EC8395 COMMUNICATION ENGINEERING

UNIT I ANALOG MODULATION

TOPIC 1.1 AMPLITUDE MODULATION

1. Carrier signal in modulation technique is _______ signal.
a) High frequency
b) Low frequency
c) High amplitude
d) Low amplitude

Answer: a
Explanation: Carrier signal in modulation technique is a high frequency signal. In amplitude modulation, the amplitude of a high frequency carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal.

2. Modulation index of an AM signal is ratio of __________ to the _______
a) Peak carrier amplitude, Peak message signal amplitude
b) Peak message signal amplitude, Peak carrier amplitude
c) Carrier signal frequency, Message signal frequency
d) Message signal frequency, Carrier signal frequency

Answer: b
Explanation: The modulation index k of an AM signal is defined as the ratio of the peak message signal amplitude to the peak carrier amplitude. The modulation index is often expressed as a percentage. It is also called percentage modulation.

3. If the peak message signal amplitude is half the peak amplitude of the carrier signal, the signal is _____ modulated.
a) 100%
b) 2%
c) 50%
d) 70%

Answer: c
Explanation: The modulation is also expressed in percentage. It is also called percentage modulation. The signal is said to be 50% modulated if the peak message signal amplitude is half the peak amplitude of the carrier signal.

4. A percentage of modulation greater than ___________ will distort the message signal.
a) 10%
b) 25%
c) 50%
d) 100%

Answer: d
Explanation: A percentage of modulation greater than 100% will distort the message signal if detected by an envelope detector. In this case the lower excursion of the signal will drive the carrier amplitude below zero, making it negative (and hence changing its phase).

5. The RF bandwidth of AM is ____________ the maximum frequency contained in the modulating message signal.
a) Equal
b) Two times
c) Four times
d) Ten times

Answer: b
Explanation: The RF bandwidth of an AM

signal is equal to $B_{AM}=2f_m$. It is double the maximum frequency contained in the modulating message signal. AM spectrum consists of an impulse at the carrier frequency and two sidebands which replicate the message spectrum.

6. Single sideband AM systems occupy same bandwidth as of conventional AM systems.
a) True
b) False

Answer: b
Explanation: Single sideband (SSB) AM systems transmit only one of the sidebands (either upper or lower) about the carrier. Hence, they occupy only half the bandwidth of conventional AM systems.

7. How is the performance of SSB AM systems in fading channels?
a) Poor
b) Best
c) Good
d) Average

Answer: a
Explanation: SSB systems have the advantage of being very bandwidth efficient. But their performance in fading channels is very poor. For proper detection, the frequency of the oscillator at the product detector mixer in the receiver must be same as that of the incoming carrier frequency.

8. Which of the following is a disadvantage of tone-in-band SSB system?
a) High bandwidth
b) Bad adjacent channel protection
c) Effects of multipath
d) Generation and reception of signal is complicated

Answer: d
Explanation: Tone-in-band SSB systems has the advantage of maintaining the low bandwidth property of the SSB signals, while at the same time providing good adjacent channel protection. The tone in band system employs feedforward automatic gain and frequency control to mitigate the effects of multipath induced fading.

9. FFSR in AM systems stands for ________
a) Feedforward signal regeneration
b) Feedbackward signal regeneration
c) Feedbackward system restoration
d) Feedforward system restoration

Answer: a
Explanation: FFSR stands for Feedforward signal regeneration. If the pilot tone and the information bearing signal undergo correlated fading, it is possible at the receiver to counteract the effects of fading through signal processing based on tracking of pilot tone. This process is called FFSR.

10. AM demodulation technique can be divided into _____ and _____ demodulation.
a) Direct, indirect
b) Slope detector, zero crossing
c) Coherent, noncoherent
d) Quadrature detection, coherent detection

Answer: c
Explanation: AM demodulation techniques may be broadly divide into two main categories. They are called coherent and noncoherent demodulation. They are differentiated by the knowledge of transmitted carrier frequency and phase at the receiver.

11. Non coherent detection requires the knowledge of transmitted carrier frequency and phase at the receiver.
a) True
b) False

Answer: b
Explanation: Non coherent detection does require the knowledge of phase information. However, coherent detection requires knowledge of the transmitted carrier frequency and phase at the receiver.

12. A product detector in AM systems is also called ___________
a) Envelope detector
b) Differentiator
c) Integrator
d) Phase detector

Answer: d
Explanation: A product detector is also called a phase detector. It forms a coherent demodulator for AM signals. It is a down converter circuit which converts the input bandpass signal to a baseband signal.

13. AM system use only product detector for demodulation. They never use envelope detectors.
a) True
b) False

Answer: b
Explanation: AM systems can use either product detector or envelope detector for demodulation. As a rule, envelope detectors are useful when input signal power is at least 10dB greater than noise power, whereas product detectors are able to process the AM signals with input signal to noise ratios well below 0 dB.

TOPIC 1.2 DSBSC

1. LCD uses ________
a) sematic crystals
b) twisted nematic crystals
c) nematic crystals
d) cholesteric crystals

Answer: b
Explanation: LCD uses liquid crystal display. It uses twisted nematic crystals which are a type of liquid crystal, consisting of a substance called the nematic. The nematic liquid crystal is placed between two plates of polarized glass.

2. Which of the following stage is present in FM receiver but not in AM receiver?
a) Amplitude limiter
b) Demodulator
c) AM amplifier
d) Mixer

Answer: a
Explanation: Amplitude Limiter circuit is used in FM receiver to remove the noise or any variation in amplitude present in the received signal. Thus, the output of the amplitude limiter has a constant amplitude. So it is only used in frequency modulation and not in amplitude modulation.

3. Function of duplexer in a RADAR is to permit the use of same antenna for transmission and reception.
a) True
b) False

Answer: a
Explanation: A duplexer is being an electronic unit, allows bi-directional communication over the same path. The transmitter and receiver can communicate simultaneously. In radar, the duplexer isolates the receiver from the transmitter while allowing them to share a common antenna.

4. Single Sideband Modulation (SSB) is generally reserved for point-to-point communication.
a) True
b) False

Answer: a
Explanation: A point-to-point communication refers to bidirectional communication between only one transmitter and one receiver. In SSB-SC modulation technique, the carrier is suppressed and only one of the two side-bands are transmitted. Thus, it reduces power consumption and lessens bandwidth. Thus, it is preferred for point-to-point communication.

5. For an AM transmitter, class C amplifier can be used after the modulation stage.
a) True
b) False

Answer: b
Explanation: In an AM transmitter, the required transmission power is obtained from class C amplifier, as it is a power amplifier, for low-level or high-level modulation. So it is not used after the modulation stage.

6. For which of the modulated system, the linear amplified modulated stage is used?
a) low level amplitude modulated system
b) high level amplitude modulated system
c) high level frequency modulated system
d) low level frequency modulated system

Answer: a
Explanation: In low-level modulation, the generation of amplitude modulated signal takes place at low power levels. The generated AM signal is then amplified using a chain of linear amplifiers, which are required to avoid waveform distortion. Thus, linear amplified modulated stage is used in low level amplitude modulated system.

7. When noise is passed through a narrow band filter, the output of filter should be?
a) triangular
b) square
c) parabolic
d) sinusoidal

Answer: d
Explanation: Narrow band filter is used to isolate a narrow band of frequencies from a wider bandwidth signal. It is a combination of band pass and band reject filter. When noise gets passed through it, the output of it should be sinusoidal.

8. A narrow band noise can exist in __________
a) AM only
b) PCM only
c) FM only
d) AM and FM both

Answer: d
Explanation: Narrow band filter is used to isolate a narrow band of frequencies from a wider bandwidth signal. It is a combination of band pass and band reject filter. So it can be used in both AM and FM to pass a band of frequencies or to attenuate a band of frequencies.

9. The upper and lower sideband frequencies for 5KHz amplitude modulation with a 30KHz carrier frequency will be?
a) 35KHz and 25KHz
b) 34KHz and 24KHz
c) 25KHz and 35KHz
d) 0.35KHz and 0.25KHz

Answer: a
Explanation: Upper sideband frequency will be (30 + 5) = 35 KHz and Lower sideband frequency will be (30 – 5) = 25 KHz.

10. Phase array radar can track many targets together.
a) True
b) False

Answer: a
Explanation: A phased array radar is an array of radiating elements, with each having a phase-shifter. The phase of the signal being emitted from the radiating element is changed to produce beams, thereby producing constructive or destructive interference for steering the beams in the required direction. Thus, it can track many targets together.

TOPIC 1.3 SSBSC

1. A duplex arrangement use separate frequencies for transmission.
a) True
b) False

Answer: a
Explanation: In duplex communication, two-way interaction is favourable simultaneously. Thus, a cordless telephone is duplex which uses separate frequencies for transmission in base and portable units.

2. VSB modulation is used in televisions because it avoids phase distortion at low frequencies.
a) True
b) False

Answer: b
Explanation: Vestigial Sideband Modulation (VSB) is a type of amplitude modulation in which the carrier and only one sideband is completely transmitted and the other sideband is partly transmitted. Thus, television production is done using VSB modulation as it reduces bandwidth to half.

3. A cordless telephone that uses separate frequencies for transmission in base and portable units is called _________
a) half duplex
b) duplex
c) simplex
d) one-way communication

Answer: b
Explanation: In duplex communication, two-way interaction is favourable simultaneously. Thus, a cordless telephone is duplex which uses separate frequencies for transmission in base and portable units.

4. Which polarization is used to reduce the depolarization effect on received waves?
a) Circular polarization
b) Linear polarization
c) Atomic polarization
d) Dipolar polarization

Answer: a
Explanation: In circular polarization at each point the electric field of electromagnetic wave has a constant magnitude but its direction changes as it rotates with time at a steady rate, in a plane perpendicular to the direction of propagation of wave. It is used to reduce depolarization effect on received waves.

5. Circular polarization involves critical alignment between transmitting and receiving antenna.
a) True
b) False

Answer: b
Explanation: In circular polarization at each point the electric field of the electromagnetic wave has a constant magnitude but its direction changes as it rotates with time at a steady rate, in a plane perpendicular to the direction of propagation of the wave. It is used to reduce depolarization effect on received waves. It does not involve alignment between transmitting and receiving antenna.

6. It is only the reflected color that decided the color of an object.
a) True
b) False

Answer: b
Explanation: Color of any object is decided by the reflected color for opaque object and wavelength transmitted through it for transparent object, while both reflector color and wavelength transmitted are considered for a translucent object.

7. What do you understand by the term "carrier" in modulation?
a) voltage to be transmitted
b) resultant wave
c) voltage for which amplitude, phase or frequency can be varied
d) voltage for which amplitude, phase or frequency remains constant

Answer: c
Explanation: Carrier wave is the wave with frequency higher than the message signal,

whose certain characteristics like amplitude, phase or frequency are varied with respect to the instantaneous amplitude of the message signal. Thus forming the modulated wave which is the wave to be transmitted.

8. Carrier wave in modulation is a resultant wave.
a) True
b) False

Answer: b
Explanation: Carrier wave is the wave with frequency higher than the message signal, whose certain characteristics like amplitude, phase or frequency are varied with respect to the instantaneous amplitude of the message signal. Thus forming the modulated wave which is the wave to be transmitted.

9. For a low level AM system, amplifier modulated stage must have __________
a) harmonic devices
b) linear devices
c) non-linear devices
d) class A amplifiers

Answer: b
Explanation: In low-level modulation, the generation of amplitude modulated signal takes place at low power levels. The generated AM signal is then amplified using a chain of linear amplifiers, which are required to avoid waveform distortion. Thus, linear devices are used in low level amplitude modulated system.

TOPIC 1.4 VSB

1. Quantization noise occurs in _________
a) Frequency Division Multiplexing
b) Time Division Multiplexing
c) Delta Modulation
d) Amplitude Modulation

Answer: d
Explanation: Quantisation is the process through which a range of continuous analog values are quantized or rounded off to a single value, thereby forming samples of a discrete digital signal. Quantisation Error occurs when there is a difference between an input value and it's quantized value. Quantisation occurs when an analog signal is converted into it's digital form, thus it occurs in Pulse Code modulation (PCM).

2. Which is the greatest disadvantage of Pulse Code Modulation?
a) highly prone to noise
b) cannot travel long distances
c) its inability to handle analog signals
d) large bandwidth is required for it

Answer: d
Explanation: Pulse code modulation (PCM) is a digital form of communication. For demodulation of PCM, it is necessary to convert it into PAM. Quantization noise occurs in PCM only. Its greatest disadvantage is its requirement for large bandwidth.

3. Inductance and capacitance of a line is 0.8 $^{\mu H}/_{m}$ and 32 $^{pF}/_{m}$. Find Z_0?
a) 158
b) 166
c) 143
d) 127

Answer: a
Explanation:

$$Z_0 = \sqrt{\frac{L}{C}} = \sqrt{\frac{0.8\times10^{-6}}{32\times10^{-12}}} = 158.11$$

4. Pulse communication system that is inherently highly immune to noise is

____
a) PCM
b) PPM
c) PAM
d) PWM

Answer: a
Explanation: Pulse Code Modulation is a

technique in which the amplitude of an analogue signal is converted to a binary value represented as a series of pulses. It is less prone to noise and can travel through long distances without loss of data.

5. What the main advantage of PCM?
a) can travel small distances
b) higher bandwidth
c) lower noise
d) good reception

Answer: c
Explanation: Pulse Code Modulation is a technique in which the amplitude of an analogue signal is converted to a binary value represented as a series of pulses. It is less prone to noise and can travel through long distances without loss of data.

6. In AM pilot carrier, transmission has _________
a) carrier and part of one side band
b) two side bands and a carrier
c) two side bands
d) carrier, one side band and part of other side band

Answer: b
Explanation: In amplitude modulated wave, the transmitted wave has two side bands and a carrier. Thus it's bandwidth is twice the maximum modulating frequency.

7. Quantization noise depends upon both sampling rate and number of quantization levels.
a) True
b) False

Answer: b
Explanation: Quantization noise in pulse code modulation (PCM) depends upon only on number of quantization levels.

8. Which of the following frequency is not transmitted in AM transmission?
a) Upper side band frequency
b) Carrier frequency
c) Lower side band frequency
d) Audio frequency

Answer: d
Explanation: Audio frequency is the frequency that is not transmitted in AM transmission.

9. Companding is used in PCM transmitters to allow amplitude limiting in the receivers.
a) True
b) False

Answer: b
Explanation: Companding is the process through which the signal to noise ratio of a wave is reduced by compressing and expanding the signal. It decreases the number of bits required to record the strongest signal. Companding also improves signal to noise ratio.

10. What is the use of Companding?
a) in PCM transmitters to allow amplitude limiting in the receivers
b) in PCM receiver to overcome impulse noise
c) to overcome quantizing noise in PCM
d) to protect small signals in PCM from quantizing distortion

Answer: d
Explanation: Companding is the process through which the signal to noise ratio of a wave is reduced by compressing and expanding the signal. It decreases the number of bits required to record the strongest signal. Companding also improves signal to noise ratio. It is mainly used to protect small signals in PCM from quantizing distortion.

TOPIC 1.5 PSD MODULATORS AND DEMODULATORS

1. Modern mobile communication systems use analog modulation techniques.

a) True
b) False

Answer: b
Explanation: Modern mobile communication systems use digital modulation techniques. Advancements in VLSI and digital signal processing technology have made digital modulation more cost effective than analog transmission systems.

2. Which of the following is not an advantage of digital modulation?
a) Greater noise immunity
b) Greater security
c) Easier multiplexing
d) Less bandwidth requirement

Answer: d
Explanation: Digital modulation offer many advantages over analog modulation. Some advantages include greater noise immunity and robustness. They provide easier multiplexing of various forms of information and greater security.

3. A desirable modulation scheme provides _________ bit error rates at __________ received signal to noise ratios.
a) Low, low
b) Low, high
c) High, high
d) High, low

Answer: a
Explanation: A desirable modulation scheme provides low bit error rates at low received signal to noise ratios. They perform well in multipath and fading conditions, occupies a minimum bandwidth and is easy and cost effective to implement.

4. The performance of modulation scheme is not measured in terms of __________
a) Power efficiency
b) Bandwidth efficiency
c) Cost and complexity
d) Transmitted power

Answer: d
Explanation: The performance of modulation scheme is often measured in terms of its power efficiency and bandwidth efficiency. Other factors also affect the choice of modulation scheme, such as cost and complexity of the subscriber receiver and modulation which is simple to detect.

5. In digital communication system, in order to increase noise immunity, it is necessary to increase _________
a) Signal power
b) Signal amplitude
c) Signal frequency
d) Signal magnitude

Answer: a
Explanation: In digital communication system, in order to increase noise immunity, it is necessary to increase signal power. However, the amount by which the signal power should be increased to obtain a certain level of fidelity depends on the particular type of modulation employed.

6. Which of the following is the ratio of signal energy per bit to noise power spectral density?
a) Bandwidth efficiency
b) Spectral density
c) Power efficiency
d) Power density

Answer: c
Explanation: Power efficiency is often expressed as the ratio of signal energy per bit to noise power spectral density required at the receiver input for a certain probability of error. Power efficiency is a measure of how favourably the trade-off between fidelity and signal power is made.

7. Increasing the data rate implies the increase in pulse width of digital symbol.
a) True
b) False

Answer: b
Explanation: There is an unavoidable relationship between data rate and bandwidth occupancy. Increasing the data rate implies decreasing the pulse width of a digital symbol, which increases the bandwidth of the signal.

8. Which of the following is the ratio of the throughput data rate per Hertz?
a) Bandwidth efficiency
b) Spectral density
c) Power efficiency
d) Power density

Answer: a
Explanation: Bandwidth efficiency reflects how efficiently the allocated bandwidth is utilized. It is defined as the ratio of throughput data rate per Hertz in a given bandwidth. It describes the ability of a modulation scheme to accommodate data within a limited bandwidth.

9. Which of the following is defined as the range of frequencies over which the signal has a non zero power spectral density?
a) Null to null bandwidth
b) Half power bandwidth
c) 3 dB bandwidth
d) Absolute bandwidth

Answer: d
Explanation: The absolute bandwidth is defined as the range of frequencies over which the signal has a non-zero power spectral density. For symbols represented as rectangular baseband pulses, the PSD profile extends over an infinite range of frequencies, and has an absolute bandwidth of infinity.

10. _______ is equal to width of main spectral lobe.
a) Null to null bandwidth
b) Half power bandwidth
c) 3 dB bandwidth
d) Absolute bandwidth

Answer: a
Explanation: Null to null bandwidth is a simpler and more widely accepted measure of bandwidth. It is equal to the width of main spectral lobe.

11. Half power bandwidth is also called _______
a) Absolute bandwidth
b) Null to null bandwidth
c) 3 dB bandwidth
d) Zero dB bandwidth

Answer: c
Explanation: Half power bandwidth is also called the 3 dB bandwidth. It is defined as the interval between frequencies at which the PSD has dropped to half power, or 3 dB below the peak value.

TOPIC 1.6 ANGLE MODULATION

1. FM is a part of general class of modulation known as ______
a) Angle modulation
b) Phase modulation
c) Amplitude modulation
d) Frequency modulation

Answer: a
Explanation: FM is a part of general class of modulation known as angle modulation. Angle modulation varies a sinusoidal carrier signal in such a way that the angle of the carrier is varied according to the amplitude of the modulating baseband signal.

2. FM is called constant envelope because ______ of carrier wave is kept constant.
a) Frequency
b) Amplitude
c) Phase
d) Angle

Answer: b
Explanation: FM is called the constant

envelope because amplitude of the carrier wave is kept constant. It is duo to the fact that the envelope of the carrier does not change with changes in the modulating signal.

3. Which of the following are two most important classes of angle modulation?
a) Amplitude modulation, frequency modulation
b) Amplitude modulation, phase modulation
c) Frequency modulation, phase modulation
d) Single sideband amplitude modulation, phase modulation

Answer: c
Explanation: The two most important classes of angle modulation are frequency modulation and phase modulation. They provide the ways in which phase of a carrier signal may be varied in accordance with the baseband signal.

4. Frequency modulated signal is regarded as the phase modulated signal in which the modulating wave is differentiated before modulation.
a) True
b) False

Answer: b
Explanation: Frequency modulated signal is regarded as the phase modulated signal in which the modulating wave is integrated before modulation. This means that an FM signal can be generated by first integrating the message signal and then using the result as an input to a phase modulator.

5. Frequency modulation index defines the relationship between the _______ and bandwidth of transmitted signal.
a) Frequency of message signal
b) Amplitude of message signal
c) Amplitude of carrier signal
d) Frequency of carrier signal

Answer: b
Explanation: The frequency modulation index defines the relationship between the message amplitude and the bandwidth of the transmitted signal. If the modulating signal is a low pass signal, maximum bandwidth of the modulating signal is equal to the highest frequency component present in the modulating signal.

6. FM bandwidth is approximated using ________ rule.
a) Carson's
b) Faraday's
c) Maxwell's
d) Armstrong's

Answer: a
Explanation: The approximation of bandwidth is done using Carson's rule. Carson's bandwidth rule defines the approximate bandwidth requirements of communications system components for a carrier signal that is frequency modulated by a continuous or broad spectrum of frequencies rather than a single frequency.

7. Which of the following are two methods for generating FM signal?
a) Coherent method, noncoherent method
b) Product detector, envelope detector
c) Direct method, indirect method
d) Slope detector, Zero crossing detector

Answer: c
Explanation: Direct method and indirect method are the methods used for generating FM signals. These methods are differentiated by the variation of the carrier frequency.

8. In indirect method, the carrier frequency is directly varied in accordance with the input modulating signal.
a) True
b) False

Answer: b
Explanation: The above is the case for direct method. In the indirect method, a narrowband FM signal is generated using a balanced

modulator, and frequency multiplication is used to increase both the frequency deviation and the carrier frequency to the required level.

9. Which of the following is used to vary the frequency of the carrier frequency in accordance with the baseband signal amplitude variations in direct method of FM generation?
a) Integrator
b) Envelope detector
c) Multivibrator
d) Voltage controlled oscillators

Answer: d
Explanation: In direct method, VCOs are used to vary the frequency of the carrier signal in accordance with the baseband signal amplitude variations. These oscillators use devices with reactance that can be varied by the application of a voltage.

10. Frequency demodulator is a frequency to amplitude converter circuit.
a) True
b) False

Answer: a
Explanation: Frequency demodulator produces an output voltage with instantaneous amplitude that is directly proportional to the instantaneous frequency of the input FM signal. Thus, frequency demodulator is a frequency to amplitude converter circuit.

11. Which of the following is not a technique for FM demodulation?
a) Slope detection
b) Zero crossing detection
c) Product detector
d) Phase locked discriminator

Answer: c
Explanation: Various techniques such as slope detection, zero crossing detection, phase locked discrimination and quadrature detection are used to demodulate FM. Product detector is used for demodulating AM signals.

12. Which of the following FM demodulator is sometimes known as pulse averaging discriminator?
a) Slope detection
b) Zero crossing detection
c) Quadrature detection
d) Phase locked discriminator

Answer: b
Explanation: Zero crossing detector is sometimes known as pulse averaging discriminator. The rationale behind this technique is to use the output of the zero crossing detector to generate a pulse train with an average value that is proportional to frequency of the input signal.

13. PLL in FM detection stands for ______
a) Phase locked loop
b) Programmable logic loop
c) Phase locked logic
d) Programmable locked loop

Answer: a
Explanation: PLL stands for phase locked loop. The PLL is a closed loop control system which can track the variations in the received signal phase and frequency.

14. In angle modulation, signal to noise ratio before detection is a function of ______
a) Modulation index
b) Input signal to noise ratio
c) Maximum frequency of the message
d) IF filter bandwidth

Answer: d
Explanation: In angle modulation systems, the signal to noise ratio before detection is the function of the receiver IF filter bandwidth, received carrier power, and received interference. However, signal to noise ratio after detection is a function of maximum

frequency of the message, input signal to noise ratio and modulation index.

15. FM can improve the receiver performance through adjustment of transmitted power.
a) True
b) False

Answer: b
Explanation: FM can improve receiver performance through adjustment of the modulation index at the transmitter, and not the transmitted power. This is not the case in AM since linear modulation techniques do not trade bandwidth for SNR.

TOPIC 1.7 PM AND FM – PSD, MODULATORS AND DEMODULATORS

1. Pre-emphasis is used to amplify __________ frequencies.
a) low
b) high
c) both low and high
d) local oscillator

Answer: b
Explanation: Pre-emphasis is used in frequency modulated transmitters to equalize the drive power of transmitting signal in terms of deviation ratio. It is done at the transmitter. It is used to amplify high frequencies.

2. De-emphasis circuit is used _________
a) before detection
b) after detection
c) before encoding
d) after encoding

Answer: b
Explanation: De-emphasis means attenuation of those frequencies by the amount by which they are boosted. It is done at the receiver end i.e. it is used after detection.

3. Why frequency fogging is used in a carrier system?
a) to reduce noise
b) to reduce cross talk
c) to converge frequencies
d) to reduce distortion

Answer: b
Explanation: The interchanging of the frequencies of carrier channels to accomplish specific purposes. It is used to prevent feedback and oscillation. It is also used to reduce cross-talk and also to correct for a high frequency response slope in the transmission line.

4. For a phase modulated signal, the frequency deviation is proportional to _________
a) frequency only
b) amplitude only
c) only width
d) phase only

Answer: b
Explanation: For a phase modulated system, it is amplitude which is directly proportional to the deviation.

5. The frequency deviation is proportional to frequency in phase modulated signal.
a) True
b) False

Answer: b
Explanation: In phase modulation, the phase of the carrier signal is varied with respect to the amplitude of the message signal. For a phase modulated signal, the frequency deviation is proportional to amplitude.

6. Which is the true statement about frequency deviation in frequency modulation?
a) frequency deviation is proportional to carrier signal frequency
b) frequency deviation is proportional to amplitude of carrier signal
c) frequency deviation is proportional to

modulating frequency
d) frequency deviation is proportional to amplitude of modulating signal

Answer: d
Explanation: In frequency modulated system, the frequency deviation is proportional to the amplitude of the modulating signal.

7. Which of the following is not necessarily an advantage of FM over AM?
a) less modulating power is required
b) better noise immunity is provided
c) higher bandwidth is required
d) carrier is of any shape

Answer: c
Explanation: In frequency modulation, frequency of carrier gets varied with respect to the wave being propagated. FM has many advantages over AM but requirement of higher bandwidth is not an efficient condition.

8. What is number of possible outputs if there is 7 line digital input?
a) 64
b) 32
c) 16
d) 128

Answer: d
Explanation: Total possible outputs will be 2^7 which is equal to 128.

9. What is the frequency of the stereo sub carrier signal in FM broadcasting?
a) 19 KHz
b) 45 KHz
c) 55 KHz
d) 38 KHz

Answer: d
Explanation: Stereo broadcasting is made possible by using a subcarrier on FM radio stations, which takes the left channel and "subtracts" the right channel from it. A subcarrier is basically a sideband of a radio frequency carrier wave, which is modulated to send additional information. The frequency set for stereo sub carrier signal in FM broadcasting is 38 KHz.

TOPIC 1.8 SUPERHETERODYNE RECEIVERS

1. What is the bandwidth required in SSB signal?
a) f_m
b) $2f_m$
c) $> 2f_m$
d) $< 2f_m$

Answer: a
Explanation: In an AM modulated system, total bandwidth required is from $f_c + f_m$ to $f_c - f_m$ i.e. bandwidth is equal to $2f_m$. In SSB-SC transmission, the carrier and one of the sideband gets suppressed, so the bandwidth becomes f_m only.

2. One of the advantage of using a high frequency carrier wave is that it dissipates very small power.
a) True
b) False

Answer: a
Explanation: The main advantage of using high frequency signals is that the signal gets transmitted over very long distances and thus dissipates very less power. The antenna height required for transmission also gets reduced at high frequencies. And also it allows less noise interference and enables multiplexing. This is the reason for sending the audio signals at high frequency carrier signals for communication purpose.

3. What is the function of RF mixer?
a) Addition of two signals

b) Multiplication of two signals
c) Subtraction of two signals
d) To reduce the amount of noise

Answer: b
Explanation: RF mixer translates the frequencies of the two incoming signals by multiplying them and bringing them to a suitable band which can be processed.

4. The antenna current is 10A. Find the percentage of modulation when the antenna current increases to 10.4A?
a) 50%
b) 30%
c) 28.5%
d) 23%

Answer: c
Explanation:

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

On substituting the values in formula we have,

$$10.4 = 10\sqrt{1+\frac{m^2}{2}}$$

which gives m = 0.285 or 28.5%.

5. Find the total power, if the carrier of an AM transmitter is 800W and it is modulated to 50%?
a) 100W
b) 800W
c) 500W
d) 900W

Answer: d
Explanation: $P_T=P_C\ (1+{u^2}/_2)$, according to the problem P_C = 800W and m = 0.5. On substituting values in the equation we get $P_T=800(1+{0.5^2}/_2)$ = 900W.

6. Aliasing refers to ________
a) Sampling of signals less than at Nyquist rate
b) Sampling of signals at Nyquist rate
c) Sampling of signals greater than at Nyquist rate
d) Unsampled the original signal

Answer: a
Explanation: Aliasing refers to the sampling of signals less than at Nyquist rate. Nyquist rate states that the rate of sampling of signals should be greater than or equal to twice the bandwidth of modulating signal. It gets reduced if sampling is done at a higher rate than nyquist rate of sampling. Aliasing can be avoided by using anti-aliasing filters.

UNITII PULSE MODULATION

TOPIC 2.1 LOW PASS SAMPLING THEOREM

1. The frequency shift can be achieved by multiplying the band pass signal as given in equation
x(t) = $u_c(t)cos2\pi F_c t - u_s(t)sin2\pi F_c t$ by the quadrature carriers cos[2πF_ct] and sin[2πF_ct] and lowpass filtering the products to eliminate the signal components of 2F_c.
a) True
b) False

Answer: a
Explanation: It is certainly advantageous to perform a frequency shift of the band pass signal by and sampling the equivalent low pass signal. Such a frequency shift can be achieved by multiplying the band pass signal as given in the above equation by the quadrature carriers cos[2πF_ct] and sin[2πF_ct] and low pass filtering the products to eliminate the signal components at 2F_c. Clearly, the multiplication and the subsequent filtering are first performed in the analog

domain and then the outputs of the filters are sampled.

2. What is the final result obtained by substituting F_c=kB-B/2, T= 1/2B and say n = 2m i.e., for even and n=2m-1 for odd in equation x(nT)= $u_c(nT)cos2\pi F_cnT - u_s(nT)sin2\pi F_cnT$?
a) $(-1)^m u_c(mT_1) - u_s$
b) $u_s(mT_1 - \frac{T_1}{2})(-1)^{m+k+1}$
c) None
d) $(-1)^m u_c(mT_1) - u_s(mT_1 - \frac{T_1}{2})(-1)^{m+k+1}$

Answer: d
Explanation:
$x(nT) = u_c(nT)cos2\pi F_cnT - u_s(nT)sin2\pi F_cnT$ $\rightarrow$ equ1
=$u_c(nT)cos\frac{\pi n(2k-1)}{2} - u_s(nT)sin\frac{\pi n(2k-1)}{2}$ $\rightarrow$ equ2
On substituting the above values in equ1, we get say n=2m,
$x(2mT) \equiv xmT_{(1)} = u_c(mT_1)cos\pi m(2k-1) = (-1)^m u_c(mT_1)$
where $T_1 = 2T = \frac{1}{B}$. For n odd, say n=2m-1 in equ2 then we get the result as follows
$u_s(mT_1 - \frac{T_1}{2})(-1)^{m+k+1}$
Hence proved.

3. Which low pass signal component occurs at the rate of B samples per second with even numbered samples of x(t)?
a) u_c-lowpass signal component
b) u_s-lowpass signal component
c) u_c & u_s-lowpass signal component
d) none of the mentioned

Answer: a
Explanation: With the even-numbered samples of x(t), which occur at the rate of B samples per second, produce samples of the low pass signal component u_c.

4. Which low pass signal component occurs at the rate of B samples per second with odd numbered samples of x(t)?
a) u_c – lowpass signal component
b) u_s – lowpass signal component
c) u_c & u_s – lowpass signal component
d) none of the mentioned

Answer: b
Explanation: With the odd-numbered samples of x(t), which occur at the rate of B samples per second, produce samples of the low pass signal component u_s.

5. What is the reconstruction formula for the bandpass signal x(t) with samples taken at the rate of 2B samples per second?
a) $\sum_{m=-\infty}^{\infty} x(mT)\frac{sin(\pi/2T)(t-mT)}{(\pi/2T)(t-mT)}cos2\pi F_c(t-m$
b) $\sum_{m=-\infty}^{\infty} x(mT)\frac{sin(\pi/2T)(t+mT)}{(\pi/2T)(t+mT)}cos2\pi F_c(t-m$
c) $\sum_{m=-\infty}^{\infty} x(mT)\frac{sin(\pi/2T)(t-mT)}{(\pi/2T)(t-mT)}cos2\pi F_c(t+m$
d) $\sum_{m=-\infty}^{\infty} x(mT)\frac{sin(\pi/2T)(t+mT)}{(\pi/2T)(t+mT)}cos2\pi F_c(t+m$

Answer: a
Explanation:
$\sum_{m=-\infty}^{\infty} x(mT)\frac{sin(\pi/2T)(t-mT)}{(\pi/2T)(t-mT)}cos2\pi F_c(t-m$
, where T=1/2B

6. What is the new centre frequency for the increased bandwidth signal?
a) F_c'= F_c+B/2+B'/2
b) F_c'= F_c+B/2-B'/2
c) F_c'= F_c-B/2-B'/2
d) None of the mentioned

Answer: b
Explanation: A new centre frequency for the increased bandwidth signal is F_c' = F_c+B/2-B'/2

7. According to the sampling theorem for low pass signals with T_1=1/B, then what is the expression for u_c(t) = ?

a) $\sum_{m=-\infty}^{\infty} u_c(mT_1)\frac{sin(\frac{\pi}{T_1})(t-mT_1)}{(\pi/T_1)(t-mT_1)}$
b) $\sum_{m=-\infty}^{\infty} u_s(mT_1-\frac{T_1}{2})\frac{sin(\frac{\pi}{T_1})(t-mT_1+T_1/2)}{(\frac{\pi}{T_1})(t-mT_1+\frac{T_1}{2})}$
c) $\sum_{m=-\infty}^{\infty} u_c(mT_1)\frac{sin(\frac{\pi}{T_1})(t+mT_1)}{(\frac{\pi}{T_1})(t+mT_1)}$
d) $\sum_{m=-\infty}^{\infty} u_s(mT_1-\frac{T_1}{2})\frac{sin(\frac{\pi}{T_1})(t+mT_1+\frac{T_1}{2})}{(\frac{\pi}{T_1})(t+mT_1+\frac{T_1}{2})}$

Answer: a
Explanation: To reconstruct the equivalent low pass signals. Thus, according to the sampling theorem for low pass signals with T_1=1/B.
$u_c(t) = \sum_{m=-\infty}^{\infty} u_c(mT_1)\frac{sin(\frac{\pi}{T_1})(t-mT_1)}{(\pi/T_1)(t-mT_1)}$.

8. According to the sampling theorem for low pass signals with T_1=1/B, then what is the expression for $u_s(t)$ = ?
a) $\sum_{m=-\infty}^{\infty} u_c(mT_1)\frac{sin(\frac{\pi}{T_1})(t-mT_1)}{(\frac{\pi}{T_1})(t-mT_1)}$
b) $\sum_{m=-\infty}^{\infty} u_s(mT_1-\frac{T_1}{2})\frac{sin(\frac{\pi}{T_1})(t-mT_1+\frac{T_1}{2})}{(\pi/T_1)(t-mT_1+\frac{T_1}{2})}$
c) $\sum_{m=-\infty}^{\infty} u_s(mT_1-\frac{T_1}{2})\frac{sin(\frac{\pi}{T_1})(t-mT_1-\frac{T_1}{2})}{(\frac{\pi}{T_1})(t-mT_1-\frac{T_1}{2})}$
d) $\sum_{m=-\infty}^{\infty} u_c(mT_1)\frac{sin(\frac{\pi}{T_1})(t+mT_1)}{(\frac{\pi}{T_1})(t+mT_1)}$

Answer: b
Explanation: To reconstruct the equivalent low pass signals. Thus, according to the sampling theorem for low pass signals with T_1=1/B .
$u_s(t) = \sum_{m=-\infty}^{\infty} u_s(mT_1-T_1/2)\frac{sin(\pi/T_1)(t-mT_1+T_1/2)}{(\pi/T_1)(t-mT_1+T_1/2)}$

9. What is the expression for low pass signal component $u_c(t)$ that can be expressed in terms of samples of the bandpass signal?
a)
$\sum_{n=-\infty}^{\infty}(-1)^{n+r+1}x(2nT'-T')\frac{sin(\pi/(2T'))(t-2nT'+T')}{(\pi/(2T'))(t-2nT'+T')}$
b) $\sum_{n=-\infty}^{\infty}(-1)^{n}x(2nT')\frac{sin(\pi/(2T'))(t-2nT')}{(\pi/(2T'))(t-2nT')}$
c) All of the mentioned
d) None of the mentioned

Answer: b
Explanation: The low pass signal components $u_c(t)$ can be expressed in terms of samples of the
band pass signal as follows:
$u_c(t) = \sum_{n=-\infty}^{\infty}(-1)^{n}x(2nT')\frac{sin(\pi/(2T'))(t-2n}{(\pi/(2T'))(t-2nT'}$
.

10. What is the expression for low pass signal component $u_s(t)$ that can be expressed in terms of samples of the bandpass signal?
a)
$\sum_{n=-\infty}^{\infty}(-1)^{n+r+1}x(2nT'-T')\frac{sin(\pi/(2T'))(t-2}{(\pi/(2T'))(t-2n}$
b) $\sum_{n=-\infty}^{\infty}(-1)^{n}x(2nT')\frac{sin(\pi/(2T'))(t-2nT')}{(\pi/(2T'))(t-2nT')}$
c) All of the mentioned
d) None of the mentioned

Answer: a
Explanation: The low pass signal components $u_s(t)$ can be expressed in terms of samples of the
band pass signal as follows:
$u_s(t) = \sum_{n=-\infty}^{\infty}(-1)^{n+r+1}x(2nT'-T')\frac{sin(\pi/}{(\pi/(}$

11. What is the Fourier transform of x(t)?
a) X (F) = $\frac{1}{2}[X_l(F-F_c)+X_l^*(F-F_c)]$
b) X (F) = $\frac{1}{2}[X_l(F-F_c)+X_l^*(F+F_c)]$
c) X (F) = $\frac{1}{2}[X_l(F+F_c)+X_l^*(F-F_c)]$
d) X (F) = $\frac{1}{2}[X_l(F-F_c)+X_l^*(-F-F_c)]$

Answer: d
Explanation:
X (F) = $\int_{-\infty}^{\infty} x(t)e^{-j2\pi Ft}dt$
$=\int_{-\infty}^{\infty}\{Re[x_l(t)e^{j2\pi F_c t}]\}e^{-j2\pi Ft}dt$
Using the identity, R_e(ε)=1/2(ε+ε^*)
X (F) =
$\int_{-\infty}^{\infty}[x_l(t)e^{j2\pi F_c t}+x_l^*(t)e^{-j2\pi F_c t}]e^{-j2\pi Ft}dt$
$=\frac{1}{2}[X_l(F-F_c)+X_l^*(-F-F_c)]$.

12. What is the basic relationship between the spectrum o f the real band pass signal x(t) and the spectrum of the equivalent low pass signal $x_l(t)$?

a) X (F) = $\frac{1}{2}[X_l(F-F_c)+X_l^*(F-F_c)]$
b) X (F) = $\frac{1}{2}[X_l(F-F_c)+X_l^*(F+F_c)]$
c) X (F) = $\frac{1}{2}[X_l(F+F_c)+X_l^*(F-F_c)]$
d) X (F) = $\frac{1}{2}[X_l(F-F_c)+X_l^*(-F-F_c)]$

Answer: d
Explanation: X(F) = $\frac{1}{2}[X_l(F-F_c)+X_l^*(-F-F_c)]$, where $X_l(F)$ is the Fourier transform of $x_l(t)$. This is the basic relationship between the spectrum o f the real band pass signal x(t) and the spectrum of the equivalent low pass signal $x_l(t)$.

TOPIC 2.2 QUANTIZATION

1. Spread spectrum is used for
a) Encrypting signal
b) Hiding signal
c) Encrypting & Hiding signal
d) None of the mentioned

Answer: c
Explanation: Spread spectrum is used for hiding and encrypting signals.

2. Which is a quantization process?
a) Rounding
b) Truncation
c) Rounding & Truncation
d) None of the mentioned

Answer: c
Explanation: Rounding and truncation are examples of quantization process.

3. Quantization is a _____ process.
a) Few to few mapping
b) Few to many mapping
c) Many to few mapping
d) Many to many mapping

Answer: c
Explanation: Quantization is a many to few mapping process.

4. Quantization is a _____ process.
a) Non linear
b) Reversible
c) Non linear & Reversible
d) None of the mentioned

Answer: a
Explanation: Quantization is a non linear and irreversible process.

5. Which conveys more information?
a) High probability event
b) Low probability event
c) High & Low probability event
d) None of the mentioned

Answer: b
Explanation: High probability event conveys less information than a low probability event.

6. If the channel is noiseless information conveyed is ___ and if it is useless channel information conveyed is ___
a) 0,0
b) 1,1
c) 0,1
d) 1,0

Answer: d
Explanation: If the channel is noiseless information conveyed is 1 and if it is useless channel information conveyed is 0.

7. The mutual information between a pair of events is
a) Positive
b) Negative
c) Zero
d) All of the mentioned

Answer: d
Explanation: The mutual information between a pair of events can be positive negative or zero.

8. The output of the source encoder is an analog signal.

a) True
b) False

Answer: b
Explanation: The output of the source encoder is a sequence of binary digits. The conversion of source output to digital form is done here in source encoder.

9. The output of an information source is
a) Random
b) Deterministic
c) Random & Deterministic
d) None of the mentioned

Answer: a
Explanation: The output of any information source is random.

10. When the base of the logarithm is e, the unit of measure of information is
a) Bits
b) Bytes
c) Nats
d) None of the mentioned

Answer: c
Explanation: The unit of measure of information is determined based on the base of logarithm. If the base is e then the unit is nats(natural unit).

TOPIC 2.3 PAM

1. Flat top sampling of low pass signals
a) Gives rise to aperture effect
b) Implies over sampling
c) Leads to aliasing
d) Introduces delay distortion

Answer: a
Explanation: Flat top sampling of low pass signals gives rise to aperture effect.

2. In a delta modulation system, granular noise occurs when the
a) Modulating signal increases rapidly
b) Pulse rate decreases
c) Pulse amplitude decreases
d) Modulating signal remains constant

Answer: d
Explanation: In a delta modulation system, granular noise occurs when the modulating signal remains constant.

3. A PAM signal can be detected using
a) Low pass filter
b) High pass filter
c) Band pass filter
d) All pass filter

Answer: a
Explanation: A PAM signal can be detected by using low pass filter.

4. Coherent demodulation of FSK signal can be performed using
a) Matched filter
b) BPF and envelope detectors
c) Discriminator
d) None of the mentioned

Answer: a
Explanation: Coherent demodulation of FSK signal can be performed using matched filter.

5. The use of non uniform quantization leads to
a) Reduction in transmission bandwidth
b) Increase in maximum SNR
c) Increase in SNR for low level signals
d) Simplification of quantization process

Answer: c
Explanation: The use of non uniform quantization leads to increase in SNR for low level signals.

6. Which of the following requires a synchronizing signal?
a) Single channel PPM system
b) PAM
c) DM
d) All of the mentioned

Answer: b
Explanation: PAM requires a synchronizing signal.

7. A PWM signal can be generated by
a) An astable multi vibrator
b) A monostable multi vibrator
c) Integrating a PPM signal
d) Differentiating a PPM signal

Answer: b
Explanation: A PWM signal can be generated by a mono stable multi vibrator.

8. TDM is less immune to cross-talk in channel than FDM.
a) True
b) False

Answer: b
Explanation: False because different message signals are not applied to the channel simultaneously.

9. In an ideal TDM system, the cross correlation between two users of the system is
a) 1
b) 0
c) Infinity
d) -1

Answer: b
Explanation: In an ideal TDM system, the cross correlation between two users of the system is 0.

10. TDM requires
a) Constant data transmission
b) Transmission of data samples
c) Transmission of data at random
d) Transmission of data of only one measured

Answer: b
Explanation: TDM requires transmission of data samples.

TOPIC 2.4 LINE CODING

1. Which waveforms are also called as line codes?
a) PCM
b) PAM
c) FM
d) AM

Answer: a
Explanation: When pulse modulation is applied to binary symbol we obtain pulse code modulated waveforms. These waveforms are also called as line codes.

2. When pulse code modulation is applied to non binary symbols we obtain waveform called as
a) PCM
b) PAM
c) M-ary
d) line codes

Answer: c
Explanation: When pulse code modulation is applied to binary symbols we get PCM waveforms and when it is applied to non binary symbols we obtain M-ary waveforms.

3. Examples of PCM waveforms are
a) Non return to zero
b) Phase encoded
c) Multilevel binary
d) All of the mentioned

Answer: d
Explanation: Some of the examples or classification of pulse code modulated signals are non return to zero, return to zero, phase encoded, multilevel binary etc.

4. Which type is used and preferred in digital logic circuits?
a) NRZ-L
b) NRZ-M
c) NRZ-S
d) None of the mentioned

Answer: a
Explanation: NRZ-L is extensively used in

digital logic circuits. In this method, logic 1 is represented by one voltage level and logic 0 is represented by another voltage level.

5. Which method is called as differential encoding?
a) NRZ-L
b) NRZ-M
c) NRZ-S
d) None of the mentioned

Answer: b
Explanation: In NRZ-M, logic 1 is represented by a change in voltage level and logic 0 is represented by no change in level. This is called as differential encoding.

6. Which method is preferred in magnetic tape recording?
a) NRZ-L
b) NRZ-M
c) NRZ-S
d) None of the mentioned

Answer: b
Explanation: NRZ-M is also called as differential encoding and it is most preferred in magentic tape recording.

7. NRZ-S is complement of _______
a) NRZ-L
b) NRZ-M
c) NRZ-L & NRZ-M
d) None of the mentioned

Answer: b
Explanation: NRZ-S is a complement of NRZ-M. Logic 0 is represented by a change in voltage level and logic 1 is represented as no change in voltage level.

8. The return to zero waveform consists of
a) Unipolar RZ
b) Bipolar RZ
c) RZ-AMI
d) All of the mentioned

Answer: d
Explanation: Different types of return to zero waveforms are unipolar RZ, bipolar RZ, RZ-AMI. These are used in baseband transmission and in magnetic recording.

9. Phase encoded group consists of
a) Manchester coding
b) Bi-phase-mark
c) Miller coding
d) All of the mentioned

Answer: d
Explanation: Different types of phase encoded waveform consists of manchester coding, bi-phase-mark, bi-phase-space, delay modulation.

10. In which waveform logic 1 is represented by half bit wide pulse and logic 0 is represented by absence of pulse?
a) Unipolar RZ
b) Bipolar RZ
c) RZ-AMI
d) Manchester coding

Answer: a
Explanation: In unipolar RZ waveform, logic 1 is represented by half bit wide pulse and logic 0 is represented by the absence of a pulse.

11. In which waveform logic 1 and logic 0 are represented by opposite one half bit wide pulses?
a) Unipolar RZ
b) Bipolar RZ
c) RZ-AMI
d) Manchester coding

Answer: b
Explanation: In bipolar return to zero waveform ones and zeroes are represented by opposite level pulses one half bit wide pulses.

12. In which waveform logic 1 is represented by equal amplitude alternating pulses?
a) Unipolar RZ

b) Bipolar RZ
c) RZ-AMI
d) Manchester coding

Answer: c
Explanation: In RZ-AMI logic 1 is represented by equal amplitude alternating pulses and logic 0 is represented by the absence of a pulse.

TOPIC 2.5 PCM, DPCM, ADPCM AND ADM

1. The signals which are obtained by encoding each quantized signal into a digital word is called as
a) PAM signal
b) PCM signal
c) FM signal
d) Sampling and quantization

Answer: b
Explanation: Pulse code modulation is the name for the class of signals which are obtained by encoding the quantized signals into a digital word.

2. The length of the code-word obtained by encoding quantized sample is equal to
a) l=log(to the base 2)L
b) l=log(to the base 10)L
c) l=2log(to the base 2)L
d) l=log(to the base 2)L/2

Answer: a
Explanation: The quantized sample which are digitally encoded into l bit value code-word. The length l can be calculated as l=log(to the base 2)L.

3. Quantization noise can be reduced by ________ the number of levels.
a) Decreasing
b) Increasing
c) Doubling
d) Squaring

Answer: b
Explanation: The process of quantization replaces the true signal with the approximation(quantization noise). By increasing the number of quantization level the quantization noise can be reduced.

4. In PCM encoding, quantization level varies as a function of ________
a) Frequency
b) Amplitude
c) Square of frequency
d) Square of amplitude

Answer: b
Explanation: In linear PCM the quantization levels are uniform. But in normal PCM encoding the quantization level vary according to the amplitude, based of A-law of Myu-law.

5. What is bit depth?
a) Number of quantization level
b) Interval between two quantization levels
c) Number of possible digital values to represent each sample
d) None of the mentioned

Answer: c
Explanation: One of the properties of PCM signal which determines its stream fidelity is bit depth which is the number of possible digital values that can be used to represent each sample.

6. Choosing a discrete value that is near but not exactly at the analog signal level leads to
a) PCM error
b) Quantization error
c) PAM error
d) Sampling error

Answer: b
Explanation: One of the limitations of PCM is quantization error which occurs when we choose a discrete value at some near by value and not at the analog signal level.

7. In PCM the samples are dependent on ________
a) Time
b) Frequency
c) Quanization leavel
d) Interval between quantization level

Answer: a
Explanation: The samples depend on time,an accurate clock is required for accurate reproduction.

8. DPCM encodes the PCM values based on
a) Quantization level
b) Difference between the current and predicted value
c) Interval between levels
d) None of the mentioned

Answer: b
Explanation: Differential PCM encodes the PCM value based on the difference between the previous sample and the present sample value.

9. Delta modulation uses _____ bits per sample.
a) One
b) Two
c) Four
d) Eight

Answer: a
Explanation: Delta modulation is used for analog to digital conversion and vice versa. It is a simple form of DPCM. Its uses 1 bit per sample. It also depends on the difference between the current and previous sample values.

10. Sample resolution for LPCM ____ bits per sample.
a) 8
b) 16
c) 24
d) All of the mentioned

Answer: d
Explanation: Common sampling resolution for LPCM are 8, 16, 20, 24 bits per sample.

11. Adaptive DPCM is used to
a) Increase bandwidth
b) Decrease bandwidth
c) Increase SNR
d) None of the mentioned

Answer: b
Explanation: Adaptive DPCM is used to decrease required bandwidth for the given SNR.

TOPIC 2.6 CHANNEL VOCODER

1. Vocoders analyse the speech signals at ______
a) Transmitter
b) Receiver
c) Channel
d) IF Filter

Answer: a
Explanation: Vocoders are a class of speech coding systems. They analyse the speech signal at the transmitter. And then transmit the parameters derived from the analysis.

2. Vocoders __________ the voice at the receiver.
a) Analyse
b) Synthesize
c) Modulate
d) Evaluate

Answer: b
Explanation: Vocoders synthesize the voice at the receiver. All vocoder systems attempt to model the speech generation process as a dynamic system and try to quantify certain physical constraints of the system.

3. Vocoders are simple than the waveform coders.

a) True
b) False

Answer: b
Explanation: Vocoders are much more complex than the waveform coders. They can achieve very high economy in transmission bit rate but are less robust.

4. Which of the following is not a vocoding system?
a) Linear predictive coder
b) Channel vocoder
c) Waveform coder
d) Formant vocoder

Answer: c
Explanation: Waveform coder is not a vocoding system. LPC (linear predictive coding) is the most popular vocoding system. Other vocoding systems are channel vocoder, formant vocoder, cepstrum vocoder etc.

5. Which of the following pronunciations lead to voiced sound?
a) 'f'
b) 's'
c) 'sh'
d) 'm'

Answer: d
Explanation: Voiced sounds are 'm', 'n' and 'v' pronounciations. They are a result of quasiperiodic vibrations of the vocal chord.

6. Speech signal can be categorised in _____ and _______
a) Voiced, unvoiced
b) Active, passive
c) Direct, indirect
d) Balanced, unbalanced

Answer: a
Explanation: Speech signal is of two types, voiced and unvoiced. Voiced sound is a result of quasiperiodic vibrations of the vocal chord. Unvoiced signals are fricatives produced by turbulent air flow through a constriction.

7. Channel vocoders are the time domain vocoders.
a) True
b) False

Answer: b
Explanation: Channel vocoders are frequency domain vocoders. They determine the envelope of the speech signal for a number of frequency bands and then sample, encode and multiplex these samples with the encoded outputs of the other filters.

8. ________ is often called the formant of the speech signal.
a) Pitch frequency
b) Voice pitch
c) Pole frequency
d) Central frequency

Answer: c
Explanation: The pole frequencies correspond to the resonant frequencies of the vocal tract. They are often called the formants of the speech signal. For adult speakers, the formants are centered around 500 Hz, 1500 Hz, 2500 Hz and 3500 Hz.

9. Formant vocoders use large number of control signals.
a) True
b) False

Answer: b
Explanation: Formant vocoders use fewer control signals. Therefore, formant vocoders can operate at lower bit rates than the channel vocoder. Instead of transmitting the power spectrum envelope, formant vocoders attempt to transmit the position of peaks of spectral envelope.

10. Cepstrum vocoder uses __________
a) Wavelet transform
b) Inverse wavelet transform
c) Cosine transform
d) Inverse Fourier transform

Answer: d
Explanation: Cepstrum vocoders use inverse Fourier transform. It separates the excitation and vocal tract spectrum by Fourier transforming spectrum to produce the cepstrum of the signal.

TOPIC 2.7 TIME DIVISION MULTIPLEXING

1. The real part of an antenna's input impedance is due to ________
a) SWR
b) radiated signal
c) reflected signal
d) refracted signal

Answer: b
Explanation: In antenna impedance, impedance related the voltage and current at the input of the antenna. The real part of antenna impedance represents power that is either radiated away or absorbed within the antenna and the imaginary part of antenna impedance represents power that is stored in the near field of antenna.

2. What is the other name for half-wave dipole antenna?
a) Helical antenna
b) Isotropic antenna
c) Hertz antenna
d) Maxwell antenna

Answer: c
Explanation: The Hertz antenna is also known as half wave dipole antenna. It consists of two straight collinear conductors of equal length separated by a small feeding gap.

3. Measured on the ground, the field strength of a horizontally polarized half wave dipole antenna is strongest ________
a) in one direction
b) in two directions
c) depends on the number of elements
d) depends on the shape of antenna

Answer: b
Explanation: As the name suggests, half wave dipole is half wavelength long. This antenna has the shortest resonant length that can be used for a resonant dipole. The field strength of a horizontally polarized half wave dipole antenna is strongest in two directions.

4. When an antenna radiates more energy in one direction than in other directions, it is called ________
a) selectivity
b) directivity
c) active antenna
d) resonance

Answer: b
Explanation: When an antenna radiates more energy in one direction than in other directions is called directivity. An antenna that radiates equally in all directions has effectively zero directionality, and the directivity of this type of antenna should be 1 (or 0dB).

5. What is the full form of ERP?
a) Effective Radiated Power
b) Effective Reflected Power
c) Equivalent Radiated Power
d) Equivalent Reflected Power

Answer: a
Explanation: ERP stands for Effective Radiated Power. Effective Radiated Power (ERP) is always given with respect to a certain direction.

6. "Ground Effect" in antenna caused by ________
a) faulty connection of the feed cable ground
b) fading
c) buildings and other structures on ground
d) radio signals reflecting off the ground

Answer: d
Explanation: Radio signals that are reflecting back from the ground is responsible for ground effects in antenna.

7. The polarization of plane waves received from satellite is changed by ________
a) Faraday rotation
b) Gamma rays
c) Helical rotation
d) Distance travelled

Answer: a
Explanation: Generally for satellite communication circular polarization is required. The polarization received by waves from satellite is changed by Faraday rotation.

8. What is the input impedance to a lossless antenna, at resonance?
a) infinite
b) 0
c) resistive
d) capacitive

Answer: c
Explanation: In antenna impedance, impedance related the voltage and current at the input of the antenna. The real part of antenna impedance represents power that is either radiated away or absorbed within the antenna and the imaginary part of antenna impedance represents power that is stored in the near field of antenna. The input impedance of a lossless antenna is purely resistive.

TOPIC 2.8 FREQUENCY DIVISION MULTIPLEXING

1. TDMA stands for ________
a) Time Division Multiple Access
b) Time Domain Multiple Access
c) Time Division Mutual Access
d) Time Domain Mutual Access

Answer: a
Explanation: TDMA stands for Time Division Multiple Access. It can be seen as a channel access method for shared-medium networks.

2. Which term is used when signals move from one line to another?
a) path switching
b) space switching
c) line switching
d) cross-point switching

Answer: b
Explanation: Space switching is the used term for signals moving from one line to another.

3. PSK stands for Pulse Shift Keying.
a) True
b) False

Answer: b
Explanation: PSK stands for Phase Shift Keying. It is a modulation scheme that conveys information by changing the phase of carrier.

4. Which term is used for moving PCM samples from one time slot to another?
a) time switching
b) space switching
c) phase switching
d) frequency switching

Answer: a
Explanation: Time switching is the used term for moving PCM samples moving from one time slot to another.

5. Power can be coupled into or out of a waveguide with a magnetic field probe.
a) True
b) False

Answer: b
Explanation: A waveguide is a line through which electromagnetic waves are passed for

various use. Power can be coupled into or out of a waveguide not only with a magnetic field probe. It can also be coupled with an electric field probe. It can also be coupled through a hole in the waveguide.

6. What is the full form of LOS?
a) Level Of Signal
b) Line Of Sight
c) Loss Of Signal
d) Level Of Sight

Answer: b
Explanation: Line Of Sight is a line between two points. It is a straight path between a transmitting antenna and a receiving antenna.

7. How we can define the satisfactory performance of an analog microwave system?
a) carrier to noise ratio that exceeds a given value
b) carrier to noise ratio that is below a given value
c) an ERP value that exceeds a given value
d) an ERP value that is below a given value

Answer: a
Explanation: We can measure performance of an analog microwave system by calculating the carrier to noise ratio that exceeds a given value. It gives the signal to noise ratio.

8. RGB stands for ________
a) Red Green Brown
b) Red Green Black
c) Red Gold Blue
d) Red Green Blue

Answer: d
Explanation: RGB stands for Red Green Blue. It is an additive color model in which red, green and blue light intensity and different shades are added together in various ways to reproduce a broad variety of colors.

9. How many lines are there in an NTSC signal?
a) 1024
b) 1856
c) 625
d) 525

Answer: d
Explanation: NTSC stands for National Television System Committee. In NTSC, it is standardized fixed that it has total 525 lines.

10. Luminance refers to ________
a) contrast
b) diffusion
c) brightness
d) aperture

Answer: c
Explanation: Luminance refers to brightness. It is a photometric measure of luminous intensity per unit area of light travelling in a given direction.

UNIT III DIGITAL MODULATION AND TRANSMISSION

TOPIC 3.1 PHASE SHIFT KEYING – BPSK, DPSK, QPSK

1. In linear modulation technique __________ of transmitted signal varies linearly with modulating digital signal.
a) Amplitude
b) Frequency
c) Phase
d) Angle

Answer: a
Explanation: In linear modulation technique, the amplitude of transmitted signal varies linearly with modulating digital signal. It is a form of digital modulation technique.

2. Linear modulation techniques are not bandwidth efficient.
a) True
b) False

Answer: b
Explanation: Linear modulation techniques are bandwidth efficient. They are used in wireless communication systems when there is an increasing demand to accommodate more and more users within a limited spectrum.

3. Which of the following is not a linear modulation technique?
a) OQPSK
b) π/4 QPSK
c) FSK
d) BPSK

Answer: c
Explanation: OQPSK, π/4 QPSK and BPSK are the most popular linear modulation techniques. They have very good spectral efficiency. However, FSK is an non-linear modulation technique.

4. In BPSK, the ________ of constant amplitude carrier signal is switched between two values according to the two possible values.
a) Amplitude
b) Phase
c) Frequency
d) Angle

Answer: b
Explanation: In binary phase shift keying (BPSK), the phase of a constant amplitude carrier signal is switched between two possible values m1 and m2. These two values corresponds to binary 1 and 0 respectively.

5. By applying cos(2πft), BPSK signal is equivalent to ________
a) Double sideband suppressed carrier amplitude modulated waveform
b) Single sideband suppressed carrier amplitude modulated waveform
c) Frequency modulated waveform
d) SSB amplitude waveform

Answer: a
Explanation: The BPSK signal is equivalent to a double sideband suppressed carrier amplitude modulated waveform, where cos(2πft) is applied as the carrier. Hence, a BPSK signal can be generated using a balanced modulator.

6. BPSK uses non-coherent demodulator.
a) True
b) False

Answer: b
Explanation: BPSK uses coherent or synchronous demodulation. It requires the information about the phase and frequency of the carrier be available at the receiver.

7. DPSK uses coherent form of PSK.
a) True
b) False

Answer: b
Explanation: Differential phase shift keying uses noncoherent form of phase shift keying. Noncoherent form avoids the need for a coherent reference signal at the receiver. Noncoherent receivers are also easy and cheap to build.

8. In DPSK system, input signal is differentially encoded and then modulated using a ________ modulator.
a) Amplitude
b) Frequency
c) BPSK
d) QPSK

Answer: c
Explanation: In DPSK system, input binary sequence is first differentially encoded and then modulated using a BPSK modulator. The differentially encoded sequence is generated

from input binary sequence by complimenting their modulo-2 sum.

9. The energy efficiency of DPSK is _________ to coherent PSK.
a) Superior
b) Same
c) Zero
d) Inferior

Answer: d
Explanation: The energy efficiency of DPSK is inferior to that of coherent PSK by about 3 dB. But, it has an advantage of reduced receiver complexity.

10. QPSK has ________ the bandwidth efficiency of BPSK.
a) Twice
b) Same
c) Half
d) Four times

Answer: a
Explanation: Quadrature phase shift keying (QPSK) has twice the bandwidth of BPSK. It is because two bits are transmitted in a single modulation symbol. The phase of the carrier takes on one of the four equally spaced values, where each value of phase corresponds to a unique pair of message bit.

11. QPSK provides twice the bandwidth efficiency and _______ energy efficiency as compared to BPSK.
a) Twice
b) Half
c) Same
d) Four times

Answer: c
Explanation: The bit error probability of QPSK is identical to BPSK but twice as much data can be sent in the same bandwidth. Thus, when compared to BPSK, QPSK provides twice the spectral efficiency with exactly the same efficiency.

12. What is the full form of OQPSK?
a) Optical Quadrature Phase Shift Keying
b) Orthogonal Quadrature Pulse Shift Keying
c) Orthogonal Quadrature Phase Shift Keying
d) Offset Quadrature Phase Shift Keying

Answer: d
Explanation: OQPSK stands for offset quadrature phase shift keying. It is a modified form of QPSK which is less susceptible to deleterious effects and supports more efficient amplification. OQPSK is sometimes also called staggered QPSK.

13. The bandwidth of OQPSK is _______ to QPSK.
a) Identical
b) Twice
c) Half
d) Four times

Answer: a
Explanation: The spectrum of an OQPSK signal is identical to that of QPSK signal. Hence, both signals occupy the same bandwidth. The staggered alignment of the even and odd bit streams in OQPSK signal does not change the nature of spectrum.

14. QPSK signals perform better than OQPSK in the presence of phase jitter.
a) True
b) False

Answer: b
Explanation: OQPSK signal perform better than QPSK in the presence of phase jitter. It is due to the presence of noisy reference signal at the receiver.

15. Which of the following is not a detection technique used for detection of π/4 QPSK signals?
a) Baseband differential detection
b) IF differential detection
c) FM discriminator detection
d) Envelope detection

Answer: d
Explanation: There are various types of detection techniques used for the detection of π/4 QPSK signals. They include baseband differential detection, IF differential detection and FM discriminator detection.

TOPIC 3.2 PRINCIPLES OF M-ARY SIGNALING M-ARY PSK & QAM

1. Which of the following is a combined linear and constant envelope technique?
a) MPSK
b) PSK
c) BPSK
d) QPSK

Answer: a
Explanation: M-ary phase shift keying (MPSK) is a combined linear and constant envelope technique. It is a part of M-ary modulation techniques. These modern modulation techniques exploit the fact that digital baseband data may be sent by varying both the envelope and phase of an RF carrier.

2. In an M-ary signalling scheme two or more bits are grouped together to form a ________
a) Chip
b) Symbol
c) Byte
d) Pattern

Answer: b
Explanation: In an M-ary signalling scheme two or more bits are grouped together to form symbols. And one of the M possible signals is transmitted during each symbol period of duration Ts.

3. The number of possible signal in M-ary signalling is given by M and M = __________ where n is an integer.
a) n
b) 2n
c) 2^n
d) n^2

Answer: c
Explanation: Two or more bits are grouped to form a symbol in M-ary modulation. And the number of possible symbols should be equal to 2^n, where n is an integer.

4. M-ary signalling techniques are not sensitive to timing jitters.
a) True
b) False

Answer: b
Explanation: Timing errors increase when smaller distances between signals in the constellation diagram are used. M-ary signalling techniques are attractive for use in bandlimited channel, but are limited in their applications due to sensitivity in timing jitter.

5. M-ary modulation schemes have very good power efficiency.
a) True
b) False

Answer: b
Explanation: M-ary modulation schemes have poor power efficiency, but they have a better bandwidth efficiency. An 8-PSK system requires a bandwidth that is 3 times smaller than a BPSK system, whereas its BER performance is very worse since signals are packed more closely in the signal constellation.

6. In M-ary PSK, the carrier ___________ takes one of M possible values.
a) Amplitude
b) Frequency
c) Angle
d) Phase

Answer: d
Explanation: In an M-ary PSK, the carrier phase takes one of the M possible values. The

possible values of phase are θ_i=2(i-1)/M, where i=1,2,…….,M.

7. The constellation of M-ary PSK is ____________ dimensional.
a) One
b) Does not exist
c) Two
d) Three

Answer: c
Explanation: The constellation of M-ary PSK is two dimensional. It is due to the presence of two basis signals. And the M-ary message points are equally spaced on a circle.

8. What is the radius of the circle in M-ary PSK on which message points are equaly spaced?
a) $\sqrt{E_s}$
b) $\sqrt{E_b}$
c) E_b
d) E_s

Answer: a
Explanation: The M-ary message points are equally spaced on a circle of radius $\sqrt{E_s}$ centred at the origin. Here E_s is energy per symbol. Thus, MPSK is a constant envelope signal when no pulse shaping is used.

9. As the value of M _________ the bandwidth efficiency ________
a) Increases, same.
b) Increases, decreases
c) Increases, increases
d) Decreases, same

Answer: c
Explanation: The first null bandwidth of M-ary PSK signals decrease as M increases while Rb is held constant. Therefore, as the value of M increases, the bandwidth efficiency also increases.

10. The power efficiency of the M ary PSK decreases because of the _____
a) Freely packed constellation
b) Increment of bandwidth efficiency
c) Fixed null bandwidth
d) Densely packed constellation

Answer: d
Explanation: Bandwidth efficiency increases as the value of M increases. But at the same time, increasing M implies that the constellation is more densely packed. Hence the power efficiency or noise tolerance is decreased.

11. In QAM, the amplitude is _______ and phase is _______
a) Varied, constant
b) Varied, varied
c) Constant, varied
d) Constant, constant

Answer: b
Explanation: Quadrature amplitude modulation (QAM) is obtained by allowing the amplitude to also vary with the phase. Thus, the constellation consists of square lattice of signal points.

12. M-ary QAM signal have constant energy per symbol.
a) True
b) False

Answer: b
Explanation: M-ary QAM does not have constant energy per symbol. It also does not have constant distance between possible symbol states. It reasons that particular values of M-ary QAM signal will be detected with higher probability than others.

13. In comparison to M-ary PSK, M-ary QAM bandwidth efficiency is _____ and power efficiency is ________
a) Identical, superior
b) Less, superior
c) Identical, identical
d) Superior, superior

Answer: a
Explanation: The power spectrum and bandwidth efficiency of QAM modulator is identical to M-ary PSK modulation. But, in terms of power efficiency QAM is superior to M-ary PSK.

14. The bandwidth efficiency of an M-ary FSK signal ________ with ________ in M.
a) Constant, increase
b) Increases, increase
c) Decreases, increase
d) Decreases, decrease

Answer: c
Explanation: The bandwidth efficiency of an M-ary FSK signal decreases with increase in M. Therefore, unlike M-PSK signals, M-FSK signals are bandwidth inefficient.

15. Power efficiency of M-ary FSK increases, since _________
a) Constellation is densely packed
b) M signals are non-orthogonal
c) Fixed null bandwidth
d) M-signals are orthogonal

Answer: d
Explanation: In M-ary FSK, all the M signals are orthogonal and there is no crowding in the signal space. Hence, power efficiency of M-ary FSK increases with M.

TOPIC 3.3 ISI, COSINE FILTERS

1. The method in which the tail of one pulse smears into adjacent symbol interval is called as
a) Intersymbol interference
b) Interbit interference
c) Interchannel interference
d) None of the mentioned

Answer: a
Explanation: Due to the effect of system filtering the received pulse can overlap on one and another. The tail of one pulse smears into the adjacent symbol interval thereby interfering the detection process. This process is called as intersymbol interference.

2. If each pulse of the sequence to be detected is in _____ shape, the pulse can be detected without ISI.
a) Sine
b) Cosine
c) Sinc
d) None of the mentioned

Answer: c
Explanation: The sinc shaped pulse is the ideal nyquist pulse. If each pulse in the sequence to be detected is in sinc shape the pulses can be detected without ISI.

3. What is symbol rate packing?
a) Maximum possible symbol transmission rate
b) Maximum possible symbol receiving rate
c) Maximum bandwidth
d) Maximum ISI value allowed

Answer: a
Explanation: A system with bandwidth Rs/2 can support a maximum transmission rate of Rs without ISI. Thus for ideal Nyquist filtering the maximum possible symbol transmission rate is called as symbol rate packing and it is equal to 2 symbols/s/Hz.

4. A nyquist pulse is the one which can be represented by _____ shaped pulse multiplied by another time function.
a) Sine
b) Cosine
c) Sinc
d) None of the mentioned

Answer: c
Explanation: A nyquist filter is one whose frequency transfer function can be represented by a rectangular function convolved with any real even symmetric frequency function and a nyquist pulse is one

whose shape can be represented by sinc function multiplied by another time function.

5. Examples of nyquist filters are
a) Root raised cosine filter
b) Raised cosine filter
c) Root raised & Raised cosine filter
d) None of the mentioned

Answer: c
Explanation: The most popular among the class of nyquist filters are raised cosine and root raised cosine filter.

6. The minimum nyquist bandwidth for the rectangular spectrum in raised cosine filter is
a) 2T
b) 1/2T
c) T^2
d) 2/T

Answer: b
Explanation: For raised cosine spectrum the minimum nyquist bandwidth is equal to 1/2T.

7. Roll off factor is the fraction of
a) Excess bandwidth and absolute bandwidth
b) Excess bandwidth and minimum nyquist bandwidth
c) Absolute bandwidth and minimum nyquist bandwidth
d) None of the mentioned

Answer: b
Explanation: The roll off factor is defined by a fraction of excess bandwidth and the minimum nyquist bandwith. It ranges from 0 to 1.

8. Which value of r (roll off factor) is considered as Nyquist minimum bandwidth case?
a) 0
b) 1
c) Infinity
d) None of the mentioned

Answer: a
Explanation: For the roll off factor of 0 an ideal rectangular nyquist pulse is obtained. This is called as nyquist minimum bandwidth case.

9. A pulse shaping filter should satisfy two requirements. They are
a) Should be realizable
b) Should have proper roll off factor
c) Should be realizable & have proper roll off factor
d) None of the mentioned

Answer: c
Explanation: A pulse shaping filter should provide the desired roll off and should be realizable, that is the impulse response needs to be truncated to a finite length.

10. Examples of double side band signals are
a) ASK
b) PSK
c) ASK & PSK
d) None of the mentioned

Answer: c
Explanation: ASK and PSK needs twice the transmission bandwidth of equivalent baseband signals. Thus these are called as double side band signals.

TOPIC 3.4 PULSE SHAPING

1. Intersymbol interference (ISI) leads to ________ probability of the receiver for making an error in detecting the symbols.
a) Increased
b) Decreased
c) Zero
d) One

Answer: a
Explanation: ISI leads to increased probability of the receiver making an error in detecting a symbol. When rectangular pulses are passed through a bandlimited channel, the

pulses will spread in time, and the pulse for each symbol will smear into the time intervals of succeeding symbols.

2. ISI is ________ by increasing channel bandwidth.
a) Maximized
b) Minimized
c) Zero
d) Infinite

Answer: b
Explanation: Increasing channel bandwidth is one of the method to minimize intersymbol interference. But mobile communication systems use minimal bandwidth, thus other methods to reduce ISI are desirable.

3. Why is pulse shaping technique used?
a) To increase ISI
b) To increase spectral width of modulated signal
c) To reduce ISI
d) To reduce power spectral density

Answer: c
Explanation: Pulse shaping techniques reduces the intersymbol interference. They are also used to reduce the spectral width of the modulated digital signal.

4. Who was the first to solve the problem of ISI?
a) Manchester
b) Faraday
c) Graham Bell
d) Nyquist

Answer: d
Explanation: Nyquist was the first to solve the problem of ISI. He overcome the problem of ISI while keeping the transmission bandwidth low. He observed that ISI can be completely nullified if at every instant, the response due to all symbols except the current symbol is equal to zero.

5. According to Nyquist, the impulse response of the overall communication system should have ______ decay with _______ magnitude for sample values not equal to zero.
a) Fast, small
b) Slow, small
c) Slow, large
d) Fast. Large

Answer: a
Explanation: According to Nyquist, the impulse response of the overall communication system should have fast decay with small magnitude for sample values not equal to zero. If the channel is ideal then it should be possible to realize approximate shaping filters at both transmitter and receiver.

6. Raised cosine filter does not satisfy Nyquist criteria.
a) True
b) False

Answer: b
Explanation: Raised cosine filter is the most popular pulse shaping filter used in mobile communication. It belongs to the class of filters that satisfy Nyquist criterion.

7. As the roll off factor in raised cosine rolloff filter __________ the occupied bandwidth ________
a) Increases, decreases
b) Decreases, constant
c) Increases, increases
d) Decreases, increases

Answer: c
Explanation: As the rolloff factor increases, the bandwidth of the filter also increases and the time sidelobe levels decrease in adjacent symbol slots. Thus, it implies that increasing rolloff factor decreases the sensitivity to timing jitter but increases the occupied bandwidth.

8. Gaussian pulse shaping filter follows Nyquist criterion.
a) True
b) False

Answer: b
Explanation: Gaussian pulse shaping filter uses non Nyquist technique. It is effective when used in conjunction with minimum shift keying (MSK) modulation, or other modulation which is well suited for power efficient nonlinear amplifiers.

9. Gaussian filter has zero crossings at adjacent symbol peaks.
a) True
b) False

Answer: b
Explanation: Nyquist filters have zero crossings at adjacent symbol peaks and a truncated transfer function. Gaussian filter does not follow Nyquist criterion and has a smooth transfer function with no zero crossings.

10. Which of the following is true for a Gaussian filter?
a) Large bandwidth
b) Minimum ISI
c) High overshoot
d) Sharp cut off

Answer: d
Explanation: The Gaussian filter has a narrow absolute bandwidth, and has a sharp cut off, low overshoot and pulse area preservation properties. This makes it attractive for use in mobile communication that uses nonlinear RF amplifiers.

11. Gaussian pulse shaping filter reduces the spectral occupancy and ISI.
a) True
b) False

Answer: b
Explanation: Gaussian pulse shaping does not satisfy Nyquist criterion for ISI cancellation. Thus, it reduces the spectral occupancy but there is degradation in the performance due to increased ISI.

12. Gaussian pulses are used when cost and power efficiency are major factors.
a) True
b) False

Answer: a
Explanation: Gaussian pulses are used when cost and power efficiency are major factors. But the bit error rates due to ISI are deemed to be lower than what is nominally required. Thus, there is a trade-off between desired RF bandwidth and irreducible error due to ISI.

TOPIC 3.5 DUO BINARY ENCODING

1. The method in which small amount of controlled ISI is introduced into the data stream rather than trying to eliminate it completely is called as
a) Correlative coding
b) Duobinary signalling
c) Partial response signalling
d) All of the mentioned

Answer: d
Explanation: The interference at the detector can be cancelled out using these methods in which some controlled amount of ISI is introduced into the data stream.

2. From digital filter we will get the output pulse as the _______ of the current and the previous pulse.
a) Summation
b) Difference
c) Product
d) Ratio

Answer: a
Explanation: The digital filter incorporates

one digit delay and thus it adds the incoming pulse with the value of the previous pulse.

3. In duobinary signalling method, for M-ary transmission, the number of output obtained is
a) 2M
b) 2M+1
c) 2M-1
d) M2

Answer: c
Explanation: In duobinary coding, the number of output obtained for M-ary transmission is 2M-1.

4. The method using which the error propagation in dubinary signalling can be avoided is
a) Filtering
b) Precoding
c) Postcoding
d) None of the mentioned

Answer: b
Explanation: In duobinary signalling method if one error occurs it repeats everywhere through out the next steps. To avoid this precoding method can be used.

5. In precoding technique, the binary sequence is _____ with the previous precoded bit.
a) And-ed
b) Or-ed
c) EXOR-ed
d) Added

Answer: c
Explanation: To avoid error propogation precoding method is used. In this each bit is encoded individually without having any effect due to its prior bit or decisions.

6. The duobinary filter, He (f) is called as
a) Sine filter
b) Cosine filter
c) Raised cosine filter
d) None of the mentioned

Answer: b
Explanation: The transfer function is 2T cos(πfT) which is called as cosine filter.

7. The method which has greater bandwidth efficiency is called as
a) Duobinary signalling
b) Polybinary signalling
c) Correlative coding
d) All of the mentioned

Answer:b
Explanation: If more than three levels are introduced in duobinary signalling technique the bandwidth efficiency increases This method is called as polybinary signalling.

8. In polybinary signalling method the present bit of binary sequence is algebraically added with ______ number of previous bits.
a) j
b) 2j
c) j+2
d) j-2

Answer: d
Explanation: In polybinary signalling method the present binary digit of the sequence is formed from the modulo-2 addition of the j-2 preceding digits of the sequence and the present digit.

9. The primary advantage of this method is
a) redistribution of spectral density
b) to favor low frequencies
c) redistribution of spectral density & to favor low frequencies
d) none of the mentioned

Answer: c
Explanation: Each bit can be independently detected in-spite of strong correlation and this provides redistribution of spectral density and also favors low frequencies.

10. Source encoding procedure does
a) Sampling
b) Quantization
c) Compression
d) All of the mentioned

Answer: d
Explanation: Source encoding includes a sampling of continuous time signals, quantization of continuous valued signals and compression of those sources.

TOPIC 3.6 EYE PATTERN, EQUALIZERS

1. The range of amplitude difference gives the value of
a) Width
b) Distortion
c) Timing jitter
d) Noise margin

Answer: b
Explanation: In the eye pattern, the amplitude difference gives the value of distortion caused by ISI.

2. As the eye opens, ISI _______
a) Increases
b) Decreases
c) Remains the same
d) None of the mentioned

Answer: b
Explanation: As the eye closes, ISI increases and as the eye opens ISI decreases.

3. Pseudo noise signal has _______ and _______ SNR for the same peak transmitted power.
a) Larger, smaller
b) Smaller, larger
c) Larger, larger
d) Smaller, smaller

Answer: c
Explanation: A training pulse is applied to the equalizer and corresponding impulse response is observed. Pseudo noise is preferred as the training pulse as it has larger SNR value and larger average power value.

4. The index value n, in transversal filter can be used as.
a) Time offset
b) Filter coefficient identifier
c) Time offset & Filter coefficient identifier
d) None of the mentioned

Answer: c
Explanation: The index n can be used as both time offset and the filter coefficient identifier, which is the address in the filter.

5. The over-determined set of equations can be solved using
a) Zero forcing
b) Minimum mean square error
c) Zero forcing & Minimum mean square error
d) None of the mentioned

Answer: c
Explanation: The matrix x in transversal equalizer if non square with dimensions 4N+1 and 2N+1. Such equations are called as over-determined set. This can be solved by two methods called as zero forcing method and minimum mean square error method.

6. If the filter's tap weight remains fixed during transmission of data, then the equalization is called as
a) Preset equalization
b) Adaptive equalization
c) Fixed equalization
d) None of the mentioned

Answer: a
Explanation: If the weight remains fixed during transmission of data then the equalization is called as preset equalization. It is a simple method which consists of setting the tap weight according to some average knowledge of the channel.

7. Equalization method which is done by tracking a slowly time varying channel response is
a) Preset equalization
b) Adaptive equalization
c) Variable equalization
d) None of the mentioned

Answer: b
Explanation: This method is implemented to perform tap weight adjustment periodically or continually. Equalization is done by tracking a slowly varying channel response.

8. Preamble is used for
a) Detect start of transmission
b) To set automatic gain control
c) To align internal clocks
d) All of the mentioned

Answer: d
Explanation: The receiver uses preamble for detecting the start of transmission, to set automatic gain control, and to align internal clocks and local oscillator with the received signal.

9. The disadvantage of preset equalizer is that
a) It doesnot requires initial training pulse
b) Time varying channel degrades the performance of the system
c) All of the mentioned
d) None of the mentioned

Answer: b
Explanation: The disadvantage of preset equalization is that it requires an initial training period that must be invoked at the start of any new transmission. Also time varying channel can degrade system performance due to ISI, since the tap weights are fixed.

10. For AWGN, the noise variance is
a) N0
b) N0/2
c) 2N0
d) N0/4

Answer: b
Explanation: The noise variance out of the correlator for AWGN is N0/2.

11. Performance of BFSK signal is ________ than BPSK.
a) 3db worse
b) 3db better
c) 6db worse
d) 6db better

Answer: a
Explanation: The performance of BFSK is 3db worse than BPSK signalling, since for a given signal power, the distance squared between orthogonal vectors is a factor of two less than the distance squared between orthopodal signals.

12. A Gaussian distribution into the non linear envelope detector yields
a) Rayleigh distribution
b) Normal distribution
c) Poisson distribution
d) Binary distribution

Answer: a
Explanation: The two output signals of Gaussian distribution yields Rayleigh and Rician distribution.

13. The non coherent FSK needs ________ Eb/N0 than coherent FSK.
a) 1db more
b) 1db less
c) 3db more
d) 3db less

Answer: a
Explanation: The non coherent receiver is easier to implement. The non coherent FSK needs 1db more Eb/N0 than coherent FSK.

14. The DPSK needs ________ Eb/N0 than BPSK.
a) 1db more
b) 1db less

c) 3db more
d) 3db less

Answer: a
Explanation: The DPSK system is easier to implement than PSK and it needs 1db more Eb/N0 than BPSK.

15. Coherent PSK and non coherent orthogonal FSK have a difference of _______ in PB.
a) 1db
b) 3db
c) 4db
d) 6db

Answer: c
Explanation: The difference of PB is approximately 4db for the best (coherent PSK) and the worst (non coherent orthogonal FSK).

16. Which is easier to implement and is preferred?
a) Coherent system
b) Non coherent system
c) Coherent & Non coherent system
d) None of the mentioned

Answer: b
Explanation: A non coherent system is desirable because there may be difficulty is establishing and maintaining a coherent reference.

17. Which is the main system consideration?
a) Probability of error
b) System complexity
c) Random fading channel
d) All of the mentioned

Answer: d
Explanation: The major system considerations are error probability, complexity and random fading channel. Considering all this non coherent system is more desirable than coherent.

UNIT IV INFORMATION THEORY AND CODING

TOPIC 4.1 MEASURE OF INFORMATION,ENTROPY

1. Self information should be
a) Positive
b) Negative
c) Positive & Negative
d) None of the mentioned

Answer: a
Explanation: Self information is always non negative.

2. The unit of average mutual information is
a) Bits
b) Bytes
c) Bits per symbol
d) Bytes per symbol

Answer: a
Explanation: The unit of average mutual information is bits.

3. When probability of error during transmission is 0.5, it indicates that
a) Channel is very noisy
b) No information is received
c) Channel is very noisy & No information is received
d) None of the mentioned

Answer: c
Explanation: When probability of error during transmission is 0.5 then the channel is very noisy and thus no information is received.

4. Binary Huffman coding is a
a) Prefix condition code
b) Suffix condition code

c) Prefix & Suffix condition code
d) None of the mentioned

Answer: a
Explanation: Binary Huffman coding is a prefix condition code.

5. The event with minimum probability has least number of bits.
a) True
b) False

Answer: b
Explanation: In binary Huffman coding the event with maximum probability has least number of bits.

6. The method of converting a word to stream of bits is called as
a) Binary coding
b) Source coding
c) Bit coding
d) Cipher coding

Answer: b
Explanation: Source coding is the method of converting a word to stream of bits that is 0's and 1's.

7. When the base of the logarithm is 2, then the unit of measure of information is
a) Bits
b) Bytes
c) Nats
d) None of the mentioned

Answer: a
Explanation: When the base of the logarithm is 2 then the unit of measure of information is bits.

8. When X and Y are statistically independent, then I (x,y) is
a) 1
b) 0
c) Ln 2
d) Cannot be determined

Answer: b
Explanation: When X and Y are statistically independent the measure of information I (x,y) is 0.

9. The self information of random variable is
a) 0
b) 1
c) Infinite
d) Cannot be determined

Answer: c
Explanation: The self information of a random variable is infinity.

10. Entropy of a random variable is
a) 0
b) 1
c) Infinite
d) Cannot be determined

Answer: c
Explanation: Entropy of a random variable is also infinity.

11. Which is more efficient method?
a) Encoding each symbol of a block
b) Encoding block of symbols
c) Encoding each symbol of a block & Encoding block of symbols
d) None of the mentioned

Answer: b
Explanation: Encoding block of symbols is more efficient than encoding each symbol of a block.

12. Lempel-Ziv algorithm is
a) Variable to fixed length algorithm
b) Fixed to variable length algorithm
c) Fixed to fixed length algorithm
d) Variable to variable length algorithm

Answer: a
Explanation: Lempel-Ziv algorithm is a variable to fixed length algorithm.

13. Coded system are inherently capable of better transmission efficiency than the

uncoded system.
a) True
b) False

Answer: a
Explanation: Yes, the coded systems are capable of better transmission efficiency than the uncoded system.

TOPIC 4.2 SOURCE CODING THEOREM

1. While recovering signal, which gets attenuated more?
a) Low frequency component
b) High frequency component
c) Low & High frequency component
d) None of the mentioned

Answer: b
Explanation: High frequency components are attenuated more when compared to low frequency components while recovering the signals.

2. Mutual information should be
a) Positive
b) Negative
c) Positive & Negative
d) None of the mentioned

Answer: c
Explanation: Mutual information can also be negative.

3. ASCII code is a
a) Fixed length code
b) Variable length code
c) Fixed & Variable length code
d) None of the mentioned

Answer: a
Explanation: ASCII code is a fixed length code. It has a fixed length of 7 bits.

4. Which reduces the size of the data?
a) Source coding
b) Channel coding
c) Source & Channel coding
d) None of the mentioned

Answer: a
Explanation: Source coding reduces the size of the data and channel coding increases the size of the data.

5. In digital image coding which image must be smaller in size?
a) Input image
b) Output image
c) Input & Output image
d) None of the mentioned

Answer: b
Explanation: In digital image coding, output image must be smaller than the input image.

6. Which coding method uses entropy coding?
a) Lossless coding
b) Lossy coding
c) Lossless & Lossy coding
d) None of the mentioned

Answer: b
Explanation: Lossy source coding uses entropy coding.

7. Which achieves greater compression?
a) Lossless coding
b) Lossy coding
c) Lossless & Lossy coding
d) None of the mentioned

Answer: b
Explanation: Lossy coding achieves greater compression where as lossless coding achieves only moderate compression.

8. A code is a mapping from
a) Binary sequence to dicrete set of symbols
b) Discrete set of symbols to binary sequence
c) All of the mentioned
d) None of the mentioned

Answer: b
Explanation: A code is a mapping from discrete set of symbols to finite binary sequence.

9. Which are uniquely decodable codes?
a) Fixed length codes
b) Variable length codes
c) Fixed & Variable length codes
d) None of the mentioned

Answer: a
Explanation: Fixed length codes are uniquely decodable codes where as variable length codes may or may not be uniquely decodable.

10. A rate distortion function is a
a) Concave function
b) Convex function
c) Increasing function
d) None of the mentioned

Answer: b
Explanation: A rate distortion function is a monotone decreasing function and also a convex function.

TOPIC 4.3 HUFFMAN CODING

1. Which of the following algorithms is the best approach for solving Huffman codes?
a) exhaustive search
b) greedy algorithm
c) brute force algorithm
d) divide and conquer algorithm

Answer: b
Explanation: Greedy algorithm is the best approach for solving the Huffman codes problem since it greedily searches for an optimal solution.

2. How many printable characters does the ASCII character set consists of?
a) 120
b) 128
c) 100
d) 98

Answer: c
Explanation: Out of 128 characters in an ASCII set, roughly, only 100 characters are printable while the rest are non-printable.

3. Which bit is reserved as a parity bit in an ASCII set?
a) first
b) seventh
c) eighth
d) tenth

Answer: c
Explanation: In an ASCII character set, seven bits are reserved for character representation while the eighth bit is a parity bit.

4. How many bits are needed for standard encoding if the size of the character set is X?
a) log X
b) X+1
c) 2X
d) X^2

Answer: a
Explanation: If the size of the character set is X, then [log X] bits are needed for representation in a standard encoding.

5. The code length does not depend on the frequency of occurrence of characters.
a) true
b) false

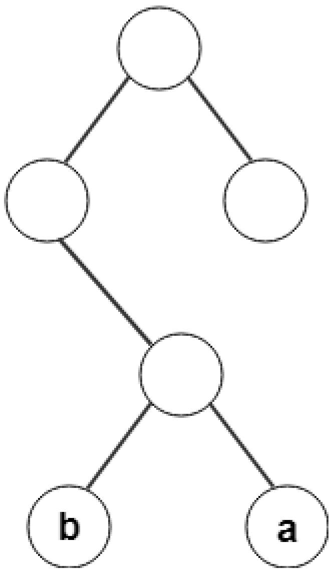
Answer: b
Explanation: The code length depends on the frequency of occurrence of characters. The more frequent the character occurs, the less is the length of the code.

6. In Huffman coding, data in a tree always occur?
a) roots
b) leaves

c) left sub trees
d) right sub trees

Answer: b
Explanation: In Huffman encoding, data is always stored at the leaves of a tree inorder to compute the codeword effectively.

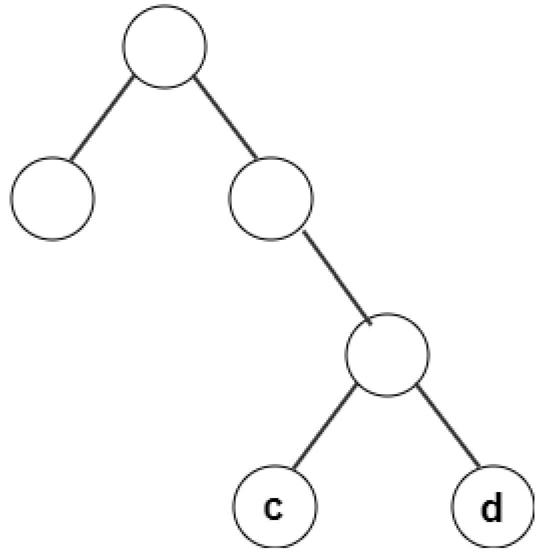
7. From the following given tree, what is the code word for the character 'a'?



a) 011
b) 010
c) 100
d) 101

Answer: a
Explanation: By recording the path of the node from root to leaf, the code word for character 'a' is found to be 011.

8. From the following given tree, what is the computed codeword for 'c'?



a) 111
b) 101
c) 110
d) 011

Answer: c
Explanation: By recording the path of the node from root to leaf, assigning left branch as 0 and right branch as 1, the codeword for c is 110.

9. What will be the cost of the code if character c_i is at depth d_i and occurs at frequency f_i?
a) c_if_i
b) $\int c_if_i$
c) $\sum f_id_i$
d) f_id_i

Answer: c
Explanation: If character c_i is at depth d_i and occurs at frequency f_i, the cost of the codeword obtained is $\sum f_id_i$.

10. An optimal code will always be present in a full tree.
a) true
b) false

Answer: a
Explanation: An optimal tree will always

have the property that all nodes are either leaves or have two children. Otherwise, nodes with one child could move up a level.

11. The type of encoding where no character code is the prefix of another character code is called?
a) optimal encoding
b) prefix encoding
c) frequency encoding
d) trie encoding

Answer: b
Explanation: Even if the character codes are of different lengths, the encoding where no character code is the prefix of another character code is called prefix encoding.

12. What is the running time of the Huffman encoding algorithm?
a) O(C)
b) O(log C)
c) O(C log C)
d) O(N log C)

Answer: c
Explanation: If we maintain the trees in a priority queue, ordered by weight, then the running time is given by O(C log C).

13. What is the running time of the Huffman algorithm, if its implementation of the priority queue is done using linked lists?
a) O(C)
b) O(log C)
c) O(C log C)
d) O(C^2)

Answer: d
Explanation: If the implementation of the priority queue is done using linked lists, the running time of Huffman algorithm is O(C^2).

TOPIC 4.4 CHANNEL CAPACITY

1. Notch is a ________
a) High pass filter
b) Low pass filter
c) Band stop filter
d) Band pass filter

Answer: c
Explanation: Notch filter is a band stop filter that allows most frequencies to pass through it, except frequencies in a specific range. It is just opposite of a band-pass filter. High pass filter allows higher frequencies to pass while Low pass filter allows lower frequencies to pass through it.

2. Sin wave is ________
a) Aperiodic Signal
b) Periodic Signal
c) Random Signal
d) Deterministic Signal

Answer: b
Explanation: Periodic signal is that which repeats itself after a regular interval. Sin wave is a periodic function since it's value can be determined at any point of time, as it repeats itself at a regular interval. Aperiodic Signal does not repeat itself at regular interval of time. Random signals are the signals which have uncertain values at any time. While Deterministic signals are the signals which are constant over a period of time.

3. What is the role of channel in communication system?
a) acts as a medium to send message signals from transmitter to receiver
b) converts one form of signal to other
c) allows mixing of signals
d) helps to extract original signal from incoming signal

Answer: a
Explanation: Channel acts as a medium to transmit message signal from source transmitter to the destination receiver. Transducer converts a signal from one form of energy to other. Mixer allows mixing of

signals while Demodulator helps to extract original message signal from incoming signal.

4. Sum of a periodic and aperiodic signal always be an aperiodic signal.
a) True
b) False

Answer: b
Explanation: Periodic signal is a signal which repeats itself after a regular interval. While Aperiodic Signal does not repeat itself at regular interval of time.
For example: Let f(x) = sin(x), be a periodic function with period 2π and g(x) = −sin(x) + sin(√2x), be an aperiodic function. Now the sum of both i.e. f(x) + g(x) = sin(√2x), which is a periodic function.
Therefore, the sum of a periodic and aperiodic signal can be periodic.

5. Noise is added to a signal ________
a) In the channel
b) At receiving antenna
c) At transmitting antenna
d) During regeneration of information

Answer: a
Explanation: Noise is an unwanted signal that gets mixed with the transmitted signal while passing through the channel. The noise interferes with the signal and provides distortion in received signal. The transmitting antenna transmits modulated message signal while the receiving antenna receives the transmitted signal. Regeneration of information refers to demodulating the received signal to produce the original message signal.

6. Agreement between communication devices are called ________
a) Transmission medium
b) Channel
c) Protocol
d) Modem

Answer: c
Explanation: Protocol is a set of rules that looks after data communication, by acting as an agreement between communication devices. Channel is the transmission medium or the path through which information travels. Modem is a device that modulates and demodulates data.

7. What is the advantage of superheterodyning?
a) High selectivity and sensitivity
b) Low Bandwidth
c) Low adjacent channel rejection
d) Low fidelity

Answer: a
Explanation: The main advantage of superheterodyning is that it provides high selectivity and sensitivity. It's bandwidth remains same. It has high adjacent channel rejection and high fidelity.

8. Low frequency noise is ________
a) Flicker noise
b) Shot noise
c) Thermal noise
d) Partition Noise

Answer: a
Explanation: Flicker noise is a type of electronic noise which is generated due to fluctuations in the density of carrier. It's also known as 1/f as it's power spectral density increases with a decrease in frequency or increase in offset from a signal.

9. Relationship between amplitude and frequency is represented by ________
a) Time-domain plot
b) Phase-domain plot
c) Frequency-domain plot
d) Amplitude-domain plot

Answer: c
Explanation: Relationship between amplitude and frequency is represented by a frequency-domain plot. Also, it represents the

relation between phase and frequency. While a time-domain plot shows how a signal varies over time.

10. A function f(x) is even, when?
a) f(x) = -f(x)
b) f(x) = f(-x)
c) f(x) = -f(x)f(-x)
d) f(x) = f(x)f(-x)

Answer: b
Explanation: Geometrically a function f(x) is even, if plot of the function is symmetric over y-axis. Algebraically, for any function f(x) to be even, f(x) = f(-x).
While for a function f(x) to be odd, f(x) = -f(-x).

TOPIC 4.5 SHANNON-HARTLEY LAW

1. The minimum nyquist bandwidth needed for baseband transmission of Rs symbols per second is
a) Rs
b) 2Rs
c) Rs/2
d) Rs2

Answer: c
Explanation: Theoretical minimum nyquist bandwidth needed for the baseband transmission of Rs symbols per second without ISI is Rs/2.

2. The capacity relationship is given by
a) C = W log2 (1+S/N)
b) C = 2W log2 (1+S/N)
c) C = W log2 (1-S/N)
d) C = W log10 (1+S/N)

Answer: a
Explanation: The capacity relationship from Shannon-hartley capacity theorem is given by C = W log2 (1+S/N).

3. Which parameter is called as Shannon limit?
a) PB/N0
b) EB/N0
c) EBN0
d) None of the mentioned

Answer: b
Explanation: There exists a limiting value for EB/N0 below which they can be no error free communication at any information rate. This EB/N0 is called as Shannon limit.

4. Entropy is the measure of
a) Amount of information at the output
b) Amount of information that can be transmitted
c) Number of error bits from total number of bits
d) None of the mentioned

Answer: a
Explanation: Entropy is defined as the average amount of information per source output.

5. Equivocation is the
a) Conditional entropy
b) Joint entropy
c) Individual entropy
d) None of the mentioned

Answer: a
Explanation: Shannon uses a correction factor called equivocation to account for uncertainty in the received signal. It is defined as the conditional entropy of the message X given Y.

6. For a error free channel, conditional probability should be
a) Zero
b) One
c) Equal to joint probability
d) Equal to individual probability

Answer: a
Explanation: For a error free channel

conditional probability should be zero, because having received Y there is complete certainty about the message X.

7. Average effective information is obtained by
a) Subtracting equivocation from entropy
b) Adding equivocation with entropy
c) Ratio of number of error bits by total number of bits
d) None of the mentioned

Answer: a
Explanation: According to Shannon the average effective information is obtained by subtracting the equivocation from the entropy of the source.

8. Turbo codes are
a) Forward error correction codes
b) Backward error correction codes
c) Error detection codes
d) None of the mentioned

Answer: a
Explanation: Turbo codes are a class of high performance forward error correction codes.

9. Components used for generation of turbo codes are
a) Inter leavers
b) Punching pattern
c) Inter leavers & Punching pattern
d) None of the mentioned

Answer: c
Explanation: There are many instances of turbo codes, using different component encoders, input/output ratios, inter leavers, punching patterns.

10. Decoders are connected in series.
a) True
b) False

Answer: a
Explanation: Two elementary decoders are connected in serial connection for decoding the turbo codes.

11. The inter leaver connected between the two decoders is used to
a) Remove error bursts
b) Scatter error bursts
c) Add error bursts
d) None of the mentioned

Answer: b
Explanation: An inter leaver installed between the two decoders connected in series is used to scatter error bursts.

12. In soft decision approach what does -127 mean?
a) Certainly one
b) Certainly zero
c) Very likely zero
d) Very likely one

Answer: b
Explanation: The decoder front end produces an integer for each bit in the data stream. This integer is the measure of how likely it is that the bit 0 or 1 and is called as soft bit. It ranges from -127 to 127. Here -127 represents certainly zero.

13. In soft decision approach 100 means?
a) Certainly one
b) Certainly zero
c) Very likely zero
d) Very likely one

Answer: d
Explanation: The decoder front end produces an integer for each bit in the data stream. This integer is the measure of how likely it is that the bit 0 or 1 and is called as soft bit. It ranges from -127 to 127. Here 100 represents very likely one.

14. In soft decision approach 0 represents
a) Certainly one
b) Certainly zero

c) Very likely zero
d) Could be either zero or one

Answer: d
Explanation: The decoder front end produces an integer for each bit in the data stream. This integer is the measure of how likely it is that the bit 0 or 1 and is called as soft bit. It ranges from -127 to 127. Here 0 represents 'could be either zero or one'.

TOPIC 4.6 ERROR CONTROL CODES – CYCLIC CODES, SYNDROME CALCULATION

1. In layering, n layers provide service to
a) n layer
b) n-1 layer
c) n+1 layer
d) none of the mentioned

Answer: c
Explanation: In layering n layer provides service to n+1 layer and use the service provided by n-1 layer.

2. Which can be used as an intermediate device in between transmitter entity and receiver entity?
a) IP router
b) Microwave router
c) Telephone switch
d) All of the mentioned

Answer: d
Explanation: IP router, microwave router and telephone switch can be used as an intermediate device between communication of two entities.

3. Which has comparatively high frequency component?
a) Sine wave
b) Cosine wave
c) Square wave
d) None of the mentioned

Answer: c
Explanation: Square wave has comparatively high frequency component in them.

4. Which has continuous transmission?
a) Asynchronous
b) Synchronous
c) Asynchronous & Synchronous
d) None of the mentioned

Answer: b
Explanation: Synchronous has continuous transmission where as asynchronous have sporadic transmission.

5. Which requires bit transitions?
a) Asynchronous
b) Synchronous
c) Asynchronous & Synchronous
d) None of the mentioned

Answer: b
Explanation: Synchronous transmission needs bit transition.

6. In synchronous transmission, receiver must stay synchronous for
a) 4 bits
b) 8 bits
c) 9 bits
d) 16 bits

Answer: c
Explanation: In synchronous transmission, receiver must stay synchronous for 9 bits.

7. How error detection and correction is done?
a) By passing it through equalizer
b) By passing it through filter
c) By amplifying it
d) By adding redundancy bits

Answer: d
Explanation: Error can be detected and corrected by adding additional information that is by adding redundancy bits.

8. Which is more efficient?
a) Parity check
b) Cyclic redundancy check
c) Parity & Cyclic redundancy check
d) None of the mentioned

Answer: b
Explanation: Cyclic redundancy check is more efficient than parity check.

9. Which can detect two bit errors?
a) Parity check
b) Cyclic redundancy check
c) Parity & Cyclic redundancy check
d) None of the mentioned

Answer: b
Explanation: CRC is more powerful and it can detect various kind of errors like 2 bit errors.

10. CRC uses
a) Multiplication
b) Binary division
c) Multiplication & Binary division
d) None of the mentioned

Answer: c
Explanation: CRC uses more math like multiplication and binary division.

TOPIC 4.7 CONVOLUTION CODING, SEQUENTIAL AND VITERBI DECODING

1. Block codes can achieve a larger coding gain than convolution coding.
a) True
b) False

Answer: b
Explanation: Convolution code can achieve a larger coding gain that can be achieved using a block coding with the same complexity. Their mapping is highly structured, enabling a decoding method considerably different from block codes.

2. Which of the following indicates the number of input bits that the current output is dependent upon?
a) Constraint length
b) Code length
c) Search window
d) Information rate

Answer: a
Explanation: Constraint length determines the number of input data bits that the current output is dependent upon. The constraint length determines how powerful and complex the code is.

3. Which of the following is not a way to represent convolution code?
a) State diagram
b) Trellis diagram
c) Tree diagram
d) Linear matrix

Answer: d
Explanation: Linear matrix is not a way to represent convolution code. Various ways of representing convolution codes are generator matrix, generator polynomial, logic tables, state diagram, tree diagram and trellis diagram.

4. Which of the following is not an algorithm for decoding convolution codes?
a) Viterbi algorithm
b) Stack algorithm
c) Fano's sequential coding
d) Ant colony optimization

Answer: d
Explanation: There are a number of techniques for decoding convolution codes. The most important of these methods is Viterbi algorithm. Other decoding algorithms for convolutional codes are Fano's sequential coding, stack algorithm and feedback coding.

5. Viterbi algorithm performs ____________ decoding of convolutional codes.
a) Maximum likelihood

b) Maximum a posteriori
c) Minimum square
d) Minimum mean square

Answer: a
Explanation: Viterbi algorithm performs maximum likelihood decoding of convolutional codes. The algorithm was first developed by A.J. Viterbi. It is one of the most important algorithm used for decoding convolutional codes.

6. Fano's algorithm searches all the paths of trellis diagram at same time to find the most probable path.
a) True
b) False

Answer: b
Explanation: Fano's algorithm searches for the most probable path through the trellis diagram by examining one path at a time. The error rate performance of Fano's algorithm is comparable to Viterbi's algorithm.

7. Which of the following is not an advantage of Fano's algorithm in comparison to Viterbi's algorithm?
a) Less storage
b) Large constraint length
c) Error rate
d) Small delays

Answer: d
Explanation: In comparison to Viterbi decoding, sequential decoding has a significantly larger delay. In advantage over Viterbi decoding is that it requires less storage, and thus codes with larger constraint lengths can be employed.

8. In comparison to stack algorithm, Fano's algorithm is simpler.
a) True
b) False

Answer: b
Explanation: In comparison to Fano's algorithm, the stack algorithm is computationally simpler since there is no retracting over the same path. But stack algorithm requires more storage than Fano's algorithm.

9. Which of the following is not an error correction and detection code?
a) Block code
b) Convolutional codes
c) Passive codes
d) Turbo codes

Answer: c
Explanation: There are three basic types of error correction and detection codes. They are block codes, convolutional codes and turbo codes. A channel coder operates on digital message data by encoding the source information into a code sequence.

10. Which decoding method involves the evaluation by means of Fano's algorithm?
a) Maximum Likelihood Decoding
b) Sequential Decoding
c) Maximum a priori
d) Minimum mean square

Answer: b
Explanation: Fano's algorithm involves sequential decoding. It searches for the most probable path through the trellis by examining one path at a time.

11. In Viterbi's algorithm, the selected paths are regarded as ___________
a) Survivors
b) Defenders
c) Destroyers
d) Carriers

Answer: a
Explanation: In Viterbi's algorithm, the selected paths are regarded as survivors. The path thus defined is unique and corresponds to the decoded output.

UNIT V SPREAD SPECTRUM AND MULTIPLE ACCESS

TOPIC 5.1 PN SEQUENCES

1. Pseudorandom signal ________ predicted.
a) Can be
b) Cannot be
c) maybe
d) None of the mentioned

Answer: a
Explanation: Random signals cannot be predicted whereas pseudorandom sequence can be predicted.

2. The properties used for pseudorandom sequence are
a) Balance
b) Run
c) Correlation
d) All of the mentioned

Answer: d
Explanation: The three basic properties that can be applied for pseudorandom sequence are balance, run and correlation properties.

3. The shift register needs to be controlled by clock pulses.
a) True
b) False

Answer: a
Explanation: The shift register operation is controlled by clock pulses.

4. A linear feedback shift register consists of
a) Feedback path
b) Modulo 2 adder
c) Four stage register
d) All of the mentioned

Answer: d
Explanation: A linear feedback shift register consists of four stage register for storage and shifting, modulo 2 adder and feedback path.

5. If the initial pulse of 1000 is fed to shift register, after how many clock pulses does the sequence repeat?
a) 15
b) 16
c) 14
d) 17

Answer: a
Explanation: If the initial pulse 1000 is given to shift register, the foregoing sequence repeats after 15 clock pulses.

6. The sequences produced by shift register depends on
a) Number of stages
b) Feedback tap connections
c) Initial conditions
d) All of the mentioned

Answer: d
Explanation: The sequences produced by shift register depends on the number of stages, the feedback tap connections and initial conditions.

7. For maximal length sequence, the sequence repetition clock pulses p is given by
a) 2n + 1
b) 2n -1
c) 2n
d) None of the mentioned

Answer: b
Explanation: For maximal length sequence, produced by n stage linear feedback shift register the sequence repetition clock pulses p is given by 2n -1 .

8. For any cyclic shift, the auto-correlation function is equal to
a) 1/p
b) -1/p
c) –p
d) p

Answer: b
Explanation: For any cyclic shift the auto-correlation function is equal to -1/p.

9. Which method is better?
a) To share same bandwidth
b) To share different bandwidth
c) To share same & different bandwidth
d) None of the mentioned

Answer: b
Explanation: If the jammer noise shares the same bandwidth, the result could be destructive.

10. Pulse jammer consists of
a) Pulse modulated excess band noise
b) Pulse modulated band-limited noise
c) Pulse width modulated excess band noise
d) Pulse width modulated band-limited noise

Answer: b
Explanation: Pulse jammer consists of pulse modulated band-limited noise.

11. Which are the design options for anti jam communicator?
a) Time diversity
b) Frequency diversity
c) Special discrimination
d) All of the mentioned

Answer: d
Explanation: The design options for anti-jam communicator are time diversity, frequency diversity and special discrimination.

12. The ratio (J/S)reqd gives the measure of
a) Vulnerability to interference
b) Invulnerability to interference
c) All of the mentioned
d) None of the mentioned

Answer: b
Explanation: The ratio (J/S)reqd gives the measure of how invulnerable the system is to interference.

13. The system should have
a) Larger (J/S)reqd
b) Greater system's noise rejection capability
c) Larger (J/S)reqd & Greater system's noise rejection capability
d) None of the mentioned

Answer: c
Explanation: The system will be efficient if it has greater (J/S)reqd and larger system's noise rejection capability.

14. The broadband jammer jams the entire
a) W
b) Wss
c) W & Wss
d) None of the mentioned

Answer: b
Explanation: The broadband jammer or wide-band jammer is the one which jams the entire Wss with its fixed power.

15. To increase error probability, the processing gain should be
a) Increased
b) Decreased
c) Exponentially increased
d) Exponentially decreased

Answer: a
Explanation: In a system, to increase the error probability the processing gain should be increased.

16. Which jamming method produces greater degradation?
a) Broadband jamming
b) Partial jamming
c) Broadband & Partial jamming
d) None of the mentioned

Answer: b
Explanation: Greater degradation is possible more with partial jamming than broadband jamming.

17. The jammer which monitors a communicator's signal is known as
a) Frequency follower jammers
b) Repeat back jammers
c) Frequency follower & Repeat back jammers
d) None of the mentioned

Answer: c
Explanation: The smart jammers which monitor a communicator's signals is known as frequency follower or repeat back jammers.

TOPIC 5.2 DSSS, FHSS

1. DS/BPSK includes
a) Despreading
b) Demodulation
c) Despreading & Demodulation
d) None of the mentioned

Answer: c
Explanation: DS/BPSK is a two step precess which includes despreading and demodulation.

2. In direct sequence process which step is performed first?
a) De-spreading
b) Demodulation
c) Despreading & Demodulation
d) None of the mentioned

Answer: a
Explanation: In direct sequence process, De-spreading correlator is followed by a modulator.

3. The processing gain is given as
a) Wss/R
b) R/Wss
c) Wss/2R
d) R/2Wss

Answer: a
Explanation: The processing gain is given by the ratio of the minimum bandwidth of the data to data rate.

4. Chip is defined as
a) Shortest uninterrupted waveform
b) Largest uninterrupted waveform
c) Shortest diversion
d) None of the mentioned

Answer: a
Explanation: A chip is defined as the shortest uninterrupted waveform in the system.

5. Processing gain is given as
a) Wss/R
b) Rch/R
c) Wss/R & Rch/R
d) None of the mentioned

Answer: c
Explanation: Processing gain is given as both as the ratio of the minimum bandwidth of the data to data rate and also the by the ratio of code chip rate and data rate as minimum bandwidth is approximately equal to code chip rate.

6. Which modulation scheme is preferred for direct sequence spread spectrum process?
a) BPSK
b) QPSK
c) BPSK & QPSK
d) None of the mentioned

Answer: c
Explanation: Both the modulation scheme BPSK and QPSK can be used for direct sequence spread spectrum process.

7. The frequency hopping system uses _______ modulation scheme.
a) FSK
b) BPSK
c) MFSK
d) MPSK

Answer: c
Explanation: The frequency hopping spread

spectrum system uses M-ary frequency shift keying modulation scheme.

8. The minimum spacing between consecutive hop positions gives the
a) Minimum number of chips necessary
b) Maximum number of chips necessary
c) Chip rate
d) None of the mentioned

Answer: a
Explanation: The minimum spacing between consecutive hop positions given the minimum number of chips necessary in the frequency word.

9. Which system allows larger processing gain?
a) Direct sequence
b) Frequency hopping
c) Direct sequence & Frequency hopping
d) None of the mentioned

Answer: b
Explanation: Frequency hopping spread spectrum system allows greater processing gain than direct sequence spread spectrum technique.

10. In which technique is phase coherence hard to maintain?
a) Direct sequence
b) Frequency hopping
c) Direct sequence & Frequency hopping
d) None of the mentioned

Answer: b
Explanation: In frequency hopping spread spectrum phase coherence is hard to maintain from hop to hop.

11. Which type of demodulator is used in the frequency hopping technique?
a) Coherent
b) Non coherent
c) Coherent & Non coherent
d) None of the mentioned

Answer: b
Explanation: As it is difficult to maintain phase coherence, non coherent demodulator is used.

12. Robustness gives the inability of a signal to withstand the impairments.
a) True
b) False

Answer: b
Explanation: Robustness gives the ability of a signal to withstand the impairments such as noise, jamming etc.

13. Chips are the
a) Repeated symbols
b) Non repeated symbols
c) Smallest length symbols
d) None of the mentioned

Answer: a
Explanation: The repeated symbols are called as chips.

14. Slow frequency hopping is
a) Several hops per modulation
b) Several modulations per hop
c) Several symbols per modulation
d) None of the mentioned

Answer: b
Explanation: Slow frequency hopping is several modulation per frequency hop.

15. Fast frequency hopping is
a) Several modulations per hop
b) Several modulations per symbol
c) Several symbols per modulation
d) None of the mentioned

Answer: d
Explanation: Fast frequency hopping is several frequency hops per modulation.

16. Which duration is shorter?
a) Hop duration
b) Symbol duration

c) Chip duration
d) None of the mentioned

Answer: a
Explanation: In frequency hopping technique hop duration is shorter than the symbol duration.

TOPIC 5.3 MULTIPLE ACCESS – FDMA (FREQUENCY DIVISION MULTIPLE ACCESS)

1. Frequency division multiple access (FDMA) assigns ______ channels to _______ users.
a) Individual, individual
b) Many, individual
c) Individual, many
d) Many, many

Answer: a
Explanation: Frequency division multiple access (FDMA) assigns individual channels to individual users. Each user is allocated a unique frequency band or channel. These channels are assigned on demand to users who request service.

2. During the period of call, other users can share the same channel in FDMA.
a) True
b) False

Answer: b
Explanation: In FDMA systems, no other user can share the same channel during the period of call. In FDD systems, the users are assigned a channel as a pair of frequencies; one is used for the forward channel while the other frequency is used for the reverse channel.

3. The FDMA channel carries ____________ phone circuit at a time.
a) Ten
b) Two
c) One
d) Several

Answer: c
Explanation: The FDMA channel carries one phone circuit at a time. Each individual band or channel is wide enough to accommodate the signal spectra of the transmissions to be propagated.

4. If the FDMA channel is not in use, it can be used by other users.
a) True
b) False

Answer: b
Explanation: If an FDMA channel is not in use, it sits idle and cannot be used by other users to increase or share capacity. It is essentially a wasted resource.

5. The bandwidth of FDMA channel is _______
a) Wide
b) Narrow
c) Large
d) Zero

Answer: b
Explanation: The bandwidth of FDMA channels is relatively narrow as each channel supports only one circuit per carrier. That is, FDMA is usually implemented in narrow band systems.

6. The symbol time in FDMA systems is _________ thus intersymbol interference is _______
a) Large, high
b) Small, low
c) Small, high
d) Large, low

Answer: d
Explanation: The symbol time of a narrowband signal is large as compared to the average delay spread. This implies that the amount of intersymbol interference is low

and, thus, little or no equalization is required in FDMA narrowband systems.

7. Due to _________ transmission scheme __________ bits are needed for overhead in FDMA systems.
a) Continuous, few
b) Discontinuous, few
c) Continuous, many
d) Discontinuous, many

Answer: a
Explanation: Since FDMA is a continuous transmission scheme, fewer bits are needed for overhead purposes (such as synchronization and framing bits) as compared to TDMA.

8. Which of the following is not true for FDMA systems as compared to TDMA systems?
a) Low complexity
b) Lower cell site system cost
c) Tight RF filtering
d) Narrow bandwidth

Answer: b
Explanation: FDMA systems have higher cell site system costs as compared to TDMA systems. It is due to single channel per carrier design, and the need to use costly bandpass filters to eliminate spurious radiation at the base station.

9. __________ is undesired RF radiation.
a) Intermodulation frequency
b) Intermediate frequency
c) Instantaneous frequency
d) Instrumental frequency

Answer: a
Explanation: Intermodulation (IM) frequency is undesired RF radiation which can interfere with other channels in the FDMA systems. The nonlinearities cause signal spreading in the frequency domain and generate IM frequency.

10. __________ is based on FDMA/FDD.
a) GSM
b) W-CDMA
c) Cordless telephone
d) AMPS

Answer: d
Explanation: The first US analog cellular system, the Advanced Mobile Phone System (AMPS) is based on FDMA/FDD. A single user occupies a single channel while the call is in progress.

11. In US AMPS, 416 channels are allocated to various operators with 10 kHz guard band and channel between them is 30 kHz. What is the spectrum allocation given to each operator?
a) 12.5 kHz
b) 30 kHz
c) 12.5 MHz
d) 30 MHz

Answer: c
Explanation: Spectrum allocated to each cellular operator is 12.5 MHz. As $B_t = NB_c + 2B_{guard}$; which is equal to $416*30*10^3+2(10*10^3) = 12.5$ MHz.

TOPIC 5.4 MULTIPLE ACCESS - TDMA (TIME DIVISION MULTIPLE ACCESS)

1. TDMA systems transmit in a continuous way.
a) True
b) False

Answer: b
Explanation: TDMA systems transmit data in a buffer and burst method. Thus, the transmission for any user is not continuous.

2. Preamble contains __________
a) Address
b) Data

c) Guard bits
d) Trail bits

Answer: a
Explanation: TDMA frame is made up of a preamble, an information message and the trail bits. In a TDMA frame, the preamble contains the address and synchronization information that both the base station and the subscribers use to identify each other.

3. __________ are utilized to allow synchronization of the receivers between different slots and frames.
a) Preamble
b) Data
c) Guard bits
d) Trail bits

Answer: c
Explanation: Guard times are utilized to allow synchronization of the receivers between different slots and frames. TDMA/FDD systems intentionally induce several time slots of delay between the forward and reverse time slots for a particular user.

4. Which of the following is not true for TDMA?
a) Single carrier frequency for single user
b) Discontinuous data transmission
c) No requirement of duplexers
d) High transmission rates

Answer: a
Explanation: TDMA share a single carrier frequency with several users, where each user makes use of non-overlapping time slots. The number of time slots per frame depends on several factors, such as modulation technique, available bandwidth etc.

5. Because of _______ transmissions in TDMA, the handoff process in __________
a) Continuous, complex
b) Continuous, simple
c) Discontinuous, complex
d) Discontinuous, simple

Answer: d
Explanation: Because of discontinuous transmissions in TDMA, the handoff process is much simpler for a subscriber unit, since it is able to listen for other base stations during idle time slots.

6. __________ synchronization overhead is required in TDMA due to _______ transmission.
a) High, burst
b) High, continuous
c) Low, burst
d) No, burst

Answer: a
Explanation: High synchronization overhead is required in TDMA systems because of burst transmissions. TDMA transmissions are slotted, and this requires the receivers to be synchronized for each data burst.

7. TDMA allocates a single time per frame to different users.
a) True
b) False

Answer: b
Explanation: TDMA has an advantage that it can allocate different numbers of time slots per frame to different users. Thus, bandwidth can be supplied on demand to different users by concatenating or reassigning time slots based on priority.

8. ___________ of TDMA system is a measure of the percentage of transmitted data that contains information as opposed to providing overhead for the access scheme.
a) Efficiency
b) Figure of merit
c) Signal to noise ratio
d) Mean

Answer: a
Explanation: Efficiency of TDMA system is a measure of the percentage of transmitted data that contains information as opposed to providing overhead for the access scheme. The frame efficiency is the percentage of bits per frame which contain transmitted data.

9. A TDMA system uses 25 MHz for the forward link, which is broken into radio channels of 200 kHz. If 8 speech channels are supported on a single radio channel, how many simultaneous users can be accommodated?
a) 25
b) 200
c) 1600
d) 1000

Answer: d
Explanation: For a TDMA system that uses 25 MHz for the forward link, which is broken into radio channels of 200 kHz. If 8 speech channels are supported on a single radio channel, 1000 simultaneous users can be accommodated as N = (25 MHz)/(200 kHz/8).

10. What is the time duration of a bit if data is transmitted at 270.833 kbps in the channel?
a) 270.833 s
b) 3 μs
c) 3.692 μs
d) 3.692 s

Answer: c
Explanation: If data is transmitted at 270.833 kbps in the channel, the time duration of a bit will be 3.692 μs, as T_b = (1/270.833 kbps) = 3.692 μs.

TOPIC 5.5 MULTIPLE ACCESS - CDMA (CODE DIVISION MULTIPLE ACCESS)

1. US digital cellular system based on CDMA was standardized as ________
a) IS-54
b) IS-136
c) IS-95
d) IS-76

Answer: c
Explanation: A US digital cellular system based on CDMA was standardized as Interim Standard 95 (IS-95). It was standardized by US Telecommunication Industry Association (TIA) and promised increased capacity.

2. IS-95 was not compatible with existing AMPS frequency band.
a) True
b) False

Answer: b
Explanation: Like IS-136, IS-95 system was designed to be compatible with the existing US analog cellular system (AMPS) frequency band. Hence, mobile and base stations can be economically produced for dual mode operation.

3. Which of the following is used by IS-95?
a) DSSS
b) FHSS
c) THSS
d) Hybrid

Answer: a
Explanation: IS-95 uses a direct sequence spread spectrum CDMA system. It allows each user within a cell to use the same radio channel, and users in adjacent cell also use the same radio channel.

4. Each IS-95 channel occupies ___________ of spectrum on each one way link.
a) 1.25 MHz
b) 1.25 kHz
c) 200 kHz
d) 125 kHz

Answer: a
Explanation: To facilitate graceful transition from AMPS to CDMA, each IS-95 channel occupies 1.25 MHz of spectrum on each one way link, or 10% of the available cellular spectrum for a US cellular provider.

5. IS-95 uses same modulation technique for forward and reverse channel.
a) True
b) False

Answer: b
Explanation: IS-95 uses different modulation and spreading technique for the forward and reverse links. On the forward link, the base station simultaneously transmits the user data for all mobiles in the cell by using different spreading sequence for each mobile.

6. IS-95 is specified for reverse link operation in _________ band.
a) 869-894 MHz
b) 849-894 MHz
c) 849-869 MHz
d) 824-849 MHz

Answer: d
Explanation: IS-95 is specified for reverse link operation in the 824-849 MHz band and 869-894 MHz for the forward link. The PCS version of IS-95 has also been designed for international use in the 1800-2000 MHz bands.

7. User data in IS-95 is spread to a channel chip rate of ________
a) 1.2288 Mchip/s
b) 9.6 Mchip/s
c) 12.288 Mchip/s
d) 0.96 Mchip/s

Answer: a
Explanation: User data is spread to a channel chip rate of 1.2288 Mchip/s (a total spreading factor of 128) using a combination of techniques. The spreading process is different for the forward and reverse links in the original CDMA specification.

8. __________ are used to resolve and combine multipath components.
a) Equalizer
b) Registers
c) RAKE receiver
d) Frequency divider

Answer: c
Explanation: At both the base station and the subscriber, RAKE receivers are used to resolve and combine multipath components, thereby reducing the degree of fading. A RAKE receiver exploits the multipath time delays in a channel and combines the delayed replicas of transmitted signal.

9. CT2 was the first generation of cordless telephones.
a) True
b) False

Answer: b
Explanation: CT2 was the second generation of cordless telephones introduced in Great Britain in 1989. It is used to provide telepoint services which allow a subscriber to use CT2 handsets at a public telepoint.

10. CT2 is analog version of first generation cordless telephones.
a) True
b) False

Answer: b
Explanation: CT2 is a digital version of the first generation, analog, cordless telephones. When compared with analog cordless phones, CT2 offers good speech quality and is more resistant to interference.