DIGITAL TRANSMISSION

Multiple Choice Type Questions

1. PCM generation requires a LPF at the beginning [WBUT 2013, 2014, 2017]
   a) to eliminate aliasing effect
   b) to eliminate quantization noise
   c) to eliminate decoding noise
   d) to eliminate white noise

Answer: (b)

2. Which of the following modulations is analog? [WBUT 2013, 2014, 2017]
   a) PCM
   b) DPCM
   c) PAM
   d) delta modulation

Answer: (c)

3. To avoid aliasing, what is the Nyquist rate of this signal $x(t) = 8\cos 200\pi t$? [WBUT 2013, 2014, 2017]
   a) 50 Hz
   b) 100 Hz
   c) 200 Hz
   d) 400 Hz

Answer: (c)

4. In a binary PCM $q = 16$, the number of bits per codeword will be equal to [WBUT 2013, 2014, 2017]
   a) 2
   b) 4
   c) 8
   d) 3

Answer: (b)

5. The use of non-uniform quantization leads to [WBUT 2015]
   a) reduction of transmission bandwidth
   b) increase in maximum SNR
   c) increase in SNR for low band signal
   d) simplification process

Answer: (c)

6. The Nyquist sampling rate for a band limited signal of 4 kHz is [WBUT 2015]
   a) 4 kHz
   b) 8 kHz
   c) 2 kHz
   d) 32 kHz

Answer: (b)

7. The bandwidth required for transmitting 4 kHz signal using PCM with 128 quantization level is [WBUT 2015]
   a) 8 kHz
   b) 16 kHz
   c) 28 kHz
   d) 32 kHz

Answer: (c)

8. AMI is another name of which process? [WBUT 2015]
   a) Polar
   b) Bipolar
   c) On –off
   d) None of these

Answer: (b)

9. The frequency spectrum of a square wave or a rectangular wave in time domain is [WBUT 2015]
   a) impulse function
   b) sinc function
   c) sin function
   d) Gaussian function

Answer: (b)

10. In a PCM system the number of quantization level is 16 and the maximum signal frequency is 4 kHz, the bit transmission rate is [WBUT 2015]

a) 32 bits/s
b) 16 bits/s
c) 32 kbits/s
d) 64 kbits/s

Answer: (c)

11. How many bits would be required to represent a 256 level quantization in PCM? [WBUT 2016]

a) 6
b) 8
c) 5
d) 7

Answer: (b)

12. PCM generation requires an LPF at the beginning because [WBUT 2016]

a) to eliminate aliasing effect
b) to eliminate quantization noise
c) to eliminate decoding noise
d) none of these

Answer: (a)

13. The Nyquist rate for a signal $x(t) = 5\cos(2\pi * 500t)$ is [WBUT 2018]

a) 1200 Hz
b) 1000 Hz
c) 2000 Hz
d) 5000 Hz

Answer: (b)

14. The signaling rate of a PCM system having f_s = 8 kHz and N = 8 is [WBUT 2018]

a) 125 kHz
b) 16 kHz
c) 32 kHz
d) 64 kHz

Answer: (d)

15. In law μ-law compression, $\mu = 0$ corresponds to [MODEL QUESTION]

a) non-uniform quantization
b) no quantization
c) better S/N ratio
d) uniform quantization

Answer: (d)

16. Eye pattern is used to study [MODEL QUESTION]

a) ISI
b) Quantization noise
c) Error rate
d) None of these

Answer: (a)

17. Eye pattern is a technique for the measurement of [MODEL QUESTION]

a) Modulation index
b) ISI
c) Equalization level
d) Time constant

Answer: (b)

18. Quantization noise in PCM, for a given input, is related to [MODEL QUESTION]

a) Sampling rate
b) Maximum frequency of the modulating signal
c) Number of digits used for coding
d) Lowest frequency of the modulating signal

Answer: (c)

19. Compading is used in PCM to [MODEL QUESTION]
   a) Improve signal to noise ratio for low level input signals
   b) Reduce the probability of errors
   c) Reduce quantization noise
   d) To increase signal strength

Answer: (a)

20. In a PCM system, non-uniform quantization leads to [MODEL QUESTION]
   a) Reduced bandwidth
   b) Simplification of the low level signals
   c) Increased SNR for low level signals
   d) Increased SNR for high level signals

Answer: (c)

21. The line speed of an 8-bit PCM system with sampling rate of 8 KHz is [MODEL QUESTION]
   a) 32 Kbps b) 64 Kbps c) 1 Kbps d) None of these

Answer: (b)

22. In PCM if the analog signal has the maximum frequency of 4 KHz, the minimum sampling rate is [MODEL QUESTION]
   a) 4 KHz b) 32 KHz c) 8 KHz d) None of these

Answer: (c)

23. A quantizer is a [MODEL QUESTION]
   a) Multiplexer b) Demultiplexer
   c) A to D converter d) D to A converter

Answer: (c)

24. A signal of $f_{max} = 10KHz$ is sampled at the Nyquist rate. The time interval between the simples is [MODEL QUESTION]
   a) 50 μs b) 10 μs c) 5 μs d) 100 μs

Answer: (a)

25. The line code that has zero d.c. component for pulse transmission of random binary data is [MODEL QUESTION]
   a) UPNRZ b) UPRZ c) BPRZ – AMI d) BPNRZ

Answer: (c)

26. The Nyquist sampling rate for the signal $s(t) = 10\cos(50\pi t)\cos^2(150\pi t)$ when t is in seconds is [MODEL QUESTION]
   a) 150 samples/second b) 200 samples/second
   c) 300 samples/second d) 350 samples/second

Answer: (d)

Short Answer Type Questions

1. Write down the expression, sketch the spectrum and waveform of the received pulse in the case of an ideal Nyquist channel. What are the drawbacks of ideal Nyquist channel? **[WBUT 2013, 2017]**

Answer:

1st Part:

We know that an ideal Nyquist channel is an ideal LPF with a bandwidth of W H and that

$$W = \frac{1}{2T}$$

$$H_C(f) = \begin{cases} 1; & |f| \leq \dfrac{1}{2T} \\ 0; & \text{otherwise} \end{cases}$$

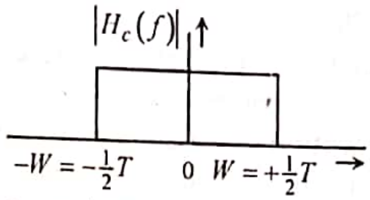


Fig: Transfer function of an ideal Nyquist channel

2nd Part:

Drawbacks of ideal Nyquist channel

(a) Not physically realizable

(b) Even slight sampling jitter causes ISI

(c) Rate of decrease of sinc pulse is only $I/|t|$

2. Derive the relation between signaling rate and transmission bandwidth in PCM system. **[WBUT 2014, 2017]**

Answer:

Let us assume that in a PCM system with linear quantizer there are L number of representation levels and n is the number of bits used for encoding. Then

$$L = 2^n$$

Let f_s = number of samples per second

So, Number of bits per second = signaling rate

= Number of bits per sample × Number of samples per second

$= n \times f_s$

Hence signaling rate $= R = nf_s$

Bandwidth needed for PCM transmission is given by half of the signaling rate.

Thus transmission bandwidth of PCM is given by

$$BW \geq \frac{1}{2}R \geq \frac{1}{2}nf_s$$

Since. $f_s \geq 2f_m$

where f_m is the maximum modulating frequency, we get

$$\boldsymbol{BW \geq nf_m}$$

3. a) Briefly explain the working principle uniform quantization. [WBUT 2015]

Answer:

Uniform quantization:

For uniform quantization the step size is fixed throughout quantization process. Therefore, the maximum quantization error is fixed irrespective of signal amplitude and it is $\pm\frac{s}{2}$. Thus for large amplitude input signal, the S/N ratio will be large i.e. the effect of Q noise will be less but for small amplitude input signal S/N ratio will be less i.e. the effect of Q noise will be more.

Uniform scalar quantization is very simple and straightforward for implementation. It is designed based on assumption that the input source is uniformly distributed. But often probability of distribution of the source symbols is not uniform in nature and the uniform scalar quantization results in poor reconstructed quality. As a result, there is a necessity to design non-uniform quantizers for these types of sources.

b) What is Nyquist rate of sampling? [WBUT 2015]

Answer:

This maximum sampling interval is known as Nyquist sampling interval. Similarly, minimum sampling rate $f_0 = (1/2f_m)$ is called as Nyquist sampling rate.

4. Draw an Eye diagram and mention the significance of its different parts. [WBUT 2015]

Answer:

Refer to Question No. 5(b) of Long Answer Type Questions.

5. a) Find the Nyquist rate and Nyquist interval for the signal: [WBUT 2016]

$$X(t) = 1/2\pi\cos(4000\pi t)\cos(1000\pi t)$$

b) Explain the operation regenerative repeater with suitable diagram.

Answer:

a) We have signal

$$X(t) = \frac{1}{2\pi}\cos(4000\pi t)\cos(1000\pi t)$$

$$= \frac{1}{2\pi}\times\frac{1}{2}\left[2\cos(4000\pi t)\cos(1000\pi t)\right]$$

$$= \frac{1}{4\pi}\left[\cos(4000\pi t + 1000\pi t) + \cos(4000\pi t - 1000\pi t)\right]$$

$$= \frac{1}{4\pi}\left[\cos(5000\pi t) + \cos(1000\pi t)\right]$$

where $\omega_1 t = 5000\pi t$ and $\omega_2 t = 3000\pi t$

$2\pi f_1 = 5000\pi$ $\qquad$ $2\pi f_2 = 3000\pi$

$$f_1 = \frac{5000\pi}{2\pi} = 2500\text{ Hz} \qquad f_2 = \frac{3000\pi}{2\pi} = 1500\text{ Hz}$$

According to Nyquist theorem

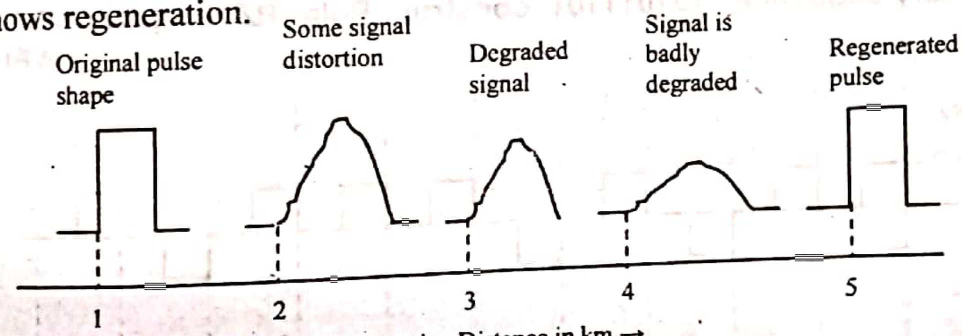
$$f_s = 2 \times \text{Maximum frequency component} = 2 \times 2500 = 5000\text{ Hz}$$

Nyquist interval

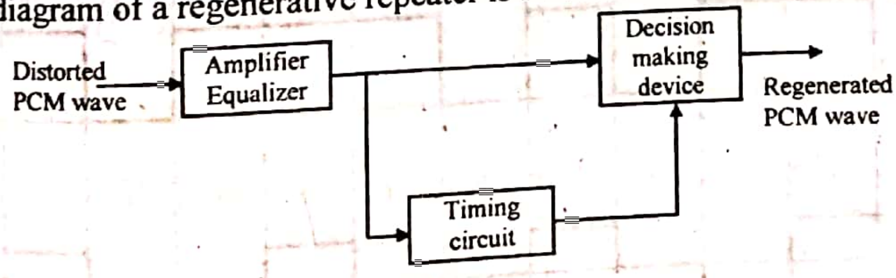
$$T_s = \frac{1}{f_s} = \frac{1}{5000}$$

$$T_s = 0.0002\text{ sec} = 0.2\text{ msec.}$$

b) Regenerative repeaters are important in Pulse Code Modulation (PCM) systems. PCM systems have the ability to control the effects of distortion and noise produced by transmitting a PCM signal through a channel. This capability of PCM is accomplished by regeneration of the PCM wave. The shape of the pulse degrades as a function of line length of the channel. During the time that the transmitted pulse still is reliably identified, the pulse is amplified by a digital amplifier that recovers its original shape. The pulse is thus 'reborn' or 'regenerated'. This regeneration of the PCM pulses is done at regular intervals along a transmission line by circuits known as regenerative repeaters. The figure below shows regeneration.



The block diagram of a regenerative repeater is shown below:



6. Establish the relation between phase sensitivity and frequency sensitivity. **[WBUT 2016]**

Answer:

In phase modulation, the angular argument $\theta(t) = 2\pi f_c t + k_p m(t)$ i.e., $\theta(t)$ is varied linearly with the message signal $m(t)$ and $2\pi f_c t$ represents the angle of the unmodulated carrier 'k_p' represents the phase sensitivity of the modulator in radians/volt.

The **phase modulated wave** is thus,

$$S(t) = A_c \cos\left[2\pi f_c t + k_p m(t)\right]$$

In **frequency modulation**, the instantaneous frequency $f_i(t)$ is varied linearly with the message signal $m(t)$, as shown by

$$f_i(t) = f_c + k_f m(t)$$

where f_c represents the frequency of the unmodulated carrier, 'k_f' represents the frequency sensitivity of the modulator in Hz/volt. The **frequency modulated** wave

$$S(t) = A_c \cos\left[2\pi f_c t + 2\pi k_f \int_0^t m(t)\,dt\right]$$

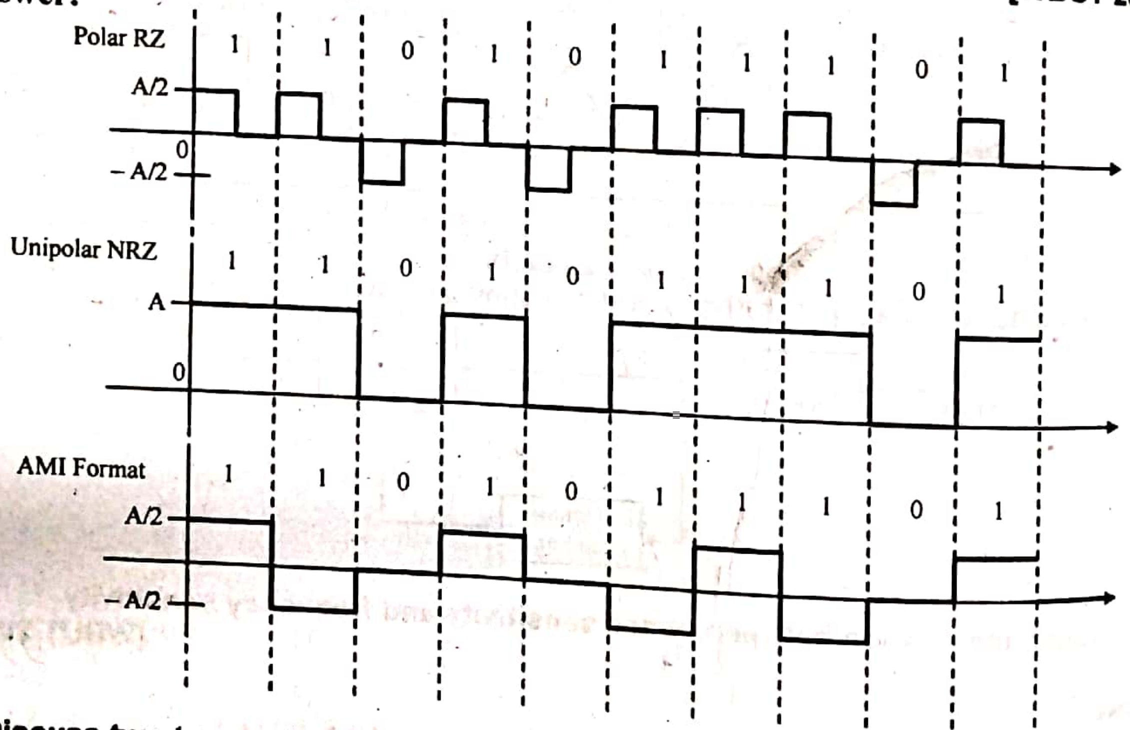
A close relationship exists between the PM and FM signals:

$$k_p = 2\pi k_f$$

or, $$\boxed{k_f = \frac{k_p}{2\pi}}$$

7. For the binary sequence 1101011101, construct Polar RZ, Unipolar NRZ and AMI format.

Answer: **[WBUT 2017]**



8. Discuss two types of quantizers.

Answer: **[MODEL QUESTION]**

Types of Quantizers

There are basically two types of quantizer, namely, (i) uniform quantizer and (ii) non-uniform quantizer.

A uniform quantizer is that type of quantizer in which the step size is uniform throughout the input signal range. Quantization process involved in such quantizer is called uniform quantization. A uniform quantizer is also called linear quantizer.
The type of quantizer in which the step size varies according to the input signal value is called non-uniform quantizer and the process involved is called non-uniform quantization. A non-uniform quantizer is also called non-linear quantizer.

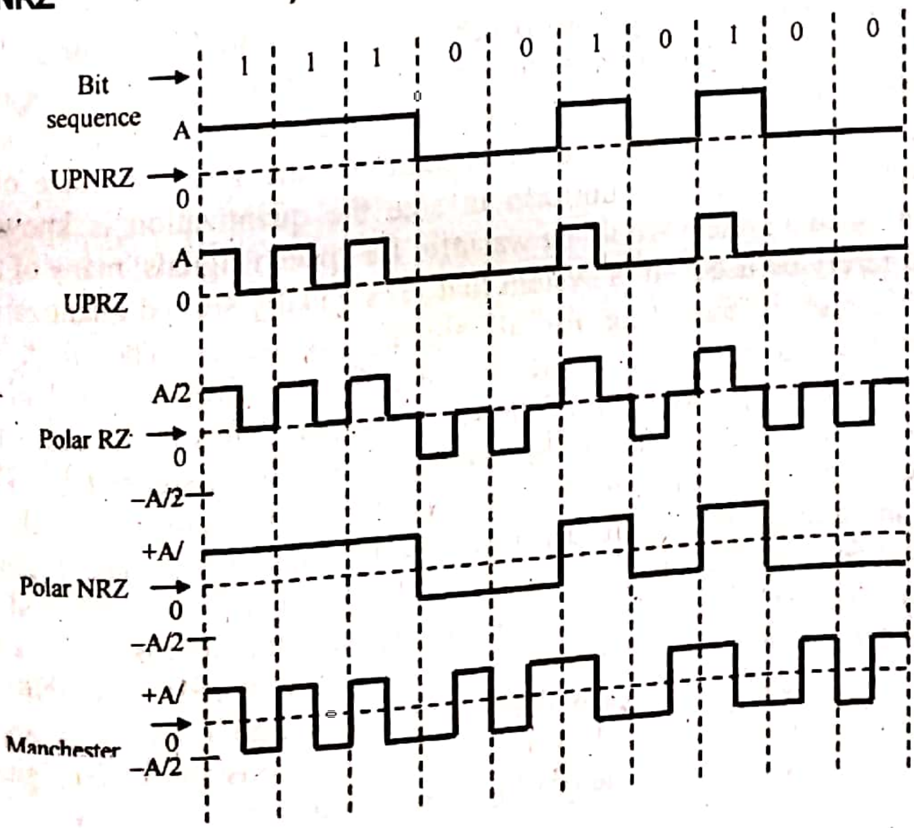
9. Define line coding. Write the properties of line coding. [MODEL QUESTION]

Answer:
Line coding is a technique in digital communication system by which standard logic levels are converted to a form which is more suitable for line transmission.
The following are the desirable properties of a line code:

1. Self synchronization i.e. timing or clock signal can be easily extracted from the code.
2. Low probability of bit error.
3. It should have a spectrum that is suitable for the channel.
4. The transmission bandwidth should be as small as possible.
5. It should have error detection capability.
6. The code should be transparent.

10. Given the data stream 1110010100, sketch the transmitted sequence of rectangular pulses for each of the following line codes:
a) Unipolar NRZ b) Unipolar RZ c) Polar RZ
d) Polar NRZ e) Manchester. [MODEL QUESTION]

Answer:



11. State the advantages and disadvantages of PCM. **[MODEL QUESTION]**

Answer:

Advantages

1. PCM offers robustness to channel noise and interference.
2. Efficient regeneration of the coded signal is possible along the transmission path with PCM.
3. PCM gives efficient exchange of increased channel bandwidth for improved SNR.
4. PCM offers a uniform format for the transmission of different kinds of baseband signals.
5. In a TDM system, message sources may be dropped or reinserted comparatively easily if PCM is used.
6. PCM provides secure communication through the use of special modulation schemes or encryption.

Disadvantages

1. PCM has increased channel bandwidth.
2. PCM has increased system complexity.

Long Answer Type Questions

1. What is the difference between Uniform and Non-uniform quantization? What is A-law and μ-law?

Answer: **[WBUT 2013]**

Communication channels, very low speech volumes predominate; 50% of the time, the voltage characterizing detected speed energy is less than one-fourth of the rms value. Large amplitude values are relatively rare; only 15% of the time does the voltage exceed the rms value. The quantization noise depends on the step size (size of the quantile interval). When the steps are uniform in size the quantization is known as uniform quantization. Such a system would be wasteful for speech signals; many of the quantizing steps would rarely be used. In a system that uses equally spaced quantization levels, the quantization noise is the same for all signal magnitudes. Therefore, with uniform quantization, the signal-to-noise (SNR) is worse for low-level signals than for high-level signals. Non-uniform quantization of the strong signals. Thus in the case of non-uniform quantization, quantization noise can be made proportional to signal size. The effect is to improve the overall SNR by reducing the noise for the predominant weak signals, at the expense of an increase in noise for the rarely occurring strong signals. Figure 1 compares the quantization of a strong versus a weak signal for uniform and non-uniform quantization. The staircase-like waveforms represent the approximations to the analog waveforms (after quantization distortion has been introduced). The SNR improvement that non-uniform quantization provides for the weak signal should be apparent. Non-uniform quantization can be used to make the SNR a constant for all signals within the input range. For voice signals, the typical input signal.

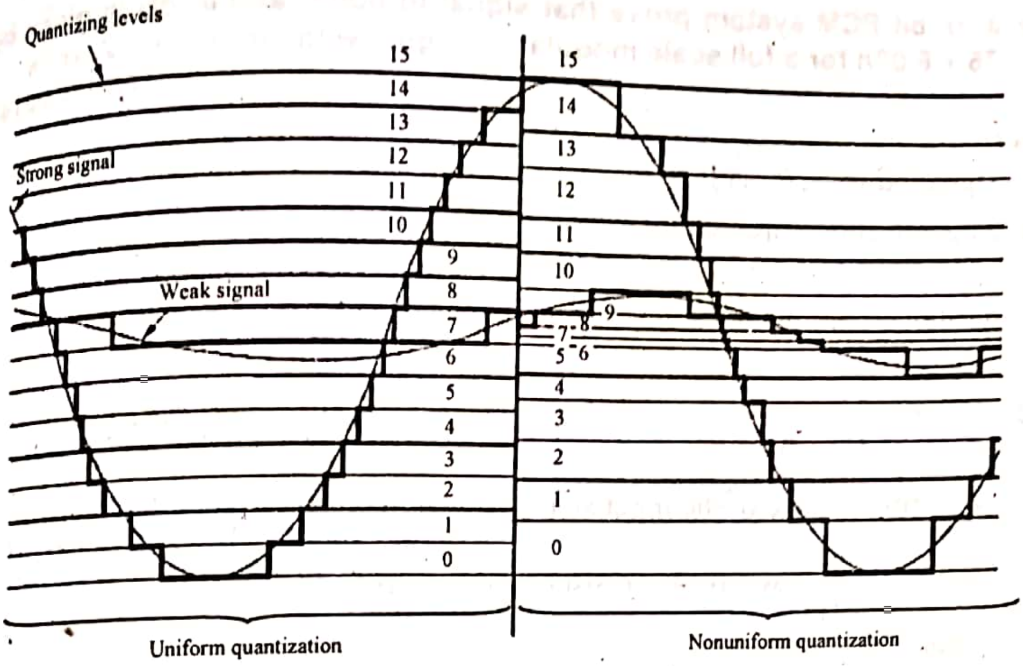


Fig: Uniform and nonuniform quantization of signals

2nd Part:

The two related laws are μ-law and A-law.

μ-law is given by

$$|v_2| = \frac{\log(1+\mu(v,1)}{\log(1+\mu)}$$

A-law is represented by

$$|v_2| = \begin{cases} \dfrac{A|v_1|}{1+\log A} & \text{for } 0 \le |v_1| \le \dfrac{1}{A} \\ \dfrac{1+\log(A|v_1|)}{1+\log A} & \text{for } \dfrac{1}{A} \le |v_1| < 1 \end{cases}$$

2. a) What is the difference between uniform and non uniform quantizer? What is the necessity of non-uniform quantizer? **[WBUT 2014, 2018]**

Answer:

1st Part: ***Refer to Question No. 1 of Long Answer Type Questions.***

2nd Part:

Non-uniform Quantization: In order to minimize the average distortion in the reconstructed image because of the quantization, we can lightly quantize the transform coefficients or prediction error values in the region of the high importance and heavily quantize the corresponding coefficients in a less important region in the image. One way to achieve this is to use non-uniform quantization such that the quantization steps are smaller for the samples those have more concentration in the curve of probability distribution of the samples.

b) For a 'n' bit PCM system prove that signal to nose ratio in dB is given by (S/N) dB = 1.76 + 6.02n for a full scale modulating signal with amplitude 'V' volts.
[WBUT 2014, 2018]

Answer:

Let the signal power $= P_s$ and

Average quantization noise output power $= P_q$

If Δ is the step size of the linear quantization, then $P_q = \dfrac{\Delta^2}{12}$

Thus the $SQR = \dfrac{P_s}{P_q} = \dfrac{P_s}{\Delta^2/12}$ where SQR = signal to quantization noise ratio

Let $\quad V_r$ = rms voltage of the input analog signal, then

$$SQR = \frac{V_r^2/R}{\Delta^2/12} \text{ where } R = \text{resistance of the load}$$

If $R = 1$, then

$$SQR = \frac{V_r^2}{\Delta^2/12} = \frac{12V_r^2}{\Delta^2} = 12\left(\frac{V_r}{\Delta}\right)^2$$

$$= 10\log 12 + 20\log\left(\frac{V_r}{\Delta}\right)\text{dB} = 10.8 + 20\log\frac{V_r}{\Delta}\text{ dB}$$

If the input signal is sinusoidal, then $V_r = \dfrac{V_m}{\sqrt{2}}$ where V_m is the maximum value of the sinusoidal signal.

Thus in the case of sinusoidal input

$$SQR = \frac{\left(V_m/\sqrt{2}\right)^2}{\Delta^2/12} = 10\log\left[\frac{\left(V_m/\sqrt{2}\right)^2}{\Delta^2/12}\right]\text{dB} = 10\log\left[6\left(\frac{V_m}{\Delta}\right)^2\right]\text{dB}$$

$$= 10\log 6 + 20\log\frac{V_m}{\Delta}\text{ dB} = 7.78 + 20\log\frac{V_m}{\Delta}\text{ dB}$$

If $\quad L$ = Number of representation levels or steps, then

$\Delta = \dfrac{2V_m}{L}$ and

$$SQR = 10\log\left[\frac{\left(V_m/\sqrt{2}\right)^2}{\left(\dfrac{4V_m^2}{12L^2}\right)}\right]\text{dB}$$

$$= 10\log 1.5L^2\text{ dB} = 10\log 1.5 + 20\log L\text{ dB} = 1.76 + 20\log L\text{ dB}$$

Since $L = 2^n$ where n = number of bits used

$SQR = 10\log 1.5 + 20\log 2^n = 1.76 + 20n\log 2\ \text{dB} = 1.76 + 6.02n\ \text{dB}$

Thus for every additional bit it gives an improvement of 6 dB in SQR. This is known as 6 dB rule of PCM. SQR can also be written as SNR.

3. a) What are the differences between uniform and non-uniform quantizers? Explain non-uniform quantization with suitable diagram.

b) Prove that the quantization error for the PCM is $(\Delta)^2/12$, where Δ is the step size.

c) What is the aliasing effect and how can it be overcome? [WBUT 2016]

Answer:

a) 1st Part:

A uniform quantizer is that type of quantizer in which the step size is uniform throughout the input signal range. Quantization process involved in such quantizer is called uniform quantization. A uniform quantizer is also called linear quantizer.

The type of quantizer in which the step size varies according to the input signal value is called non-uniform quantizer and the process involved is called non-uniform quantization. A non-uniform quantizer is also called non-linear quantizer.

2nd Part:

Quantization is an essential step in the generation of PCM signal. Quantization is the process of converting a discrete-time continuous-amplitude signal such as a PAM signal into a discrete-amplitude discrete-time signal.

An analog signal such as voice signal or video signal has a continuous range of amplitudes. Such a signal when sampled will have a continuous amplitude range. Thus within the finite amplitude range of the signal there will be an infinite number of amplitude levels. In practice it is not necessary to transmit the exact amplitudes of the samples since human ear or eye cannot detect small finite differences in amplitude of the signals. Thus the original analog signal may be approximated by a signal constructed of discrete amplitudes selected on a minimum error basis from an available set. It is clearly seen that if we select the discrete amplitude levels with sufficiently close spacing, we may make the approximated signal practically indistinguishable from the original analog signal. This approximation of the sampled analog signal is called quantization.

Let us consider the diagram below:

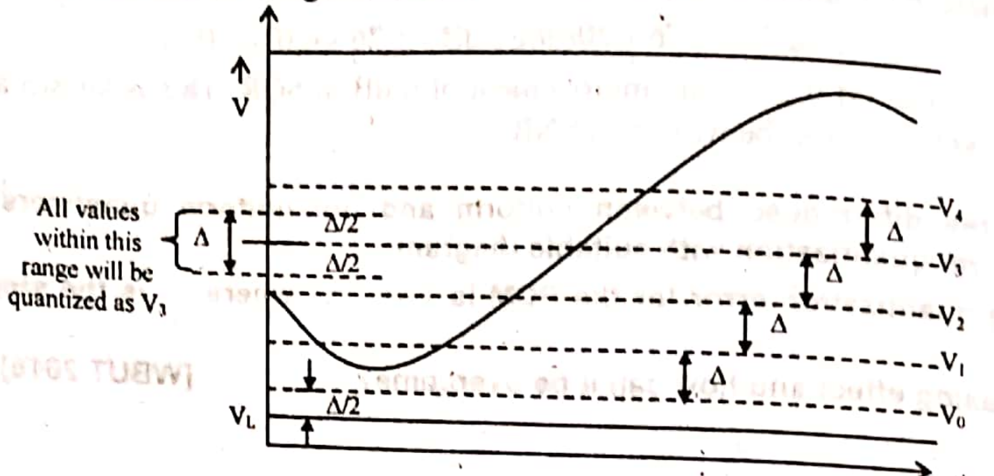


Fig: Quantization

Let V_H and V_L are the maximum and minimum values of the signal V. Let the range divided into M (say M = 8) equal steps of size Δ such that $\Delta = (V_H - V_L)/M$. Quantiz values of the signal are given by

$$V_q = V_0 \text{ if } (V_0 - \frac{\Delta}{2}) \le V < (V_0 + \frac{\Delta}{2})$$

$$V_q = V_1 \text{ if } (V_1 - \frac{\Delta}{2}) \le V < (V_1 + \frac{\Delta}{2})$$

$$V_q = V_2 \text{ if } (V_2 - \frac{\Delta}{2}) \le V < (V_2 + \frac{\Delta}{2})$$

$$V_q = V_3 \text{ if } (V_3 - \frac{\Delta}{2}) \le V < (V_3 + \frac{\Delta}{2})$$

and so on.

b) For linear quantization, the probability distribution of error can be assumed to b constant within the range ± Δ/2. The variation of probability p(e) with error is shown i the figure below.

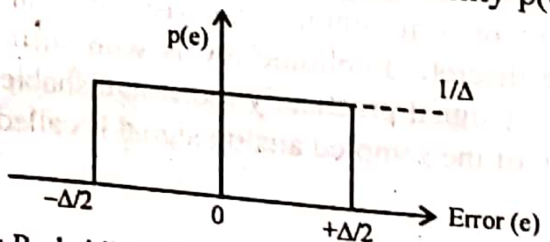


Fig: Probability distribution of quantization error due to linear quantization

The average quantization noise output power is given by the variance

$$\sigma^2 = \int_{-\infty}^{\infty} (e-\mu)^2 \, p(e) \, de$$

where μ is the mean value of the error, *e*.

In this case, the limits of integration are $+\Delta/2$ and $-\Delta/2$ and $\mu=0$. Also $p(e)=\frac{1}{\Delta}$ and thus

$$\sigma^2=\int_{-\Delta/2}^{+\Delta/2} e^2 p(e)de=\int_{-\Delta/2}^{+\Delta/2} e^2\left(\frac{1}{\Delta}\right)de=\frac{1}{\Delta}\int_{-\Delta/2}^{\Delta/2} e^2 de=\frac{1}{\Delta}\left[\frac{e^3}{3}\right]_{-\Delta/2}^{+\Delta/2}=\frac{\Delta^2}{12}$$

Thus the average power of the quantization noise is proportional to the square of the step size, Δ. The signal distortion due to quantizing noise can be minimized by reducing the step size to a very small value.

c) When an analog signal is sampled at a rate below the Nyquist rate, the sidebands overlap producing an interference effect. This interference effect is called aliasing effect. If aliasing takes place, it is not possible to recover the original analog signal.

4. a) Draw and explain the block diagram of baseband binary data transmission system. **[WBUT 2017]**

Answer:

Refer to Question No. 5(c) of Long Answer Type Questions.

b) What is Nyquist rate and Nyquist interval? **[WBUT 2017]**

Answer:

Nyquist Rate and Nyquist Interval:

The sampling frequency which is greater than twice the maximum frequency content of the signal to be sampled is called the Nyquist Rate. The reciprocal of the Nyquist Rate is called Nyquist Interval.

5. Write short notes on the following: **[WBUT 2013]**

a) Matched filter **[WBUT 2013, 2017]**

b) Eye pattern **[WBUT 2014, 2018]**

c) ISI in digital communication **[WBUT 2014, 2018]**

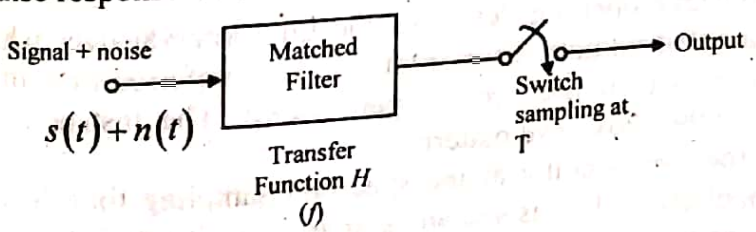
d) Sampling Theorem and Aliasing Effect **[WBUT 2014, 2017, 2018]**

e) Companding

Answer:

a) Matched filter:

Matched filter is a device for the optimal detection of a digital pulse. It is named so because the impulse response of the matched filter matches the pulse shape.



Properties of matched filter

- The maximum output signal-to-noise ratio only depends on the energy of the input, and is nothing to do with the pulse shape itself.

- Namely, whether the pulse shape is sinusoidal, rectangular, triangular, etc is irrelevant to the maximum output signal-to-noise ratio, as long as these pulse shapes have the same energy.

b) Eye Pattern:

Inter symbol interference in a PCM or data transmission system can be studied experimentally with the help of a display in the oscilloscope. Here the received distorted wave is applied to the vertical deflection plates of an oscilloscope and a saw tooth wave at a transmitted symbol rate of $R = \frac{1}{T}$ is applied to the horizontal plates. The resulting display on the oscilloscope is called an eye pattern or eye diagram. It is so called because of its resemblance to the human eye. The interior region of the eye pattern is called the eye opening.

A typical eye pattern is shown below:

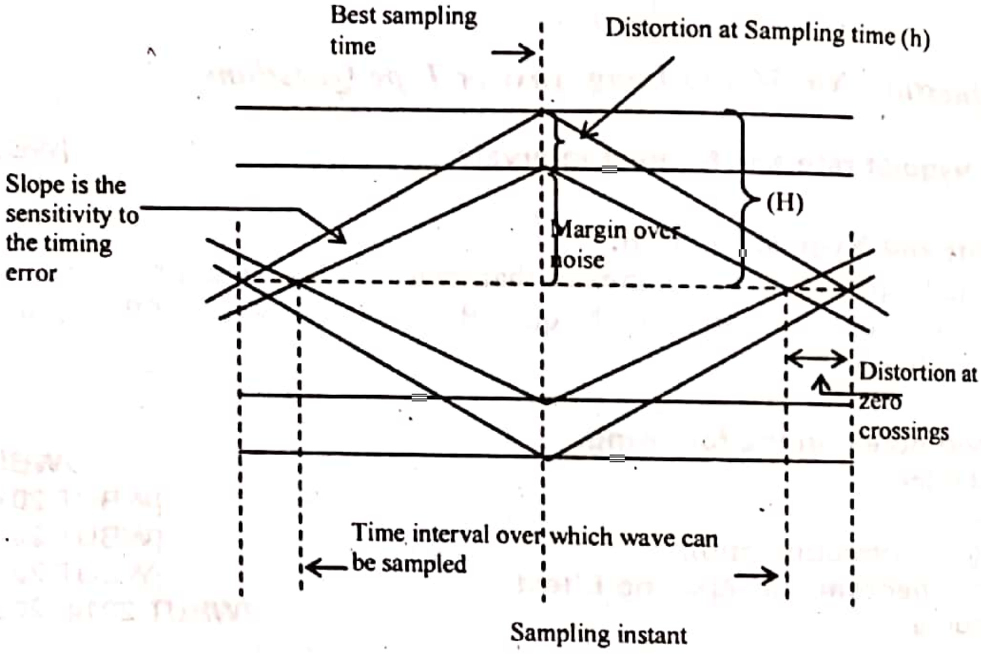


Fig: An Eye Pattern

An eye pattern provides the following information about the performance of a digital communication system.

1. The width of the eye opening describes the time interval over which the received wave can be sampled without error from ISI. The preferred time of sampling is the instant of time at which the eye is open widest. The instant is shown as 'best sampling time' in the above eye pattern.
2. The height of the eye opening at the specified sampling time is a measure of the margin of channel noise. This is shown as 'margin over noise' in the above diagram.
3. The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied.

4. Any non-linear transmission distortion will reveal itself in an asymmetric or squinted eye.

A typical set-up for Eye Diagram measurement is shown below:

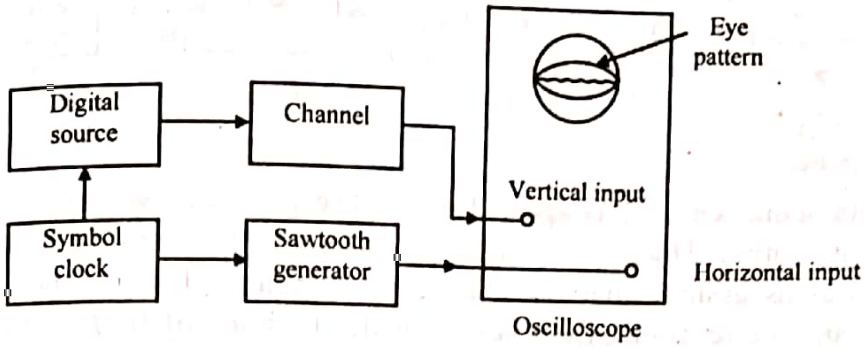


Fig: Eye diagram measurement

ISI degradation can be measured from the eye diagram with the formula

$$ISI = 20\log\frac{h}{H}$$

where H = ideal vertical opening in cm.
h = degraded vertical opening in cm.

An eye pattern provides the following information about the performance of a digital communication system.

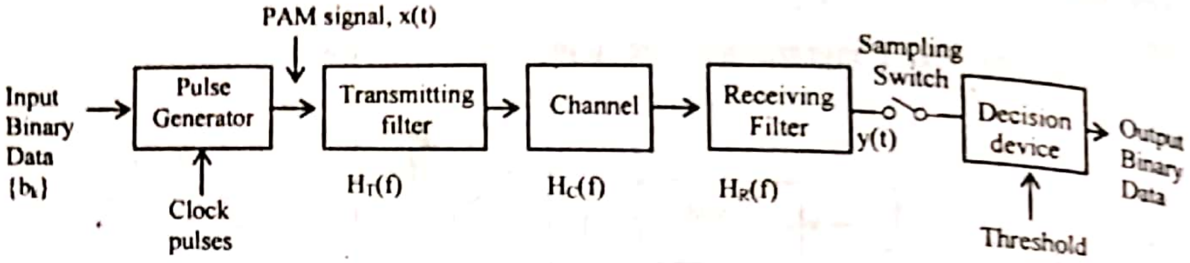
1. The width of the eye opening describes the time interval over which the received wave can be sampled without error from ISI. The preferred time of sampling is the instant of time at which the eye is open widest. The instant is shown as 'best sampling time' in the above eye pattern.
2. The height of the eye opening at the specified sampling time is a measure of the margin of channel noise. This is shown as 'margin over noise' in the above diagram.
3. The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied.
4. Any non-linear transmission distortion will reveal itself in an asymmetric or squinted eye.

c) ISI in digital communication

When digital data are transmitted over a band-limited channel, dispersion in the channel causes an overlap in time between successive symbols. This effect is known as Inter Symbol Interference or ISI.

A baseband communication channel can be considered as a low pass filter. It has limited bandwidth and non-linear frequency response. So when digital pulses are transmitted through this channel, the shape of the pulses gets distorted. Because of this distortion, one distorted pulse will affect other pulses and the cumulative effect of this distortion will make the decision process in favour of 'one' or 'zero' erroneous.

Let us consider a baseband binary data transmission system as shown below:



The binary data sequence $\{b_k\}$ is applied to a pulse generator which produces discrete PAM signal at its output. The transmitting filter has a transfer function $H_T(f)$. The channel is dispersive. Let us assume that the channel is noiseless. The channel has a transfer function $H_C(f)$ and the receiving filter has a transfer function of $H_R(f)$. The output of the receiving filter is sampled and applied to the decision device to get the binary output data.

Obviously PAM signal is given by $x(t) = \sum_{k=-\infty}^{\infty} a_k\, g(t - kT_b)$

where $g(t)$ denotes the basic pulse such that $g(t) = 1$ for $t = 0$ and $g(t) = 0$ for $t = \pm T_b, \pm 2T_b, \ldots$..and a_k is the amplitude of b_k and T_b is the bit duration.

The receiving filter output is given by

$$y(t) = \mu \sum_{k=-\infty}^{\infty} a_k\; p(t - kT_b - t_d)$$

where μ is a scaling factor and t_d is the time delay introduced by the channel. $p(t)$ is normalized such that $p(t) = 1$ at $t = 0$. We may assume $t_d = 0$. The pulse $\mu\; a_k\, p(t)$ is the response of the transmitting filter, the channel and the receiving filter. We may relate $g(t)$ and $p(t)$ in the frequency domain by writing

$$\mu P(f) = G(f) H_T(f) H_C(f) H_R(f)$$

where $P(f)$ and $G(f)$ are the Fourier transforms of $p(t)$ and $g(t)$ respectively.
The receiving filter output is sampled at time $t_i = i\; T_b$ where i is an integer. Thus the sampled output at $t = t_i$ is given by

$$y(t_i) = \mu \sum_{k=-\infty}^{\infty} a_k\; p(i\,T_b - KT_b) = \mu\; a_i\; k + \mu \sum_{\substack{k=-\infty \\ k \neq i}}^{\infty} a_k\; p(i\,T_b - kT_b)$$

where $i = 0, \pm 1, \pm 2, \pm 3, \ldots\ldots\ldots$

It is clearly seen that the sampled values of the receiving filter output at $t = t_i$ is not only determined by the i th transmitted bit but also depends on the contribution of all other transmitted bits. The first term μa_i is produced by the ith-transmitted bit. The second term represents the residual effect of all other transmitted bits on the decoding of the ith bit. This residual effect is called Inter Symbol Interference or ISI.

Thus when a sequence of pulses are transmitted through the above band-limited system the frequency components of the input pulse will be differentially attenuated and delayed by the system such that the pulse appearing at the output of the system is dispersed over an interval longer than T_b seconds. These dispersed responses originating from different symbol intervals will interfere with each other and result in ISI.

d) Sampling Theorem

The mathematical basis of sampling process has been laid by Nyquist sampling theorem. It also gives an idea about the recovery or reconstruction of the original analog signal completely from its samples. The statement of the sampling theorem is thus given in two parts.

1. A band limited signal of finite energy which has no frequencies beyond W Hz is completely described by specifying the values of the signals at the instants of time separated by 1/(2W) seconds.
2. A band-limited signal of finite energy which has no frequency components beyond W Hz may be completely recovered from a knowledge of its samples taken at a rate of 2W samples per second.

The sampling rate 2W per second is called the Nyquist rate. The reciprocal $\frac{1}{2W}$ is called Nyquist interval.

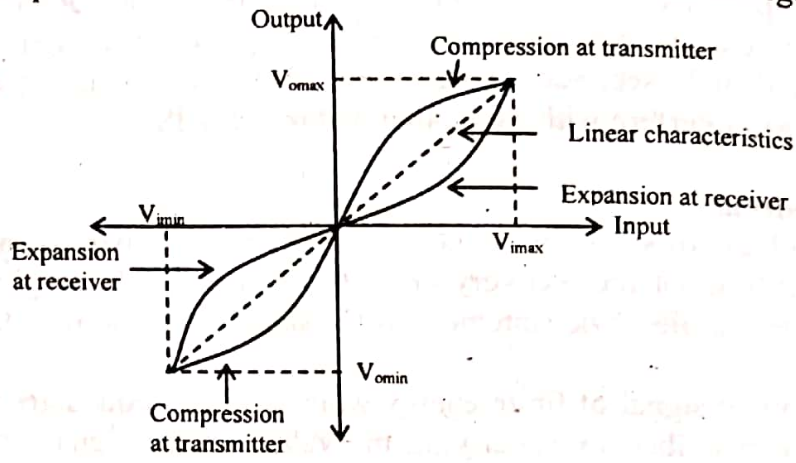
The sampling theorem serves as the basis for conversion of continuous-time signals to discrete-time signals and vice versa. The theorem has tremendous application in digital communication and digital signal processing (DSP).

When an analog signal is sampled at a rate below the Nyquist rate, the sidebands overlap producing an interference effect. This interference effect is called aliasing effect. If aliasing takes place, it is not possible to recover the original analog signal.

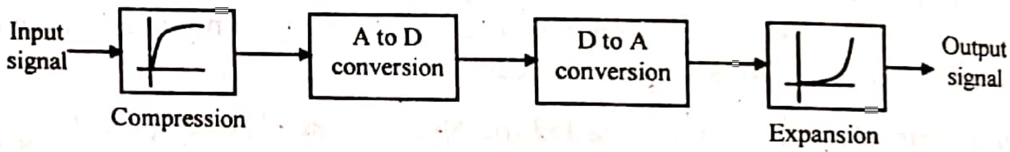
e) Companding

Quantization is of two types viz. uniform quantization and non-uniform quantization. Non-uniform quantization is achieved through companding. This is a process in which compression of the input signal is done in the transmitter whereas expansion of the signal is done at the receiver. The combination of compressing and expanding is companding. In linear or uniform quantization the small amplitude signals would have a poorer SNR than the large amplitude signals. This is a disadvantage of linear quantization. To remove this problem, non-linear quantization is used in which the step size varies with the amplitude of the input signal. The step size variation is achieved by distorting the input signal before the quantization process. This process of distorting the input signal before quantization is known as compression in which the signal is amplified at low signal levels and attenuated at high signal levels. After compression, uniform quantization is applied. A reverse operation known as expansion is done at the receiver. Here the signal is attenuated at low signal levels and amplified at high signal levels. The overall effect of companding, is to make the overall transmission distortionless.

A typical input-output characteristics of a compander is shown in the figure below:



The process of companding is shown in block diagram below.



6. a) What is Nyquist criterion for zero ISI? **[MODEL QUESTION]**

Answer:

Nyquist Criterion for Zero ISI

ISI causes pulse spreading or overlapping. However pulse amplitudes can be detected correctly despite ISI provided there is no ISI at the decision making instants $t = i\,T_b$. Thus the effect of ISI can be eliminated by proper shaping of the band-limited pulse. Nyquist achieved zero ISI by choosing a pulse shape that has a non-zero amplitude at its center, say at $t = 0$, and zero amplitudes at all other sampling instants i.e. $t = \pm i\,T_b$ where $i = 1,2,3\ldots$and T_b is the separation between successive transmitted pulses. Thus we can write the condition for zero ISI as

$$p(t) = \begin{cases} 1 & \text{when } t = 0 \\ 0 & \text{when } t = \pm i\,T_b \text{ for } i = 1,2,3 \text{ and } T_b = \dfrac{1}{R_b} \end{cases}$$

A pulse having a shape satisfying the above criterion causes zero ISI at all the remaining pulse centers. This is depicted graphically below:

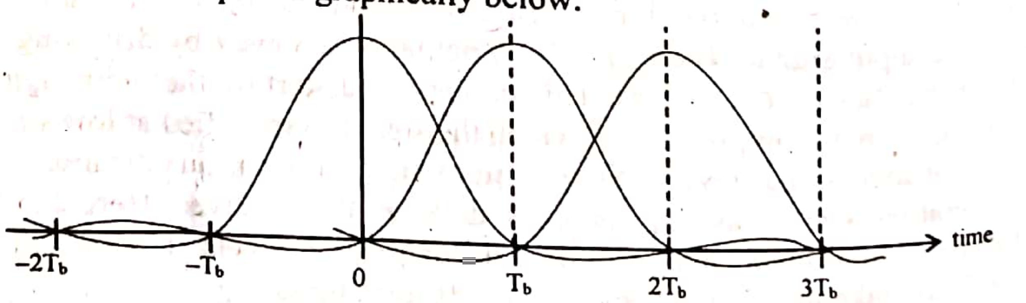


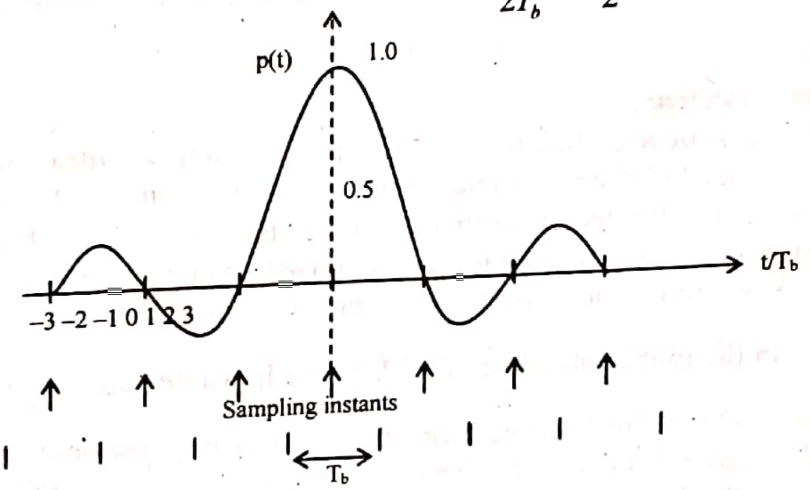
Fig: Nyquist Criterion for zero ISI

The above diagram shows several successive pulses centered at $t = 0, T_b, 2T_b, 3T_b, \ldots,$. From the figure it is clear that the samples at $t = 0$, T_b, $2T_b$, $3T_b$....consist of the amplitude of only one pulse centered at the sampling instant with no interference from the remaining pulses.

There exists one and only one pulse that meets Nyquist criterion above and it is given by sinc function as $p(t) = \sin c(2\,B_o t) = \dfrac{\sin(2\pi B_o t)}{2\pi B_o t}$

where, $B_o = \dfrac{1}{2T_b}$ and $T_b = \dfrac{1}{R_b}$, R_b is the bit rate.

The parameter B_0 is called the Nyquist bandwidth. It is the minimum transmission bandwidth required for zero ISI. Using this sinc pulse, we can transmit at a rate of R_b pulses per second without ISI over a bandwidth of $\dfrac{1}{2T_b}$ or $\dfrac{R_b}{2}$. A sinc pulse is shown below:



Sampling intervals

The above pulse is the ideal basic pulse shape for zero ISI. Such a pulse shape $p(t)$ achieves economy in bandwidth since it solves the problem of zero ISI with the minimum bandwidth possible.

b) What are the limitations of Nyquist pulse? How is it solved using Raised Cosine Pulse. **[MODEL QUESTION]**

Answer:

Limitations of Ideal Solution

The above criterion gives an ideal solution for zero ISI which is practically unrealizable. For the idealized pulse shape the frequency response of the filter $P(f)$ is needed to be flat in the range $-B_0$ to $+B_0$ and zero elsewhere. This abrupt transitions at $\pm B_0$ is physically unrealizable. The ideal amplitude response is shown below:

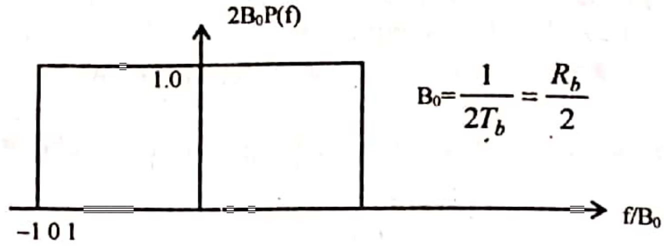


Fig: Ideal amplitude response in the frequency domain

The second limitation is that the function $p(t)$ decreases as $\frac{1}{|t|}$ for large $|t|$. This results in a slow rate of decay. This is caused by the discontinuity of $p(f)$ at $\pm B_0$. Thus there is practically no margin of error in sampling times in the receiver.
The above limitations of the ideal solution can be solved with the help of Raised Cosine function.

Raised Cosine Spectrum
There are two practical difficulties encountered in using an ideal pulse shape of sinc function to combat Inter symbol interference. One difficulty is that it is impossible to achieve an ideal band limited pulse shape and the other difficulty is that even a very small change in sampling times causes bit error. A raised cosine pulse is a pulse shape, which overcomes these practical difficulties. This is, however, achieved by extending bandwidth from the minimum value of $R_b/2$ to an adjustable value between $\frac{R_b}{2}$ and R_b.
It is easily seen that there is a need for overall frequency response to decrease towards zero gradually rather than abruptly. Thus the frequency response $P(f)$ should have a flat portion and a roll-off portion. Raised cosine function is one such possible form of $P(f)$ which is defined by

$$P(f) = \begin{cases} \frac{1}{B_0} & \text{for } 0 < |f| < f_1 \\ \frac{1}{4B_0}\left\{1 + \cos\left[\frac{\pi|f| - f_1}{2B_0 - 2f_1}\right]\right. & \text{for } f_1 \le |f| < 2B_0 - f_1 \\ 0 & \text{for } |f| \ge 2B_0 - f_1 \end{cases}$$

where, B_0 = Nyquist Bandwidth = $\frac{1}{2T_b}$

The portion $P(f) = \frac{1}{2B_0}$ for $|f| < f_1$ represents the flat portion and the sinusoidal portion is the roll off portion of the raised cosine function.

The frequency f_1 and bandwidth B_0 are related as $1-\frac{f_1}{B_0}=\alpha$

where α is a parameter called the roll-off factor.
For $\alpha=0, f_1=B_0$ and for $\alpha=1, f_1=2B_o$. The normalized frequency response $2B_0\,P(f)$ is plotted against f/B_0 which is shown below for various values of α

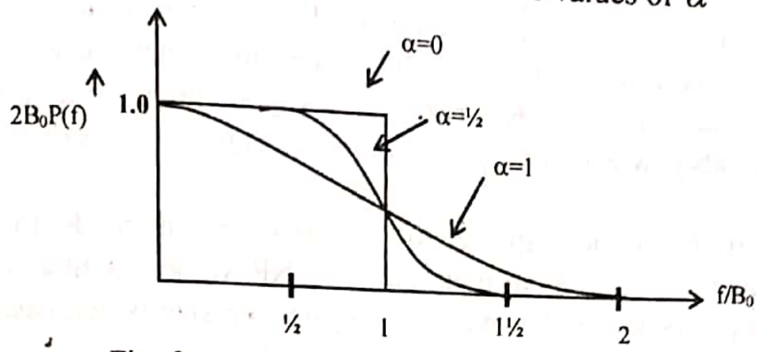


Fig: frequency response of a raised cosine filter

For $a=0, f_1=B_o$ and $P(f)$ has an abrupt variation at B_0. This results is minimum bandwidth ideal solution of zero ISI.

For non-zero values of α, the function P(f) cuts off gradually and it is therefore easier to realize in practice. Moreover, the function P(f) shows odd symmetry about the cut-off frequency B_0 of the ideal low-pass filter as shown for $\alpha=1$ and $\alpha=\frac{1}{2}$.

c) A communication channel of bandwidth 75 kHz is required to transmit binary data at a rate of 0.1 Mbps using raised cosine pulses. Determine the roll- off factor.
[MODEL QUESTION]

Answer:
Bit rate, $R_b = 0.1$ Mbps
We know transmission bandwidth, $B = B_0(1+\alpha)$
where B_0 = Nyquist Bandwidth and α = Roll-off factor

Also $B_0=\frac{R_b}{2}=\frac{0.1}{2}\,MHz=0.05\,MHz=50\,KHz$

Here $B = 75$ KHz

Hence $75 = 50\,(1+\alpha)$ or, $1+\alpha=\frac{75}{50}=1.5$

So, $\alpha = 1.5-1 = 0.5$

7. a) How is non-uniform quantization utilized. **[MODEL QUESTION]**
b) Deduce the relation of signal to quantization noise.
c) Calculate the number of quantization levels for the signal $x(t)=5\sin(500\pi+\theta)$.

Answer:

a) If the step size, Δ, is kept constant in the quantization process, the quantization is said to be linear quantization. In this case there is uniform separation between the quantizing levels. Linear quantization is not always desirable since SNR decreases with a decrease in input power level relative to the overload point of the quantizer. In certain applications such as in transmission of speech signals using PCM, the input signals vary over a wide range from weak signal to loud signal. The range of voltages from the peaks of loud talk to the weak talk is of the order of 1000 : 1. If a linear quantizer is used it will give high SNR for the loud signal but low SNR for the weak signal. Thus the quality of the signal deteriorates considerably when one speaks quietly. This is a disadvantage of linear quantization.

To overcome this problem the step size of the quantizer is made to vary with the amplitude of the input signal so as to maintain the SNR value essentially constant for a wide range of input power levels. For weak signal the step size is decreased and for loud signal the step size is increased. In fact, the ratio σ_x^2/Δ^2 is kept constant within the dynamic range of the input signal. Such a quantization is known as non-linear or non-uniform quantization. A non-linear quantizer is designed in such a way that the step size increases as the separation from the origin of the input-output amplitude characteristic is increased. The weak signals which need more protection are favoured at the expense of the loud signals. In this way a nearly uniform percentage precision is achieved throughout the amplitude range of the input signal. The diagrams below show the differences between linear and non-linear quantization.

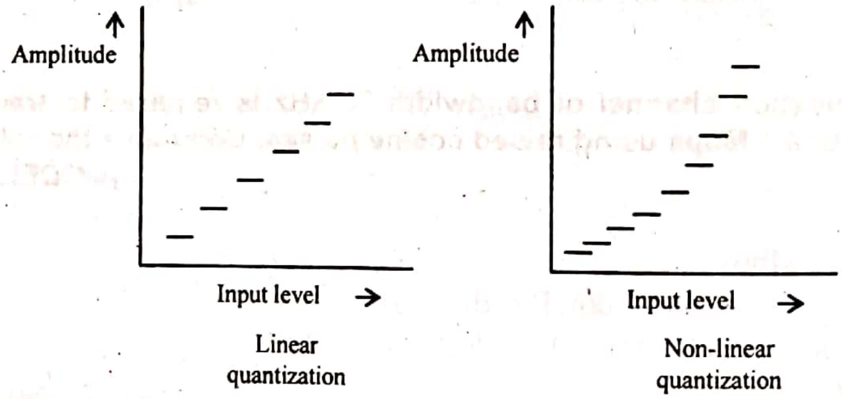


Fig: Linear vs. non-linear quantization

b) Calculation of signal- to-quantization noise ratio (SQR) for sinusoidal input signal

Let the signal power = P_S and

Average quantization noise output power = P_q

If Δ is the step size of the linear quantization, then $P_q = \dfrac{\Delta^2}{12}$

Thus the $SQR = \dfrac{P_s}{P_q} = \dfrac{P_s}{\Delta^2/12}$ where SQR = signal to quantization noise ratio

Let V_r = rms voltage of the input analog signal, then

$$SQR = \frac{V_r^2/R}{\Delta^2/12}$$ where R = resistance of the load

If R = 1, then

$$SQR = \frac{V_r^2}{\Delta^2/12} = \frac{12V_r^2}{\Delta^2} = 12\left(\frac{V_r}{\Delta}\right)^2$$

$$= 10\log 12 + 20\log\left(\frac{V_r}{\Delta}\right) \text{ dB} = 10.8 + 20\log\frac{V_r}{\Delta} \text{ dB}$$

If the input signal is sinusoidal, then $V_r = \frac{V_m}{\sqrt{2}}$ where V_m is the maximum value of the sinusoidal signal.

Thus in the case of sinusoidal input

$$SQR = \frac{\left(V_m/\sqrt{2}\right)^2}{\Delta^2/12}$$

$$= 10\log\left[\frac{\left(V_m/\sqrt{2}\right)^2}{\Delta^2/12}\right] \text{ dB} = 10\log\left[6\left(\frac{V_m}{\Delta}\right)^2\right] \text{ dB}$$

$$= 10\log 6 + 20\log\frac{V_m}{\Delta} \text{ dB} = 7.78 + 20\log\frac{V_m}{\Delta} \text{ dB}$$

If L = Number of representation levels or steps, then

$$\Delta = \frac{2V_m}{L}$$ and

$$SQR = 10\log\left[\frac{\left(V_m/\sqrt{2}\right)^2}{\left(\frac{4V_m^2}{12L^2}\right)}\right] \text{dB}$$

$$= 10\log 1.5L^2 \text{ dB}$$
$$= 10\log 1.5 + 20\log L \text{ dB}$$
$$= 1.76 + 20\log L \text{ dB}$$

Since $L = 2^n$ where n = number of bits used

$$SQR = 10\log 1.5 + 20\log 2^n$$
$$= 1.76 + 20n\log 2 \text{ dB} = 1.76 + 6.02n \text{ dB}$$

Thus for every additional bit it gives an improvement of 6 dB in SQR. This is known as 6 dB rule of PCM. SQR can also be written as SNR.

c) Let Δ be the step size.

Then the number of quantization level is given by $L = \frac{2V_m}{\Delta}$ where V_m = Amplitude of the signal.

Here Amplitude of the signal, $V_m = 5$ volts

So, $L = \frac{2\times 5}{\Delta} = \frac{10}{\Delta}$

8. a) What is companding? Discuss the two laws of companding.
b) An audio output of a microphone, m(t), swings between ±V volts and is characterized by uniform probability density function. If the signal is sampled and uniformly quantized by an R-bit quantizer, derive the expression for output signal-to-noise ration in dB in terms of R. **[MODEL QUESTION]**

Answer:

a) 1st Part:

The combined process of compression at the transmitter and expansion at the receiver is called companding.

2nd Part:

Two types of compression laws are in use. These are, namely, μ-law companding and A-Law companding. μ-law companding is used in U.S.A., Canada and Japan. A-law companding recommended by CCITT is used in India and European countries.

μ-law is defined by the expression

$$|v_2| = \frac{\log(1+\mu|v_1|)}{\log(1+\mu)}$$

where v_1 = normalized input voltage
v_2 = normalized output voltage
μ = a positive constant.

The μ-law compression characteristic is shown below:

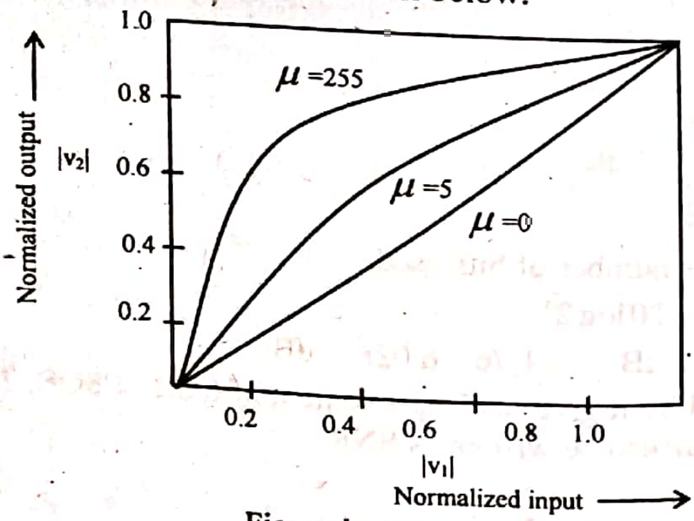


Fig: μ–law compression characteristic

$\mu = 0$ corresponds to uniform quantization. A practical value for μ is 255. μ - law is neither strictly linear and nor strictly logarithmic. It is approximately linear at low input levels and approximately logarithmic at high input levels.

A-law companding is defined by the following expression.

$$|v_2| = \begin{cases} \dfrac{A|v_1|}{1+\log A} & \text{for } 0 \le |v_1| \le \dfrac{1}{A} \\ \dfrac{1+\log(A|v_1|)}{1+\log A} & \text{for } \dfrac{1}{A} \le |v_1| < 1 \end{cases}$$

where $|v_1|$ = normalised input voltage

$|v_2|$ = normalised output voltage

A = a constant

A-law compressor characteristic is shown in the figure below:

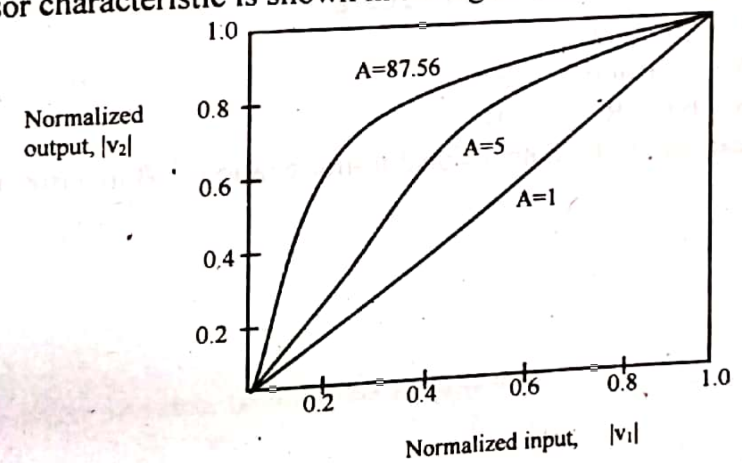


Fig: A-law compression characteristic

The A-law compression characteristic is made up of a linear segment for low-level inputs and a logarithmic segment for high level inputs. The special case A = 1 corresponds to uniform quantization. A practical value for A is 87.56.

A-law companding is inferior to μ-law in terms of small-signal quality i.e. ideal channel noise.

b) Let us assume sinusoidal input signal of amplitude V volts. Signal to quantization noise ratio, SQR is given by

$SQR = \dfrac{P_s}{P_q}$ where P_s = signal power and P_q = quantization noise power.

$P_s = V_r^2 / R$ where V_r = rms value of the input signal

or, $P_s = \dfrac{(V/\sqrt{2})^2}{R}$

It R = 1, P_s is normalized to $P_s = \frac{V^2}{2}$

$$P_q = \frac{\Delta^2}{12} \text{ where } \Delta = \text{step size}$$

Hence, $SQR = \frac{V^2/2}{\Delta^2/12} = (6)\left(\frac{V}{\Delta}\right)^2$

But $\quad \Delta = \frac{2V}{L}$ where L = No. of representation level

So $\quad SQR = (6)\left(\frac{L^2}{4}\right) = 1.5\,L^2$

$L = 2^R$ where R = number of bits used

Hence $SQR|_{dB} = 10\log 1.5 + 20\log L = 1.76 + 20\log L$

So, $\quad SQR = 1.76 + 20\log 2^R \quad$ dB

or, $\quad SQR = 1.76 + 20\,R\log 2 \quad$ dB

or, $\quad SQR = 1.76 + 6.02\,R \quad$ dB

This is the required expression for output signal-to-noise ratio in dB in terms of R.

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