate for attenuation, for example fading).
iii. frequency shifting of the RF signal to the equivalent baseband I and Q signals, or to an intermediate frequency (IF) signal, by multiplying the RF signal with a local oscillator sinewave and cosine wave frequency (see the superheterodyne receiver principle).
iv. sampling and analog-to-digital conversion (ADC) (Sometimes before or instead of the above point, for example by means of undersampling)
v. equalisation filtering, for example a matched filter, compensation for multipath propagation, time spreading, phase distortion and frequency selective fading, to avoid intersymbol interference and symbol distortion
vi. detection of the amplitudes of the I and Q signals, or the frequency or phase of the IF signal
vii. quantisation of the amplitudes, frequencies or phases to the nearest allowed symbol values
viii. mapping of the quantised amplitudes, frequencies or phases to codewords (bit groups)
ix. parallel-to-serial conversion of the codewords into a bit stream.
x. pass the resultant bit stream on for further processing such as removal of any error-correcting codes.

As it is common to all digital communication systems, the design of both the modulator and demodulator must be done simultaneously.

Digital modulation schemes are possible because the transmitter-receiver pair have prior knowledge of how data is encoded and represented in the communications system. In all digital communication systems, both the modulator at the transmitter and the demodulator at the receiver are structured so that they perform inverse operations.

Non-coherent modulation methods do not require a receiver reference clock signal that is phase synchronised with the sender carrier wave. In this case, modulation symbols (rather than bits, characters, or data packets) are asynchronously transferred. The opposite is coherent modulation.

Self-Assessment Exercise

What factors govern the choice of digital modulation?

4.0 Conclusion

In this unit, the concept of digital bandpass modulation and its various techniques, as well as the factors that govern the choice of digital modulation were discussed.

5.0 Summary

In this unit you have learnt that:

- the aim of digital modulation is to transfer a digital bit stream over an analog passband channel

- in digital modulations, instead of transmitting one bit at a time, two or more bits are transmitted simultaneously
- the factors that influence the choice of digital (bandpass) modulation are power efficiency and bandwidth efficiency
- power efficiency describes the ability of a modulation technique to preserve the fidelity of the digital message at low power levels
- bandwidth efficiency describes the ability of a modulation scheme to accommodate data within a limited bandwidth
- the common digital modulation techniques are PSK, FSK, ASK, OOK,QAM,CPM,OFDM, Wavelet modulation, TCM and spread-spectrum techniques
- the most fundamental digital modulation techniques are PSK, FSK, ASK and QAM.

6.0 Tutor-Marked Assignment

Explain briefly four major digital modulation techniques**.**

7.0 References/Further Reading

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Sharma, S. (2006). *Wireless & Cellular Communications*. New Delhi: S. K. Kataria & Sons.

Unit 4 Digital Baseband and Pulse Shaping Modulation Techniques

1.0 Introduction

In this unit, you will learn baseband and pulse shaping modulation techniques. Baseband modulation aims at transferring a digital bit stream over a baseband channel, as an alternative to carrier-modulated approaches. The term digital baseband modulation (or digital baseband transmission) is synonymous to line codes. Line codes are codes chosen for use within a communications system for baseband transmission purposes.

2.0 Objectives

At the end of this unit, you should be able to:

- discuss digital baseband modulation
- write concisely on pulse modulation
- explain briefly the types of pulse modulation.

3.0 Main Content

3.1 Digital Baseband and Pulse Shaping Modulation

3.1.1 Digital Baseband Modulation or Line Coding Techniques

Line coding is a method which transfer a digital bit stream over an analog baseband channel (lowpass channel) using a pulse train, i.e. a discrete number of signal levels, by directly modulating the voltage or current on a cable such as fiber optical cables and short-range copper cables, for example serial cables and LAN networks.

Baseband signal is used to modulate a higher frequency carrier wave so that it may be transmitted via radio. Shifting the signal to higher frequencies (radio frequencies, or RF) than it originally spanned result in modulation. The key consequence of the usual double-sideband amplitude modulation (AM) is that, the range of frequencies of the signal span (its spectral bandwidth) is doubled. Thus, the RF bandwidth of a signal (measured from the lowest frequency as opposed to 0 Hz) is usually twice its baseband bandwidth. Steps may be taken to reduce this effect, such as single-sideband modulation; the highest frequency of such signals greatly exceeds the baseband bandwidth.

Some signals can be treated as baseband or not, depending on the situation. For example, a switched analog connection in the telephone network has energy below 300 Hz and above 3400 Hz removed by bandpass filtering; since the signal has no energy very close to zero frequency, it may not be considered a baseband signal, but in the telephone systems frequency-division multiplexing hierarchy, it is usually treated as a baseband signal, by comparison with the modulated signals used for long-distance transmission. The 300 Hz lower band edge in this case is treated as “near zero”, being a small fraction of the upper band edge.

Figure 4.1 below depicts what happens with AM modulation:

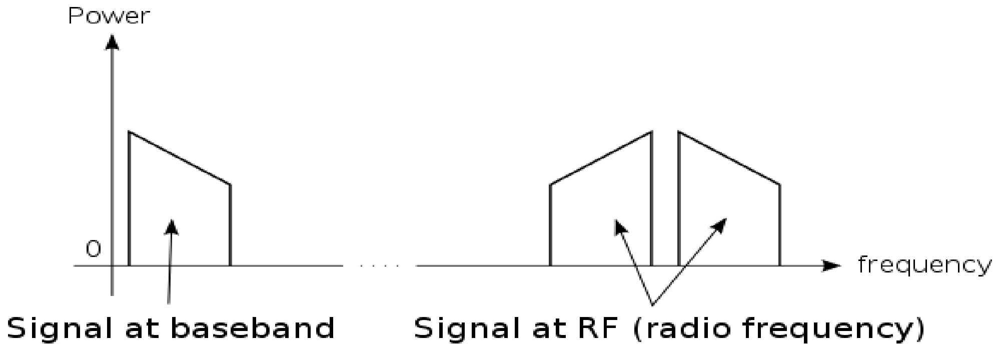

Fig. 4.1:Comparing Baseband Version of a Signal and its Equivalent AM-modulated (double-sideband) RF Version, showing the Typical Doubling of the Occupied Bandwidth

*Source*Wikipedia (2009)

The composite video signal created by devices such as the new VCRs, game consoles and DVD players are commonly used baseband signal.

3.1.2 Pulse Shaping Modulation Techniques

In digital telecommunication, pulse shaping is the process of changing the waveform of transmitted pulses. Its purpose is to make the transmitted signal suit better to the communication channel by limiting the effective bandwidth of the transmission. By filtering the transmitted pulses this way, the intersymbol interference caused by the channel can be kept in control. In RF communication pulse shaping is essential for making the signal fit in its frequency band.

These methods are used to transfer a digital bit stream over an analog baseband channel (lowpass channel) using a pulse train. Some pulse modulation schemes also allow the narrowband analog signal to be transferred as a digital signal (i.e. as a quantised discrete-time signal) with a fixed bit rate, which can be transferred over an underlying digital transmission system, for example some line code. These are not modulation schemes in the conventional sense since they are not channel coding schemes, but should be considered as source coding schemes, and in some cases analog-to-digital conversion techniques.

3.1.2.1 Analog-Over-Analog Methods

Pulse-Amplitude Modulation (PAM) is a form of signal modulation where the message information is encoded in the amplitude of a series of signal pulses. For example, a two bit modulator (PAM-4) will take two bits at a time and will map the signal amplitude to one of four possible levels, for example −3 volts, −1 volt, 1 volt, and 3 volts.

Pulse-width modulation (PWM) is a very efficient way of providing intermediate amounts of electrical power between fully on and fully off. A simple power switch with a typical power source provides full power only, when switched on. PWM is a comparatively-recent technique, made practical by modern electronic power switches.

Pulse-position modulation (PPM) is a form of signal modulation in which M message bits are encoded by transmitting a single pulse in one of 2^M possible time-shifts. This is repeated every T seconds, such that the transmitted bit rate is M/T bits per second. It is primarily useful for optical communications systems, where there tends to be little or no multipath interference.

3.1.2.2 Analog-Over-Digital Methods

Pulse-Code Modulation (PCM) is a digital representation of an analog signal where the magnitude of the signal is sampled regularly at uniform intervals, then quantised to a series of symbols in a numeric (usually binary) code. PCM has been used in digital telephone systems, 1980s-era electronic musical keyboards, digital audio in computers, digital video and the compact disc "red book" format.

Differential pulse-code modulation (DPCM) is a signal encoder that uses the baseline of PCM but adds some functionality based on the prediction of the samples of the signal. The input can be an analog signal or a digital signal. If the input is a continuous-time analog signal, it needs to be sampled first so that a discrete-time signal is the input to the DPCM encoder.

Adaptive DPCM (ADPCM) is a variant of DPCM (differential pulse-code modulation) that varies the size of the quantisation step, to allow further reduction of the required bandwidth for a given signal-to-noise ratio.

Delta modulation (DM or Δ-modulation) is an analog-to-digital and digital-to-analog signal conversion technique used for transmission of voice information where quality is not of primary importance. DM is the simplest form of differential pulse-code modulation (DPCM) where the difference between successive samples is encoded into n-bit data streams. In delta modulation, the transmitted data is reduced to a 1-bit data stream.

Its main features are:

- the analog signal is approximated with a series of segments
- each segment of the approximated signal is compared to the original analog wave to determine the increase or decrease in relative amplitude
- the decision process for establishing the state of successive bits is determined by this comparison
- only the change of information is sent, that is, only an increase or decrease of the signal amplitude from the previous sample is sent whereas a no-change condition causes the modulated signal to remain at the same 0 or 1 state of the previous sample.

Delta-Sigma (ΔΣ) or sigma-delta (ΣΔ) modulation is a method for encoding high resolution signals into lower resolution signals using pulse-density modulation. This technique has found increasing use in a range of modern electronic components, such as analog-to-digital and digital-to-analog converters, frequency synthesizers, switched-mode power supplies and motor controls. One of the earliest and most widespread uses of delta-sigma modulation is in data conversion.

Continuously variable slope delta modulation (CVSDM) also called Adaptive-delta modulation (ADM) is a voice coding method. It is a delta modulation with variable step size (i.e. special case of adaptive delta modulation).CVSD encodes at 1 bit per sample, so that audio sampled at 16 kHz is encoded at 16 kbit/s.

Pulse-density modulation (PDM) is a form of modulation used to represent an analog signal in the digital domain. In a PDM signal, specific amplitude values are not encoded into pulses as they would be in PCM. Instead, it is the relative density of the pulses that corresponds to the analog signal's amplitude. Pulse-width modulation (PWM) is the special case of PDM where all the pulses corresponding to one sample are contiguous in the digital signal.

Self-Assessment Exercise

What is digital baseband modulation?

4.0 Conclusion

Digital baseband modulations and pulse modulation were discussed in this unit.

5.0 Summary

In this unit you have learnt that:

- baseband modulation aims at transferring a digital bit stream over a baseband channel, as an alternative to carrier-modulated approaches
- pulse shaping is the process of changing the waveform of transmitted pulses
- some pulse shaping modulation schemes also allow the narrowband analog signal to be transferred as a digital signal (i.e. as a quantized discrete-time signal) with a fixed bit rate, which can be transferred over an underlying digital transmission system.

6.0 Self-Assessment Exercise

Explain briefly two types of pulse modulation.

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Unit 5 Diversity Techniques for Fading Channel

1.0 Introduction

Diversity is a powerful communication receiver technique that provides wireless link improvement at relatively low cost. Diversity exploits the random nature of radio propagation by finding independent or at least highly uncorrected signal paths for communication.

2.0 Objectives

At the end of this unit, you should be able to:

- explain diversity concept
- discuss the types of diversity techniques
- outline space diversity scheme
- illustrate the application of space diversity technique.

3.0 Main Content

3.1 Concept of Diversity

A signal transmitted at a particular carrier frequency and at a particular instant of time may be received in a multipath null. Diversity reception reduces the probability of occurrence of communication failures (outages) caused by fades by combining several copies of the same message received over different channels.

In telecommunications, diversity scheme is referred to as a method for improving the reliability of a message signal by using two or more communication channels with different characteristics. Diversity plays an important role in combating fading and co-channel interference and avoiding error bursts. It is based on the fact that individual channels experience different levels of fading and interference. Multiple versions of the same signal may be transmitted and/or received and combined in the receiver. Alternatively, a redundant forward error correction code may be added and different parts of the message transmitted over different channels. Diversity techniques may exploit the multipath propagation, resulting in a diversity gain, often measured in decibels. In general, the efficiency of the diversity techniques reduces if the signal fading is correlated at different branches.

Diversity is a commonly used technique in wireless systems to combat channel fading, due to the following facts:

(i) the degradation of transmission quality due to channel fading cannot be simply overcome by increasing the transmitted signal power. This is because, even with high transmitted power, when the channel is in deep fading, the instantaneously received SNR can still be very low, resulting in a high probability of transmission error during the deep fading period.

(ii) in wireless communications, the power available on the reverse link is severely limited by the battery capacity in hand-held subscriber units. With diversity, the required transmitted power can be greatly reduced.

(iii) cellular communications systems are mostly limited by interference. Also, mitigation of channel fading by diversity reception can translate into improved interference tolerance which, in turn, means greater ability to support additional users and therefore higher system capacity.

3.1.1 Diversity Techniques

The following classes of diversity schemes can be identified:

Time diversity: multiple versions of the same signal are transmitted at different time instants. Alternatively, a redundant forward error correction code is added and the message is spread in time by means of bit-interleaving before it is transmitted. Thus, error bursts are avoided, which simplifies the error correction. In time diversity, the time difference between two transmissions should be large compared to the time it takes the mobile antenna to move half a wavelength. In systems with stationary antennas, such as indoor wireless communication, time diversity will be less effective as the channel characteristics do not change very much with time. However, time diversity may be helpful if uncorrected interference signals are experienced during successive attempts.

Frequency diversity: in frequency diversity, the same message is transmitted more than once, respectively at different carrier frequencies. The difference in carrier frequency should be more than the coherence bandwidth to achieve effective diversity. Digital cellular system can use slow frequency hopping (SFH) for diversity reason: each block of bits is transmitted at a different carrier.

Angle diversity: the desired message is received simultaneously by several directed antennas pointing in widely different directions. The received signal consists of scattering wave coming from all directions. It has been observed that the scattered signals associated with the different (non-overlapping) directions are uncorrelated. Angle diversity can be viewed as a special case of space diversity since it also requires multiple antennas.

Multiuser diversity: multiuser diversity is obtained by opportunistic user scheduling at either the transmitter or the receiver. Opportunistic user scheduling is as follows: the transmitter selects the best user among candidate receivers according to the qualities of each channel between the transmitter and each receiver. In Frequency Division Duplex (FDD) systems, a receiver must feedback the channel quality information to the transmitter with the limited level of resolution.

Co-operative diversity: is a co-operative multiple antenna techniques which exploit user diversity by decoding the combined signal of the relayed signal and the direct signal in wireless multihop networks. A conventional single hop system uses direct transmission where a receiver decodes the information only based on the direct signal while regarding the relayed signal as interference, whereas the cooperative diversity considers the other signal as contribution. That is, cooperative diversity decodes the information from the combination of two signals. Hence, it can be seen that cooperative diversity is an antenna diversity that uses distributed antennas belonging to each node in a wireless network. Note that user co-operation is another definition of co-operative diversity. User co-operation considers an additional fact that each user relays the other user's signal while co-operative diversity can also be achieved by multi-hop relay networking systems.

Co-operative diversity achieves antenna diversity gain by using the cooperation of distributed antennas belonging to each node. Co-operative diversity decodes information from the combination of two signals.

Space diversity, also known as **Antenna diversity,** is any one of several wireless diversity schemes that use two or more antennas to improve the quality and reliability of a wireless link. Often, especially in urban and indoor environments, there is no clear line-of-sight (LOS) between transmitter and receiver. Instead the signal is reflected along multiple paths before finally being received. Each of these bounces can introduce phase shifts, time delays, attenuations, and even distortions that can destructively interfere with one another at the aperture of the receiving antenna. Antenna diversity is especially effective at mitigating these multipath situations.

The desired message is transmitted by using multiple transmitting antennas (transmit diversity) and/or receiving antennas (reception diversity). This is because multiple antennas afford a receiver several observations of the same signal. The space separation between adjacent antennas should be large enough to ensure that the signals from different antennas are independently faded; each of the antennas will experience a different interference environment. Thus, if one antenna is experiencing a deep fade, it is likely that another has a sufficient signal. Collectively, such a system can provide a robust link.

3.2 Space/Antenna Diversity Scheme

Antenna diversity can be realised in several ways. Depending on the environment and the expected interference, designers can employ one or more of these methods to improve signal quality. In fact, multiple methods are frequently used to further increase reliability.

Spatial Diversity – Spatial diversity employs multiple antennas, usually with the same characteristics, that are physically separated from one another. Depending upon the expected incidence of the incoming signal, sometimes a space on the order of a wavelength is sufficient. Other times much larger distances are needed. Cellularisation or sectorisation, for example, is a spatial diversity scheme that can have antennas or base stations miles apart. This is especially beneficial for the mobile communications industry since it allows multiple users to share a limited communication spectrum and avoid co-channel interference.

Pattern Diversity – Pattern diversity consists of two or more co-located antennas with different radiation patterns. This type of diversity makes use of directive antennas that are usually physically separated by some (often short) distance. Collectively they are capable of discriminating a large portion of angle space and can provide a higher gain versus a single omnidirectional radiator.

Polarisation Diversity – Polarisation diversity combines pairs of antennas with orthogonal polarisations (i.e. horizontal/vertical, ± slant 45°, Left-hand/Right-hand CP etc). Reflected signals can undergo polarization changes depending on the media. By pairing two complementary polarisations, this scheme can immunise a system from polarisation mismatches that would otherwise cause signal fade. Additionally, such diversity has proven valuable at radio and mobile communication base stations since it is less susceptible to the near random orientations of transmitting antennas.

Transmitter/Receiver Diversity–Transmitter/Receiver diversity uses two separate, collocated antennas for transmitting and receiving functions. Such a configuration eliminates the need for a duplexer and can protect sensitive receiver components from the high power used in transmission.

Adaptive Arrays – Adaptive arrays can be a single antenna with active elements or an array of similar antennas with ability to change their combined radiation pattern as different conditions persist. Active electronically scanned arrays (AESAs) manipulate phase shifters and attenuators at the face of each radiating site to provide a near instantaneous scan ability as well as pattern and polarization control. This is especially beneficial for radar applications since it affords a signal antenna the ability to switch among several different modes such as searching, tracking, mapping and jamming countermeasures.

3.2.1 Applications of Space/Antenna diversity Techniques

- A well-known practical application of diversity reception is in wireless microphones, and in similar electronic devices such as wireless guitar systems. A wireless microphone with a non-diversity receiver (a receiver having only one antenna) is prone to random drop-outs, fades, noise, or other interference, especially if the transmitter (the wireless microphone) is in motion. A wireless microphone or sound system using diversity reception will switch to the other antenna within microseconds if one antenna experiences noise, providing an improved quality signal with fewer drop-outs and noise. Ideally, no drop-outs or noise will occur in the received signal.

- Another common usage is in Wi-Fi networking gear and cordless telephones to compensate for multipath interference. The base station will switch reception to one of two antennas depending on which is currently receiving a stronger signal. For best results, the antennas are usually placed one wavelength apart. For microwave bands, where the wavelengths are under 100 cm, this can often be done with two antennas attached to the same hardware. For lower frequencies and longer wavelengths, the antennas must be multiple meters apart, making it much less reasonable.

- Mobile phone towers also often take advantage of diversity - each face of a tower will often have three antennas; one is transmitting, while the other two perform diversity reception.

- The uses of multiple antennas at both transmitter and receiver results in a multiple-input multiple-output (MIMO) system. The use of diversity techniques at both ends of the link is termed space–time coding.

Self-Assessment Exercise

What is meant by diversity?

4.0 Conclusion

In this unit, the concept of diversity and different types of diversity techniques were discussed. It was revealed that diversity improves transmission performance by making use of more than one independently faded version of the transmitted signal. If several replicas of the signal, carrying the same information, are received over multiple channels that exhibit independent fading with comparable strengths, the chances that all the independently faded signal components experience deep fading simultaneously are greatly reduced.

5.0 Summary

- diversity is a powerful communication receiver technique that provides wireless link improvement at relatively low cost
- diversity reception reduces the probability of occurrence of communication failures (outages) caused by fades by combining several copies of the same message received over different channels
- diversity scheme is refers to as a method for improving the reliability of a message signal by using two or more communication channels with different characteristics
- the classes of diversity schemes are time diversity, frequency diversity, space diversity, angle diversity, multiple diversity and cooperative diversity techniques.

6.0 Tutor-Marked Assignment

i. Write short notes on any four diversity techniques.
ii. State the application areas of space diversity techniques.
iii. Mention the types of space diversity schemes.

7.0 References/Further Reading

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Unit 6 Multiple Access Techniques

1.0 Introduction

Multiple Access is a signal transmission situation in which two or more users wish to simultaneously communicate with each other using the same propagation channel. This is precisely the uplink transmission situation in a wireless communication. In the uplink or reverse channel, multiple users will want to transmit information simultaneously.

Without proper coordination among the transmitting users, collisions will occur when two or more users transmit simultaneously. Access methods that incur collision are referred to as random access and variants of random access.

Multiple access method allows several terminals connected to the same multi-point transmission medium to transmit over it and to share its capacity.

2.0 Objectives

At the end of this unit, you should be able to:

- explain Multiple Access Concept
- discuss the types of Multiple Access Techniques
- define Carrier Sense Multiple Access
- state the types of Carrier Sense Multiple Access
- describe the attribute of Carrier Sense Multiple Access
- mention the application areas of Multiple Access.

3.0 Main Content

3.1 Multiple Access in a Radio Cell

In each radio cell, the transmission from the base station in the downlink can be heard by each and every mobile user in the cell. For this reason, this mode of transmission is referred to as broadcasting. On the other hand, a transmission from the mobile user in the uplink to the base station is many-to-one, and is referred to as multiple access.

Transmission in the uplink mode has the following properties:

- multiple mobile users want to access the common resource (base station) simultaneously;
- if the transmission from two or more users arrive at the base station at the same time, there will be destructive interference, umless the multiple arriving signals are mutually orthogonal;
- orthogonality between two signals x_i (t) and x_j (t), t [0,T], means that their inner product over the signaling interval vanishes.

Mathematically, this means that

$$\int_0^T x_i(t)\, x_j(t)\, dt = 0. \quad for\ i \neq j$$

The key element in multiple access is to make the transmitted signals from the different users orthogonal to each other.

3.1.1 Multiple Access Techniques

Multiple Access method allows several terminals connected to the same multi-point transmission medium to transmit over it and to share its capacity. Examples of shared physical media are wireless networks, bus networks, ring networks, hub networks and half-duplex point-to-point links. Multiple Access technique is based on a multiplex method that allows several data streams or signals to share the same communication channel or physical media.

Multiple Access methods address the problem of how many users can share the same spectrum resources in an efficient manner. We distinguish between:

- multiple access within one cell, i.e., a fixed assignment of resources in time or bandwidth to specific users.
- random access, i.e., a dynamic assignment of spectrum resources in time or bandwidth to users, according to their needs.
- frequency reuse, i.e., assignment of spectrum resources considering the location of users and the attenuation of radio signals that travel over sufficiently large distances.

Multiple access is a signal transmission situation in which two or more users wish to simultaneously communicate with each other using the same propagation channel. Multiple access methods/techniques can be categorized under circuit mode and channelisation methods, packet mode methods and duplex methods.

3.1.1.1 Circuit Mode and Channelisation Methods

(a)**Frequency Division Multiple Access** (FDMA)

FDMA is based on frequency-division multiplex (FDM). It gives users an individual allocation of one or several frequency bands, or Channels.

FDMA provides different frequency bands to different users or nodes. In FDMA, the total bandwidth is divided into non-overlapping frequency subband. Each user is allocated a unique frequency subband for the duration of the connection, whether the connection is an active or idle state. Othogonality among transmitted signals from different mobile users is achieved by bandpass filtering in the frequency domain. This type of multiple access support is narrowband, and is not suitable for multimedia communications with various transmission rates. In addition, it incurs a waste of bandwidth when the user is in a dormant state.

An example of FDMA systems were the first-generation (1G) cell-phone systems. A related technique is wave-length division multiple access (WDMA), based on Wavelength division multiplex (WDM), where different users get different colors in fiber-optical communication.

- Wavelength division multiple access (WDMA): is a technology which transmit multiple radio carrier signals on a single channel by using different frequency to carry different signals.
- Orthogonal Frequency Division Multiple Access (OFDMA), based on Orthogonal frequency-division multiplexing (OFDM). Multiple access is achieved in OFDMA by assigning subsets of subcarriers to individual users.

- Single-carrier FDMA (SC-FDMA) can be viewed as linearly-preceded OFDMA (LP-OFDMA), based on single-carrier frequency-domain-equalisation (SC-FDE).

(a) **Time-Division Multiple Access** (TDMA)

TDMA is based on time-division multiplex (TDM). It allows several users to share the same frequency channel by dividing the signal into different time slots. The users transmit in rapid succession, one after the other, each using his alloted time slot. This allows multiple stations to share the same transmission medium (e.g. radio frequency channel) while using only a part of its channel capacity.

TDMA provides different time-slots to different transmitters in a cyclically repetitive frame structure. In a TDMA, the channel time is partitioned into frames. The length of a frame is long enough so that every user in service has an opportunity to transmit once per frame. To achieve this, a TDMA frame is further partitioned into time slots. Users have to transmit in their assigned slots from frame to frame. For example, user 1 may use time slot 1, user 2 time slot 2, etc until the last user. Then it starts all over again. TDMA is used in the digital 2G cellular systems such as Global System for Mobile Communications (GSM), IS-136, Personal Digital Cellular (PDC) and in the Digital Enhanced Cordless Telecommunications (DECT) standard for portable phones. It is also used extensively in satellite systems, and combat-net radio systems. Multi-Frequency Time Division Multiple Access (MF-TDMA) is one of the types of TDMA.

(c) **Code Division Multiple Access** (CDMA), or **Spread Spectrum Multiple Access** (SSMA)

CDMA is a spread spectrum multiple access method. The principle of spread spectrum communications is that the bandwidth of the baseband information-carrying signals from the different users is spread by different signals with a bandwidth much larger than that of the baseband signals. Ideally, the spreading signals used for different users are orthogonal to each other. Thus, at the receiver, the same spreading signal is used as the dispreading signals to coherently extract the baseband signal from the target user, while suppressing the transmissions from any other users. An example of CDMA is the 3G cell phone system.

In Spread Spectrum communication, the bandwidth occupancy of a single transmitted signal is much higher than in systems using conventional modulation methods. This band-spreading is achieved by selecting appropriate transmission waveforms with a wide bandwidth. A very popular method is to multiply the user data signal with a fast code sequence, which mostly is independent of the transmitted data message.

With Code Division Multiple Access (CDMA) multiple users can share the same portion of the radio spectrum but use different codes to distinguish their transmissions.

CDMA Schemes

Various spread-spectrum techniques are:

- Direct-Sequence CDMA (DS-CDMA**),** based on Direct-sequence spread spectrum (DSSS): DS-CDMA is a method which shares spectrum among multiple simultaneous users. Moreover, it can exploit frequency diversity, using a RAKE receiver. However, in a dispersive multipath channel, DS-CDMA with a spread factor N can accommodate N simultaneous users only if highly complex interference cancellation techniques are used.

In practice this is difficult to implement. MC-CDMA can handle N simultaneous users with good bit error rate (BER), using standard receiver techniques.

- Frequency-Hopping CDMA (FH-CDMA), based on Frequency-hopping spread spectrum (FHSS) is a method of transmitting radio signals by rapidly switching a carrier among many frequency channels, using a pseudorandom sequence known to both transmitter and receiver.
- Orthogonal frequency-hopping multiple access (OFHMA).
- Multi-Carrier Code Division Multiple Access (MC-CDMA): is a multiple access scheme used in OFDM-based telecommunication systems, allowing the system to support multiple users at the same time. Its development aimed at improved performance over multipath links.
- Ultra Wide Band (UWB) techniques using very steep pulses at well defined instants. This is sometimes called "Time Hopping". Time hopping technique is a spread spectrum technique in which the carrier is turned on and off by the pseudorandom code sequence. It is usually used in combination with other methods, in which the transmitted pulse occurs in a manner determined by a pseudorandom code which places the pulse in one of several possible positions per frame.

Attributes of CDMA in Cellular Systems

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

- **Soft-handoff:** since every cell uses the same radio frequency band, the only difference between user channels is the spreading code sequences. Therefore, there is no jump from one frequency to another frequency when a user moves between cells. The mobile terminal receives the same signal in one cell as it does in the next, and thus there is no harsh transition from one receiving mode to another. Two or more neighboring base stations can receive the signal of a particular user, because they all use the same channel. Moreover, two base stations can simultaneously transmit to the same user terminals. The mobile (rake) receiver can resolve the two signals separately and combine them. This feature is called soft handoff.

- **Soft capacity or graceful degradation:** in FDMA and TDMA, N channels can be used virtually without interference from other users in the same cell but potential users *N+1, N+2, ...,* are blocked until a channel is released. The capacity of FDMA and TDMA is therefore fixed at N users and the link quality is determined by the frequency reuse pattern. In theory, it does not matter whether the spectrum is divided into frequencies, time slots, or codes, the capacity provided from these three multiple access schemes is the same. However, in CDMA, all the users in all cells share one radio channel and are separated by codes. Therefore, an additional user may be added by sacrificing somewhat the link quality, with the effect that voice quality is just slightly degraded compared to that of the normal N-channel cell. Thus, degradation of performance with an increasing number of simultaneous users is "graceful" in CDMA systems, versus the hard limits placed on FDMA and TDMA systems.

- **Multipath-tolerance:** spread spectrum techniques are effective in combating the frequency selective fading that takes place in multipath channels. The underlying principle is that when a signal is spread over a wide bandwidth, a frequency selective fade will corrupt only a small portion of the signal's power spectrum, while passing the remaining spectrum unblemished. As a result, upon dispreading there is a better probability that the signal can be recovered correctly. For an unspread signal whose spectral density happens to be misplaced in a deep fade, an unrecoverable signal at the receiver is virtually assured. To optimally combine signals received over various delayed paths, a rake receiver can be used.

- **No channel equalisation needed:** when the transmission rate is much higher than 10 kbps in both FDMA and TDMA; an equaliser is needed for reducing the intersymbol interference caused by time delay spread. This is because when the bit period becomes smaller than about ten times the time delay spread, intersymbol interference becomes significant. However, in CDMA a correlation is needed at minimum. To achieve good performance a rake receiver is needed to combat delay spread.

- **Privacy:** an important requirement of spreading signals is that they are "noise-like", or pseudorandom. Dispreading the signal requires knowledge of the user's code, and for a binary code with spreading factor N there exist 2^N possible random sequences. In military systems these codes are kept secret, so it is very difficult for an unauthorised attacker to tap into or transmit on another users channel. Often it is even difficult to detect the presence of a spread-spectrum signal because it is below the noise that is present in the transmit bandwidth.
  Note that in cellular systems, the codes are fully described in publicly available standards. In digital systems, security against eavesdropping (confidentiality) is obtained through encryption. This is a highly desirable alternative to the analog FDMA cellular phone system in wide use today, where with an inexpensive scanner one can tune in to the private conversations of unwary neighbours.

Disadvantages of CDMA

There are, of course, a number of disadvantages associated with CDMA; two of the most severe are the problem of "self-interference," and the related problem of the "near-far" effect.

- Self-interference arises from the presence of delayed replicas of signal due to multipath. The delays cause the spreading sequences of the different users to lose their orthogonality, as by design they are orthogonal only at zero phase offset. Hence in dispreading a given user's waveform, nonzero contributions to that user's signal arise from the transmissions of the other users in the network. This is distinct from both TDMA and FDMA, wherein for reasonable time or frequency guardbands, respectively, orthogonality of the received signals can be preserved.
- The near-far problem arises from the fact that signals closer to the receiver are received with smaller attenuation than signals located further away. Therefore the strong signal from the nearby transmitter will mask the weak signal from the remote transmitter. In TDMA and FDMA, this is not a problem since mutual interference can be filtered. In CDMA, however, the near-far effect combined with imperfect orthogonality between codes (e.g. due to different time sifts), leads to substantial interference. Accurate and fast power control appears essential to ensure reliable operation of multi-user DS-CDMA systems.

Applications of spread spectrum are in:

- Military System: this is the oldest known application. It is popular for security reasons
- Positioning Systems: high bandwidth signals allow accurate measurements of propagation delays. This is used to estimate the distance of a transmitter
- Cellular Radio: it is mainly used to combat dispersion and to provide multiple access.

- Wireless LANs: if conventional modulation would be used, frequency management for many coexisting links can be very difficult. Moreover, narrowband transmission could be impaired by deep local fades.

(a) **Space Division Multiple Access** (SDMA)

SDMA enables creating parallel spatial pipes next to higher capacity pipes through spatial multiplexing and/or diversity, by which it is able to offer superior performance in radio multiple access communication systems. In traditional mobile cellular network systems, the base station has no information on the position of the mobile units within the cell and radiates the signal in all directions within the cell in order to provide radio coverage. These results in wasting power on transmissions when there are no mobile units to reach, in addition to causing interference for adjacent cells using the same frequency, so called co-channel cells.

Likewise, in reception, the antenna receives signals coming from all directions including noise and interference signals. By using smart antenna technology and by leveraging the spatial location of mobile units within the cell, space-division multiple access techniques offer attractive performance enhancements. The radiation pattern of the base station, both in transmission and reception is adapted to each user to obtain highest gain in the direction of that user. This is often done using phased array techniques.

3.1.1.2Packet Mode Methods

Packet mode methods are typically also based on time-domain multiplexing, but not in a cyclically repetitive frame structure, and therefore it is not considered as TDM (Time Division Multiplexing) or TDMA (Time Division Multiple Access). Due to its random character, it can be categorised as statistical multiplexing methods, making it possible to provide dynamic bandwidth allocation.

(b) **Contention based random multiple access methods**

In packet mode communication networks, contention is a media access method that is used to share a broadcast medium.

i. **Aloha**: any terminal is allowed to transmit without considering whether channel is idle or busy. If packet is received correctly, the base station transmits an acknowledgement. If no acknowledgement is received by the mobile, it retransmits the packet after waiting a random time. The mode of random access in which users can transmit at anytime is called pure aloha. In pure aloha system, where the packet length is a fixed constant, the vulnerability period (i.e. the maximum interval over which two packets can overlap and destroy each other) is two slot times i.e. the time interval required to transmit two packets.

ii. **Slotted Aloha:** a version in which users are restricted to transmit only from the instant corresponding to the slot boundary. Any slot is available for utilization without regards to prior usage

iii. **Multiple Access with Collision Avoidance (MACA):** is a slotted media access control protocol used in wireless LAN data transmission to avoid collisions caused by the hidden station problem and to simplify exposed station problem. The basic idea of MACA is that, a wireless network node makes an announcement before it sends the data frame to inform other nodes to keep silent. When a node wants to transmit, it sends a signal called Request-To-Send (RTS) with the length of the data frame to send. If the receiver allows the transmission, it replies the sender with a signal called Clear-To-Send (CTS) with the length of the frame it is about to receive. Meanwhile, a node that hears RTS should remain silent to avoid conflict with CTS; a node that hears CTS should keep silent until the data transmission is complete.

iv. **Multiple Access with Collision Avoidance for Wireless (MACAW):** if WLAN data transmission collisions occur after data transmission completion in MACA, then the MACA for Wireless (MACAW) is introduced to extend the function of MACA. It requires nodes sending acknowledgements after each successful frame transmission, as well as the additional function of Carrier sense.

v. **Carrier Sense Multiple Access (CSMA):** is a probabilistic Media Access Control (MAC) protocol in which a node verifies the absence of other traffic before transmitting on a shared transmission medium, such as an electrical bus, or a band of the electromagnetic spectrum. In Carrier Sense Multiple Access (CSMA), users listen before transmission and the listening is referred to as sensing the channel. A station wishing to transmit has to first listen to the channel for a predetermined amount of time so as to check for any activity on the channel. If the channel is sensed "idle" then the station is permitted to transmit. If the channel is sensed as "busy" the station has to defer its transmission.

Types of CSMA

- **1-persistent CSMA**: when the sender (station) is ready to transmit data, it checks if the physical medium is busy. If so, it senses the medium continually until it becomes idle, and then it transmits a piece of data (a frame). In case of a collision, the sender waits for a random period of time and attempts to transmit again.

- **P-persistent CSMA**: this protocol is a generalisation of 1-persistent CSMA. When the sender is ready to send data, it checks continually if the medium is busy. If the medium becomes idle, the sender transmits a frame with a probability *p*. In case the transmission did not happen (the probability of this event is *1-p*) the sender waits until the next available time slot and transmits again with the same probability *p*. This process repeats until the frame is sent or some other sender starts transmitting. In the latter case the sender waits a random period of time, checks the channel, and if it is idle, transmits with a probability *p*, and so on.

- **Non-persistent CSMA:** when the sender is ready to send data, it checks if the medium is busy. If so, it waits for a random amount of time and checks again. When the medium becomes idle, the sender starts transmitting. If collision occurs, the sender waits for a random amount of time, and checks the medium, repeating the process.

- **Carrier Sense Multiple Access with Collision Detection (CSMA/CD) -** suitable for wired networks. CSMA/CD is a modification of pure Carrier Sense Multiple Access (CSMA). Collision detection is used to improve CSMA performance by terminating transmission as soon as a collision is detected, and reducing the probability of a second collision on retry.

- **Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) -** suitable for wireless networks. In CSMA/CA (Local Talk), once the channel is clear, a station sends a signal telling all other stations not to transmit, and then sends its packet. In Ethernet 802.3, the station continues to wait for a time, and checks to see if the channel is still free. If it is free, the station transmits, and waits for an acknowledgment signal that the packet was received.

- **Distributed Coordination Function (DCF):** requires a station wishing to transmit to listen for the channel status for a DCF Interframe Space (DIFS) interval. If the channel is found busy during the DIFS interval, the station defers its transmission. In a network where a number of stations contend for the multi-access channel, if multiple stations sense the channel busy and defer their access, they will also virtually simultaneously find that the channel is released and then try to seize the channel. As a result, collisions may occur. In order to avoid such collisions, DCF also specifies random back-off, which forces a station to defer its access to the channel for an extra period. DCF also has an optional virtual carrier sense mechanism that exchanges short Request-to-send (RTS) and Clear-to-send (CTS) frames between source and destination stations during the intervals between the data frame transmissions. DCF includes a positive acknowledgement scheme, which means that if a frame is successfully received by the destination it is addressed to, the destination needs to send an ACK frame to notify the source of the successful reception.

- **Point Coordination Function (PCF):** is a Media Access Control (MAC) technique used in wireless networks which relies on a central node, often an Access Point (AP), to communicate with a node listening to see if the airwaves are free (i.e., all other nodes are not communicating). Since most APs have logical bus topologies (they are shared circuits) only one message can be processed at one time (it is a contention based system), and thus a media access control technique is required.

- **Carrier Sense Multiple Access with Collision Avoidance and Resolution Using Priorities (CSMA/CARP):** instead of detecting network collisions, CSMA/CARP attempts to avoid collisions by using a system of transmission priorities. When a station wants to transmit on a CSMA/CARP network it first listens for network traffic and if the medium is clear instead of immediately transmitting as a station would in CSMA/CD, it waits a predefined amount of time. This waiting period is called the interframe spacing (IFS) and it varies by the type of data being transmitted. High priority data will transmit almost immediately whereas lower priority data such as polling will have a longer IFS. This system allows CSMA/CARP to avoid many collisions that would occur if it was not used. In addition to having different IFS per priority, a station in a CSMA/CARP network will add a "random back-off" to its waiting period, to reduce the collision probability between stations that have to transmit packets in the same priority.

- **Carrier Sense Multiple Access/Bitwise Arbitration (CSMA/BA):** based on constructive interference (Controller Area Network-bus) .When two nodes on CAN attempt to transmit at the same time, a non-destructive arbitration technique guarantees the messages are sent in order of priority.

a. **Token passing:** is where a signal called a token is passed around between nodes that authorise the node to communicate. Token passing schemes are technique in which only the system which has the token can communicate. The token is a control mechanism which gives authority to the system to communicate or use the resources of that network. Once the communication is over, the token is passed to the next candidate in a sequential manner. The most well-known examples are token ring and token bus.

b. **Token ring:** token ring local area network (LAN) technology is a local area network protocol which resides at the data link layer (DLL) of the OSI model. It uses a special three-byte frame called a token that travels around the ring. Token ring frames travel completely around the loop.

c. **Token bus:** is a network implementing the token ring protocol over a "virtual ring" on a coaxial cable. A token is passed around the network nodes and only the node possessing the token may transmit. If a node doesn't have anything to send, the token is passed on to the next node on the virtual ring. Each node must know the address of its neighbour in the ring, so a special protocol is needed to notify the other nodes of connections to, and disconnections from, the ring.

d. **Polling:** refers to actively sampling the status of an external device by a client program as a synchronous activity. Polling also refers to the situation where a device is repeatedly checked for readiness, and if it is not the computer returns to a different task.

e. **Resource Reservation (scheduled) Packet-Mode Protocols:** is a transport layer protocol designed to reserve resources across a network for an integrated internet services.

i. **Dynamic Time Division Multiple Access** (Dynamic TDMA) is a scheduling algorithm which dynamically reserves a variable number of time slots in each frame to variable bit-rate data streams, based on the traffic demand of each data stream. Dynamic TDMA is used in HIPERLAN/2 broadband radio access network, IEEE 802.16a WiMax , Bluetooth , etc.

ii. **Packet Reservation Multiple Access (PRMA)** is a packet-based TDMA concept where the users contend for the time slots. In situations where the system is not near capacity, a user can reserve a time slot for future uses.

iii. **Reservation ALOHA (R-ALOHA)** is a multiple access method for wireless transmission which allows uncoordinated users to share a common transmission resource. Reservation ALOHA (and its parent scheme, Slotted ALOHA) is a schema or rule set for the division of transmission resources over fixed time increments also known as slots. If this rule set or schema is properly followed, the bandwidth users are allowed to cooperatively utilise a shared transmission resource (i.e. the allocation of transmission time).

3.1.1.3 Duplex Methods

A **duplex** communication system is a system composed of two connected parties or devices that can communicate with one another in both directions. Duplex systems are employed in many communications networks, either to allow for “two-way street" communication between two connected parties or to provide a "reverse path" for the monitoring and remote adjustment of equipment in the field. Where these methods are used for dividing forward and reverse communication channels, they are known as duplex methods. Examples are:

(i)Time Division Duplex (TDD) is the application of time-division multiplexing to separate outward and return signals. It emulates full-duplex communication over a half-duplex communication link. Time-division duplex has a strong advantage in the case where there is asymmetry of the uplink and downlink data rates. As the amount of uplink data increases, more communication capacity can be dynamically allocated, and as the traffic load becomes lighter, capacity can be taken away. The same applies in the downlink direction.

(ii)Frequency Division Duplex (FDD) means that the transmitter and receiver operate at different carrier frequencies. The term is frequently used in ham radio operation, where an operator is attempting to contact a repeater station. The station must be able to send and receive a transmission at the same time, and does so by slightly altering the frequency at which it sends and receives. This mode of operation is referred to as duplex mode or offset mode.

- Uplink and downlink sub-bands are said to be separated by the frequency offset. Frequency- division duplex can be efficient in the case of symmetric traffic.
- Frequency-division duplex makes radio planning easier and more efficient, since base stations do not "hear" each other (as they transmit and receive in different sub-bands) and therefore will normally not interfere with each other.

3.1.2Comparison of Multiple Access Techniques

Modulation: TDMA and FDMA depend on the choice of a modulation scheme to maximize spectral efficiency. To achieve a higher throughput in the same bandwidth, a higher order modulation scheme must be used. With CDMA, the simple method of BPSK modulation is required, although, for practical symmetry considerations, QPSK is often used. In fact, the choice of modulation strategy and the use of SDMA are independent.

Forward Error-Correction (FEC) Coding: all multiple-access techniques are affected by the distortions offered by the wireless channel. With FDMA and TDMA, if the same basic throughput is to be maintained, a higher transmission rate and a greater bandwidth is required as a result of the redundancy introduced by FEC coding. This is the classic trade-off between bandwidth and power efficiency. With CDMA, FEC coding can be added without increasing the system bandwidth or harming the processing gain. The inclusion of FEC is transparent to SDMA. If transmit diversity is implemented, then there can be increased bandwidth with SDMA.

Source Coding: the use of source coding improves the bandwidth efficiency of all multiple-access techniques. However, CDMA is in a position to take greater advantage of voice activation than other techniques, since its bandwidth efficiency is determined by average interference.

Diversity: multiple transmitters or receivers or both are required to obtain diversity with FDMA, which is an added hardware expense. The same is applied to TDMA, except when it is used as part of a TDMA/FDMA hybrid. In that case, frequency-hopped TDMA can provide some diversity advantage. The large bandwidth of CDMA naturally provides some frequency diversity, and this can be used advantageously with a rake receiver (A rake receiver is a radio receiver designed to counter the effects of multipath fading). The implementation cost of a rake receiver is less than the dual-receiver cost of an FDMA system with frequency diversity.

User Terminal Complexity: with the progression from FDMA through TDMA to CDMA comes an evolution of terminal complexity. SDM systems introduce a different and additional form of complexity that is related to the antennas which is not present in any of the other systems.

Handover: with their single-receiver terminals, both FDMA and TDMA are somehow handicapped when they must switch between base stations at a cell boundary. With CDMA, since the frequencies are used in adjacent cells, it is easier to implement a "dual receiver" and provide a soft handover capability.

System Complexity: with an FDMA system, users can operate quite independently. With TDMA, the level of cooperation among users must increase to share slots. With CDMA, the system must delegate spreading codes, power control information, and synchronisation information.

Multiple-Access Interference (MAI): because FDMA and TDMA tend to be limited by worst-case interference, interference is often limited in the system planning stage by the fixed assignment of frequency groups to specific cells. With CDMA, the same bandwidth is used everywhere, and performance is limited by average interference levels. However, CDMA relies heavily on accurate power control to eliminate the near-far problem.

Fading-Channel Sensitivity: FDMA systems are typically narrowband and therefore suffer from flat fading. If the fading is not severe, then simple channel estimation and forward error correction can often compensate for its effects. TDMA systems are typically medium-bandwidth solutions. As a result of this, they observe some frequency selectivity. This requires the implementation of an equaliser. In fact, the implementation of a robust tracking equaliser in wireless channels is of utmost importance. CDMA systems face frequency-selective channels because of their large bandwidth, but take advantage of this natural diversity with a RAKE receiver.

Bandwidth Efficiency: for single-cell systems, FDMA and TDMA systems are generally more bandwidth efficient than CDMA systems, because they do not have to cope with multiple-access interference (MAI). However, once their frequency plan is made and the modulation selected, the maximum throughput is fixed. CDMA holds an advantage because it can reuse frequencies everywhere, while FDMA and TDMA have much lower frequency reuse rates because they are limited by peak interference levels. CDMA can often add a user at the expense of a small degradation of existing users.

Synchronisation: wireless system using FDMA, TDMA and CDMA show a progression in synchronisation resolution and a corresponding progression in complexity. The main concern of FDMA is symbol timing. In fact, TDMA terminals must contend with chip timing.

Flexibility: FDMA is the least flexible of the techniques. Once the service is designed, any change requires a redesign. With TDMA, higher data rates can be provided by assigning more slots per user, usually with very little change to the hardware. With CDMA, different data rates can be provided by trading off the spreading rate (processing gain), making it very flexible. Out of these techniques Space Division Multiple Access (SDMA) is transparent.

Voice and Data Integration: the comments regarding flexibility also apply to the integration of voice and data over the same terminal. With TDMA, it is possible as well to make use of periods of voice inactivity to transmit data, thus making the system more efficient. CDMA can easily integrate voice and data, but usually it leads to multicode transmissions, which may reduce the efficiency of the user-terminal power amplifier.

Evolution: evolving from a small system to a large system is easiest with the FDMA approach. We can easily start with a single-user system and remain relatively efficient at each step. With TDMA, start-up efficiency is related to the transmission rate; the system can evolve easily through the addition of more TDMA channels using an FDMA overlay. With CDMA, there is a large start-up cost, because a large bandwidth to serve perhaps only a few initial user terminals is needed.

Table 6.1:Comparison of Different Multiple Access Strategies

Source: Sharma S. (2006)

Parameters of comparison	**FDMA**	**TDMA**	**CDMA**	**SDMA**
Modulation	Relies on bandwidth efficient modulation	Relies on bandwidth efficient modulation	Simple modulation	Transparent
Forward error correction	Increases power efficiency at expense of bandwidth efficiency	Increases power efficiency at expense of bandwidth efficiency	Can be implemented without affecting bandwidth efficiency	Transparent
Source coding	Improves efficiency	Improves efficiency	Improves efficiency voice activation advantage	Transparent
Diversity	Requires multiple transmitters or receivers	Requires multiple transmitters or receivers can be frequency hopped	Includes frequency diversity when implemented with a RAKE receiver.	Single antenna reduces space diversity, orthogonal coding improves diversity with multiple transmit antennas
User terminal complexity	Simple	Medium complexity	More complex	Requires smart antennas
Handover	Hard	Hard	Soft	Potentially soft
System complexity	Large number of simple components	Reduced number of channel unit	Large number of complex interacting	Additional complexity related to antennas

			components	
Multiple-access interference	Limited by system planning	Limited by system planning	Dynamic power control	Limited by resolution of antennas
Fading	Flat-fading no diversity simple to track	May need frequency-selective and equalizer	Frequency-selective diversity via RAKE receiver	Reduced multipath
Bandwidth efficiency	Hard limits based on modulation and channel spacing	Hard limits based on modulation and channel spacing	Soft limits	Depends on antenna resolution
Synchronisation	Low resolution	Mid-resolution	High resolution	Requires terminal location
Flexibility	Fixed data rate	Data rate variable in discrete steps	Can provide a variety of data rates without affecting signal in space	Transparent
Voice and data integration	Possible, but may require revisions to system	Straight forward using multiple slots	Multicode transmission , which may decrease efficiency of mobile terminal	Transparent
Evolution	Bandwidth to fit application	Requires medium initial bandwidth	Requires large initial bandwidth	Flexible, can be added as far as it does not affect mobile

3.1.3Application Areas of Multiple Access

Local and metropolitan area networks : in local area networks (LANs) and metropolitan area networks (MANs), multiple access methods enable bus networks, ring networks, hubbed networks, wireless networks and half duplex point-to-point communication, but are not required in full duplex point-to-point serial lines between network switches and routers, or in switched networks (logical star topology). The most common multiple access method is CSMA/CD, which is used in Ethernet. Although today's Ethernet installations typically are switched, CSMA/CD is utilised to achieve compatibility with hubs.

Satellite communications: in satellite communications, multiple access is the capability of a communications satellite to function as a portion of a communications link between more than one pair of satellite terminals concurrently. Three types of multiple access presently used with communications satellites are code-division, frequency-division, and time-division multiple access.

Switching centers: in telecommunication switching centers, multiple access is the connection of a user to two or more switching centers by separate access lines using a single message routing indicator or telephone number.

Self-Assessment Exercise

i. What is Multiple Access?
ii. Define CSMA
iii. Discuss the types of CSMA

4.0 Conclusion

In this unit, different multiple access techniques were discussed under circuit and packet mode, channelisation and duplex methods. Also, areas where multiple access can be applied and the comparison of multiple access techniques were discussed.

5.0 Summary

In this unit, you have learnt that:

- Multiple Access is a signal transmission situation in which two or more users wish to simultaneously communicate with each other using the same propagation channel.
- Multiple Access can be categorised under:
- Circuit mode and channelisation methods: FDMA, TDMA, CDMA, SDMA.
- Packet mode
- Duplex Method
- The basic multiple access techniques are FDMA, TDMA, CDMA, SDMA, and PMMA
- FDMA is based on frequency division multiplex. It gives users an individual allocation of one or several frequency bands, or channels. Each user is allocated a unique frequency subband for the duration of the connection, whether the connection is an active or idle state.
- TDMA is based on time-division multiplex (TDM). It allows several users to share the same frequency channel by dividing the signal into different time slots. The users transmit in rapid succession, one after the other, each using his own time slot.

- CDMA is a spread spectrum multiple access method. The principle of spread spectrum communications is that the bandwidth of the baseband information-carrying signals from the different users is spread by different signals with a bandwidth much larger than that of the baseband signals.

- SDMA enables creating parallel spatial pipes next to higher capacity pipes through spatial multiplexing and/or diversity, by which it is able to offer superior performance in radio multiple access communication systems

6.0 Self-Assessment Exercise

i. Discuss on any three types of multiple access techniques.
ii. Write short note on the following: OFDMA, Dynamic TDMA and Aloha.
iii. Discuss the concept of CDMA techniques.
iv. Define and explain TDMA systems.

7.0 References/Further Reading

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