SHEKHAWATI INSTITUTE OF ENGINEERING AND TECHNOLOGY, SIKAR, (RAJASTHAN)

1ST MID TERM EXAMINATION 2017-18 (3RD YEAR ECE)

Subject code & Name: **6EC4A DIGITAL COMMUNICATION**

MM: 20 Time: 1:30 hr

Student Instructions

1. Use pencil for diagrams.
2. **1ST question is compulsory. Attempt any four questions from rest.**
3. **All questions carry equal marks**
4. Write your ID before starting question paper.

6EC4A DIGITAL COMMUNICATION

Q.1. What are the factors due to which Non- uniform quantization is needed? How these factors are dealt with in non-uniform quantization?

Q.2. State and prove sampling theorem.

Q.3. What are the different types of line coding system being used in digital communication? Show with an example of 8-bit binary data.

Q.4. Derive a relation for signal to quantization noise ratio for a PCM system.

Q.5. Describe modulation and demodulation process of DPCM.

Q.6. What is delta modulation? what are the drawbacks of delta modulation? Name and describe the method through which these drawbacks can be removed?

Q.7. Explain the term: Inter symbol interference

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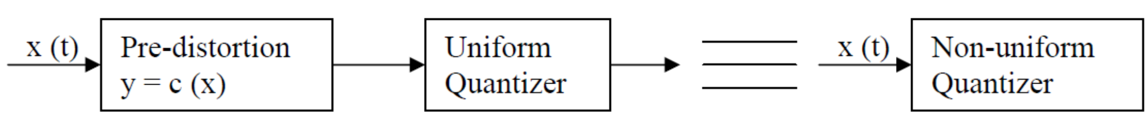
Q.1. What are the factors due to which Non- uniform quantization is needed? How these factors are dealt with in non-uniform quantization?

Solution: Granular distortion and overload distortion: Often the design of a quantizer involves supporting only a limited range of possible output values and performing clipping to limit the output to this range whenever the input exceeds the supported range. The error introduced by this clipping is referred to as overload distortion. Within the extreme limits of the supported range, the amount of spacing between the selectable output values of a quantizer is referred to as its granularity, and the error introduced by this spacing is referred to as granular distortion. It is common for the design of a quantizer to involve determining the proper balance between granular distortion and overload distortion. For a given supported number of possible output values, reducing the average granular distortion may involve increasing the average overload distortion, and vice versa. A technique for controlling the amplitude of the signal or, equivalently, the quantization step size to achieve the appropriate balance is the use of automatic gain control (AGC). However, in some quantizer designs, the concepts of granular error and overload error may not apply (e.g., for a quantizer with a limited range of input data or with a countably infinite set of selectable output values).

A non-uniform quantizer ensures smaller quantization error for small amplitude of the input signal and relatively larger step size when the input signal amplitude is large.



There are two popular standards for non-linear quantization known as

(a) The μ - law companding
(b) The A – law companding.

The μ - law has been popular in the US, Japan, Canada and a few other countries while the A - law is largely followed in Europe and most other countries, including India, adopting ITU-T standards.

The compression function c (x) for μ - law companding is (**Fig. 3.12.4** and **Fig. 3.12.5**):

$$\frac{c(|x|)}{V} = \frac{\ln\left(1+\frac{\mu|x|}{V}\right)}{\ln(1+\mu)}, \qquad 0 \le \frac{|x|}{V} \le 1.0 \qquad 3.12.13$$

'μ' is a constant here. The typical value of μ lies between 0 and 255. μ = 0 corresponds to linear quantization.

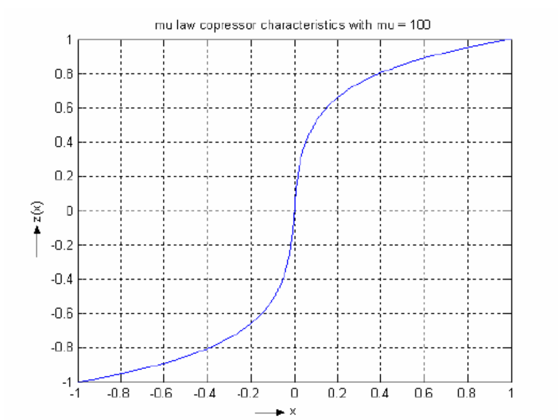


Fig. 3.12.4 *μ-law companding characteristics(mu = 100)*

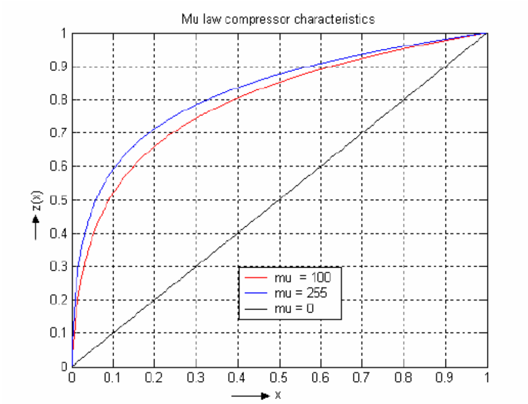


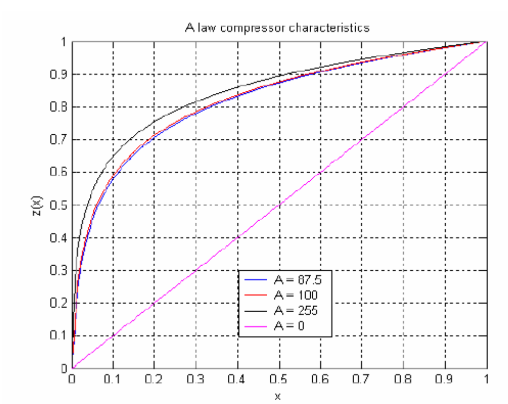
Fig. 3.12.5 *μ-law companding characteristics (mu = 0, 100, 255)*

The compression function c (x) for A - law companding is (**Fig. 3.12.6**):

$$\frac{c(|x|)}{V} = \frac{A\frac{|x|}{V}}{1+\ln A}, \qquad 0 \le \frac{|x|}{V} \le \frac{1}{A}$$

$$= \frac{1+\ln\left(A\frac{|x|}{V}\right)}{1+\ln A}, \qquad \frac{1}{A} \le \frac{|x|}{V} \le 1.0 \qquad 3.12.14$$

'A' is a constant here and the typical value used in practical systems is 87.5.



Q.2. State and prove sampling theorem.

Solution: Part - I If a signal x(t) does not contain any frequency component beyond W Hz, then the signal is completely described by its instantaneous uniform samples with sampling interval (or period) of $T_S < 1/(2W)$ sec.

Part – II The signal x(t) can be accurately reconstructed (recovered) from the set of uniform instantaneous samples by passing the samples sequentially through an ideal (brick-wall) low pass filter with bandwidth B, where $W \le B < fs - W$ and $fs = 1/(T_S)$.

If x(t) represents a continuous-time signal, the equivalent set of instantaneous uniform samples {x(n)} may be represented as,

{x(nTs)}≡ xs(t) = Σ x(t).δ(t- nTs) ….. (1)

where x(nTs) = x(t)| t =nTs , δ(t) is a unit pulse singularity function and 'n' is an integer Conceptually, one may think that the continuous-time signal x(t) is multiplied by an (ideal) impulse train to obtain {x(nTs)} as (1.2.1) can be rewritten as,

xs(t) = x(t).Σ δ(t- nTs) …………………..(2)

Now, let X(f) denote the Fourier Transform F(T) of x(t), i.e.

$$X(f) = \int_{-\infty}^{+\infty} x(t).\exp(-j2\pi ft)dt$$

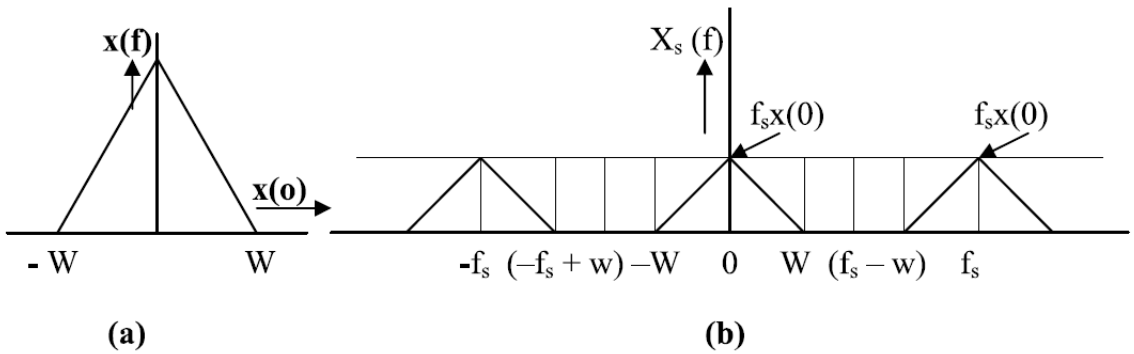
Now, from the theory of Fourier Transform, we know that the F.T of Σ δ(t- nTs), the impulse train in time domain, is an impulse train in frequency domain:

$$F\{\Sigma\ \delta(t- nT_s)\} = (1/T_s).\Sigma\ \delta(f- n/T_s) = f_s.\Sigma\ \delta(f- nf_s)$$

If Xs(f) denotes the Fourier transform of the energy signal xs(t), we can write and the convolution property:

$$\begin{aligned} X_s(f) &= X(f)^*\ F\{\Sigma\ \delta(t- nT_s)\} \\ &= X(f)^*[f_s.\Sigma\ \delta(f- nf_s)] \\ &= f_s.X(f)^*\Sigma\ \delta(f- nf_s) \\ &= f_s. \int_{-\infty}^{+\infty} X(\lambda).\Sigma\ \delta(f- nf_s-\lambda)d\lambda = f_s.\ \Sigma \int X(\lambda).\delta(f- nf_s-\lambda)d\lambda = f_s.\Sigma\ X(f- nf_s) \end{aligned}$$

This equation, when interpreted appropriately, gives an intuitive proof to Nyquist's theorems as stated above and also helps to appreciate their practical implications. Let us note that, we assumed that x(t) is an energy signal so that its Fourier transform exists. Further, the impulse train in time domain may be viewed as a periodic singularity function with almost zero (but finite) energy such that its Fourier Transform [i.e. a train of impulses in frequency domain] exists. With this setting, if we assume that x(t) has no appreciable frequency component greater than W Hz and if fs > 2W, then it implies that Xs(f), the Fourier Transform of the sampled signal xs(t) consists of infinite number of replicas of X(f), centered at discrete frequencies n.fs, -∞ < n < ∞ and scaled by a constant fs= 1/Ts .



Q.3. What are the different types of line coding system being used in digital communication? Show with an example of 8-bit binary data.

Solution:

NRZ–L Non return to zero level. This is the standard positive logic signal format used in digital circuits.

1 forces a high level

0 forces a low level

NRZ–M Non return to zero mark

1 forces a transition

0 does nothing (keeps sending the previous level)

NRZ–S Non return to zero space

1 does nothing (keeps sending the previous level)

0 forces a transition

RZ Return to zero

1 goes high for half the bit period and returns to low

0 stays low for the entire period

Biphase–L Manchester. Two consecutive bits of the same type force a transition at the beginning of a bit period.

1 forces a negative transition in the middle of the bit

0 forces a positive transition in the middle of the bit

Biphase–M Variant of Differential Manchester. There is always a transition halfway between the conditioned transitions.

1 forces a transition

0 keeps level constant

Biphase–S Differential Manchester used in Token Ring. There is always a transition halfway between the conditioned transitions.

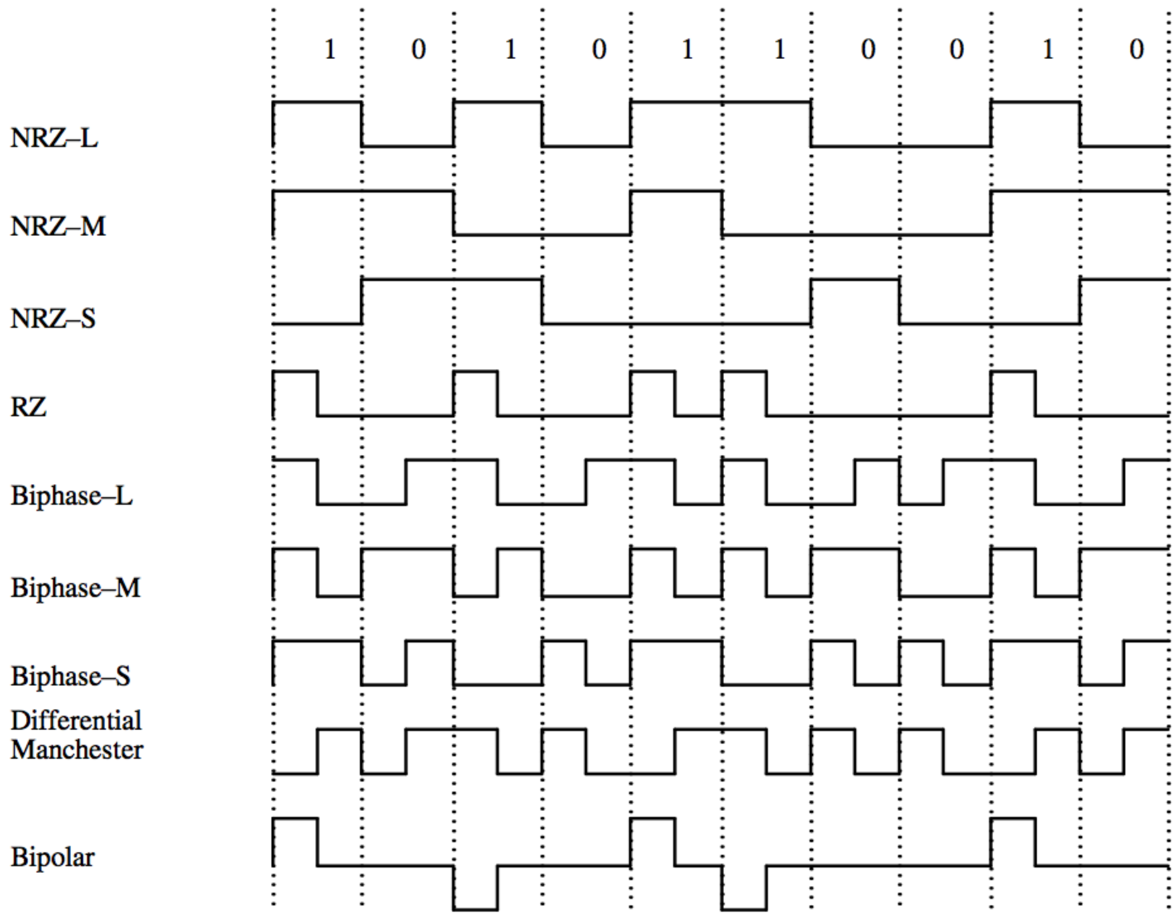
1 keeps level constant

0 forces a transition

Bipolar The positive and negative pulses alternate.

1 forces a positive or negative pulse for half the bit period

0 keeps a zero level during bit period


1 0 1 0 1 1 0 0 1 0
NRZ–L
NRZ–M
NRZ–S
RZ
Biphase–L
Biphase–M
Biphase–S
Differential Manchester
Bipolar

Q.4. Derive a relation for signal to quantization noise ratio for a PCM system.

SQNR for uniform quantizer

In **Fig.3.11.1** $x(kT_s)$ represents a discrete time ($t = kT_s$) continuous amplitude sample of x(t) and $x_q(kT_s)$ represents the corresponding quantized discrete amplitude value. Let e_k represents the error in quantization of the k^{th} sample i.e.

$$e_k = x_q(kT_s) - x(kT_s) \qquad 3.11.1$$

Let,

M = Number of permissible levels at the quantizer output.

N = Number of bits used to represent each sample.

±V = Permissible range of the input signal x (t).

Hence,

$M=2^N$ and,

$M.\delta \cong 2.V$ [Considering large M and a mid-riser type quantizer]

Let us consider a small amplitude interval dx such that the probability density function (pdf) of x(t) within this interval is p(x). So, p(x)dx is the probability that x(t) lies in the range $(x-\frac{dx}{2})$ and $(x+\frac{dx}{2})$. Now, an expression for the mean square quantization error $\overline{e^2}$ can be written as:

$$\overline{e^2} = \int_{x_1-\delta/2}^{x_1+\delta/2} p(x)(x-x_1)^2 dx + \int_{x_2-\delta/2}^{x_2+\delta/2} p(x)(x-x_2)^2 dx + ... \qquad 3.11.2$$

For large M and small δ we may fairly assume that p(x) is constant within an interval, i.e. $p(x) = p_1$ in the 1^{st} interval, $p(x) = p_2$ in the 2^{nd} interval, …., $p(x) = p_k$ in the k^{th} interval.

Therefore, the previous equation can be written as

$$\overline{e^2} = (p_1 + p_2 + ...) \int_{-\delta/2}^{\delta/2} y^2 dy$$

Where, $y = x - x_k$ for all 'k'.

So,

$$\overline{e^2} = (p_1 + p_2 + ...)\frac{\delta^3}{12}$$

$$= [(p_1 + p_2 + ...)\delta]\frac{\delta^2}{12}$$

Now, note that $(p_1 + p_2 + \ldots + p_k + \ldots)\delta = 1.0$

$$\therefore \overline{e^2} = \frac{\delta^2}{12}$$

The above mean square error represents power associated with the random error signal. For convenience, we will also indicate it as N_Q.

Calculation of Signal Power (S_i)

After getting an estimate of quantization noise power as above, we now have to find the signal power. In general, the signal power can be assessed if the signal statistics (such as the amplitude distribution probability) is known. The power associated with x (t) can be expressed as

$$S_i = \overline{x^2(t)} = \int_{-V}^{+V} x^2(t) p(x)\, dx$$

where p (x) is the pdf of x (t). In absence of any specific amplitude distribution it is common to assume that the amplitude of signal x (t) is uniformly distributed between ±V.

In this case, it is easy to see that

$$S_i = \overline{x^2(t)} = \int_{-V}^{+V} x^2(t)\frac{1}{2V}dx = \left[\frac{x^3}{3.2V}\right]_{-V}^{+V} = \frac{V^2}{3} = \frac{(M\delta)^2}{12}$$

Now the SNR can be expressed as,

$$\frac{S_i}{N_Q} = \frac{\frac{V^2}{3}}{\frac{\delta^2}{12}} = \frac{\frac{(M\delta)^2}{12}}{\frac{\delta^2}{12}} = M^2$$

It may be noted from the above expression that this ratio can be increased by increasing the number of quantizer levels N.

Also note that S_i is the power of x (t) at input of the sampler and hence, may not represent the SQNR at the output of the low pass filter in PCM decoder. However, for large N, small δ and ideal and smooth filtering (e.g. Nyquist filtering) at the PCM decoder, the power S_o of desired signal at the output of the PCM decoder can be assumed to be almost the same as S_i i.e.,

$$S_o \simeq S_i$$

With this justification the SQNR at the output of a PCM codec, can be expressed as,

$$SQNR = \frac{S_o}{N_Q} \simeq M^2 = \left(2^N\right)^2 = 4^N$$

and in dB,

$$\left.\frac{S_o}{N_Q}\right|_{dB\triangleright} = 10\log_{10}\left(\frac{S_o}{N_Q}\right) \simeq 6.02NdB$$

Q.5. Describe modulation and demodulation process of DPCM.

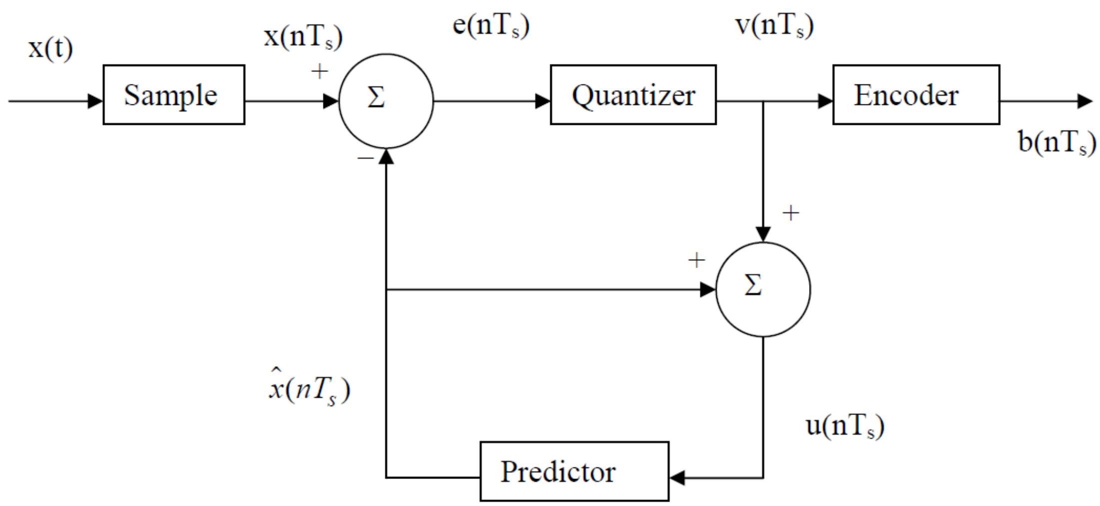
Differential pulse-code modulation (DPCM) is a signal encoder that uses the baseline of pulse-code modulation (PCM) but adds some functionalities based on the prediction of the samples of the signal. The input can be an analog signal or a digital signal.

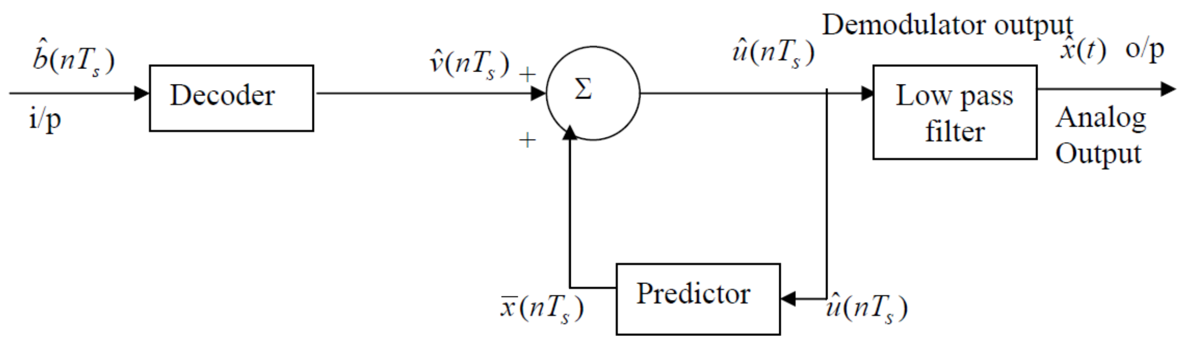
If the input is a continuous-time analog signal, it needs to be sampled first so that a discrete-time signal is the input to the DPCM encoder.

Option 1: take the values of two consecutive samples; if they are analog samples, quantize them; calculate the difference between the first one and the next; the output is the difference, and it can be further entropy coded.

Option 2: instead of taking a difference relative to a previous input sample, take the difference relative to the output of a local model of the decoder process; in this option, the difference can be quantized, which allows a good way to incorporate a controlled loss in the encoding.

Applying one of these two processes, short-term redundancy (positive correlation of nearby values) of the signal is eliminated; compression ratios on the order of 2 to 4 can be achieved if differences are subsequently entropy coded, because the entropy of the difference signal is much smaller than that of the original discrete signal treated as independent samples.





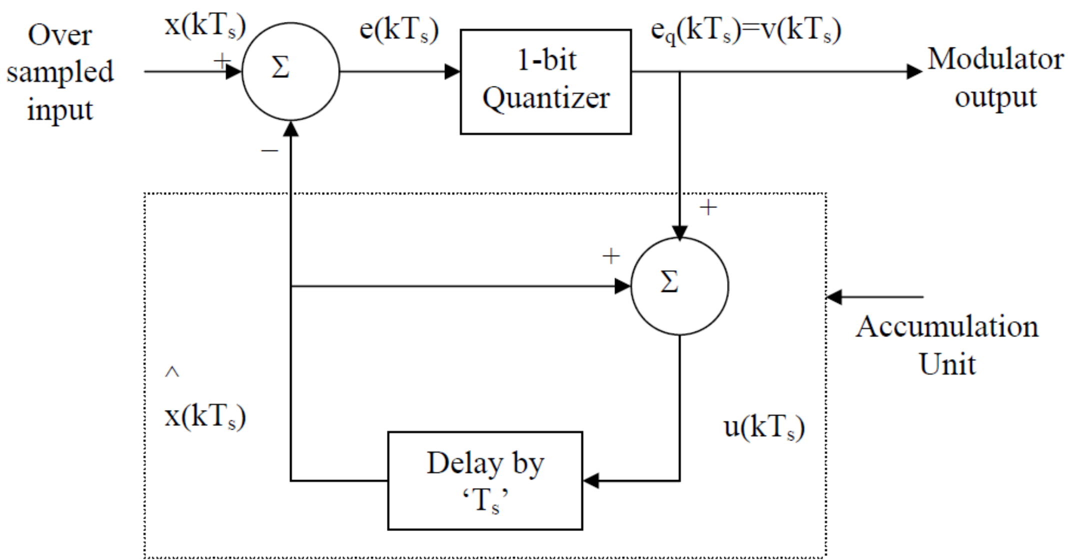
Q.6. What is delta modulation? what are the drawbacks of delta modulation? Name and describe the method through which these drawbacks can be removed?

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

a. **The analog signal is approximated with a series of segments.**
b. **Each segment of the approximated signal is compared of successive bits is determined by this comparison.**

c. **Only the change of information is sent, that is, only an increase or decrease of the signal amplitude from the previous sample is sent whereas a no-change condition causes the modulated signal to remain at the same 0 or 1 state of the previous sample.**

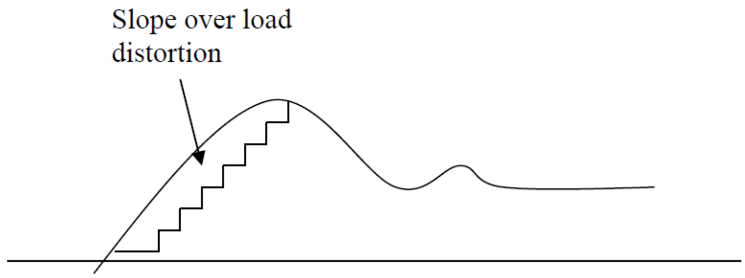
To achieve high signal-to-noise ratio, delta modulation must use oversampling techniques, that is, the analog signal is sampled at a rate several times higher than the Nyquist rate.



However DM also suffers from a few limitations such as the following:

a) **Slope over load distortion: If the input signal amplitude changes fast, the step-by-step accumulation process may not catch up with the rate of change . This happens initially when the demodulator starts operation from cold-start but is usually of negligible effect for speech. However, if this phenomenon occurs frequently (which indirectly implies smaller value of auto-correlation co-efficient Rxx(τ) over a short time interval) the quality of the received signal suffers. The received signal is said to suffer from slope-overload distortion. An intuitive remedy for this problem is to increase the step-size δ but that approach has another serious lacuna as noted in b).**

b)



Granular noise: If the step-size is made arbitrarily large to avoid slope-overload distortion, it may lead to 'granular noise'. Imagine that the input speech signal is fluctuating but very close to zero over limited time duration. This may happen due to pauses between sentences or else. During such moments, our delta modulator is likely to produce a fairly long sequence of 101010…., reflecting that the accumulator output is close but alternating around the input signal. This phenomenon is manifested at the output of the delta demodulator as a small but perceptible noisy background. This is known as 'granular noise'. An expert listener can recognize the crackling sound. This noise should be kept well within a tolerable limit while deciding the step-size. Larger step-size increases the granular noise while smaller step size increases the degree of slope-overload distortion. In the first level of design, more care is given to avoid the slope-overload distortion. We will briefly discuss about this approach while keeping the step-size fixed. A more efficient approach of adapting the step-size, leading to Adaptive Delta Modulation (ADM) , is excluded.

Adaptive delta modulation or [continuously (CVSD) is a modification of DM in which the step size is not fixed. Rather, when several consecutive bits have the same direction value, the encoder and decoder assume that slope overload is occurring, and the step size becomes progressively larger.

Q.7. Explain the terms: Inter symbol interference

Solution: Inter symbol interference: In telecommunication, intersymbol interference (ISI) is a form of distortion of a signal in which one symbol interferes with subsequent symbols. This is an unwanted phenomenon as the previous symbols have similar effect as noise, thus making the communication less reliable. The spreading of the pulse beyond its allotted time interval causes it to interfere with neighboring pulses. ISI is usually caused by multipath propagation or the inherent linear or non-linear frequency response of a channel causing successive symbols to "blur" together. The presence of ISI in the system introduces errors in the decision device at the receiver output. Therefore, in the design of the transmitting and receiving filters, the objective is to minimize the effects of ISI, and thereby deliver the digital data to its destination with the smallest error rate possible.

Causes : 1) Multipath propagation

One of the causes of intersymbol interference is multipath propagation in which a wireless signal from a transmitter reaches the receiver via multiple paths. The causes of this include reflection (for instance,

the signal may bounce off buildings), refraction (such as through the foliage of a tree) and atmospheric effects such as atmospheric ducting and ionospheric reflection. Since the various paths can be of different lengths, this results in the different versions of the signal arriving at the receiver at different times. These delays mean that part or all of a given symbol will be spread into the subsequent symbols, thereby interfering with the correct detection of those symbols. Additionally, the various paths often distort the amplitude and/or phase of the signal, thereby causing further interference with the received signal.

2) Bandlimited channels: Another cause of intersymbol interference is the transmission of a signal through a bandlimited channel, i.e., one where the frequency response is zero above a certain frequency (the cutoff frequency). Passing a signal through such a channel results in the removal of frequency components above this cutoff frequency. In addition, components of the frequency below the cutoff frequency may also be attenuated by the channel. This filtering of the transmitted signal affects the shape of the pulse that arrives at the receiver. The effects of filtering a rectangular pulse not only change the shape of the pulse within the first symbol period, but it is also spread out over the subsequent symbol periods. When a message is transmitted through such a channel, the spread pulse of each individual symbol will interfere with following symbols. Bandlimited channels are present in both wired and wireless communications. The limitation is often imposed by the desire to operate multiple independent signals through the same area/cable; due to this, each system is typically allocated a piece of the total bandwidth available. For wireless systems, they may be allocated a slice of the electromagnetic spectrum to transmit in (for example, FM radio is often broadcast in the 87.5 MHz - 108 MHz range). This allocation is usually administered by a government agency; in the case of the United States this is the Federal Communications Commission (FCC). In a wired system, such as an optical fiber cable, the allocation will be decided by the owner of the cable.

The bandlimiting can also be due to the physical properties of the medium - for instance, the cable being used in a wired system may have a cutoff frequency above which practically none of the transmitted signal will propagate.

Communication systems that transmit data over bandlimited channels usually implement pulse shaping to avoid interference caused by the bandwidth limitation. If the channel frequency response is flat and the shaping filter has a finite bandwidth, it is possible to communicate with no ISI at all. Often the channel response is not known beforehand, and an adaptive equalizer is used to compensate the frequency response.