UNIT II

PULSE MODULATION

Low pass sampling theorem – Quantization – PAM – Line coding – PCM, DPCM, DM, and ADPCM And ADM, Channel Vocoder - Time Division Multiplexing, Frequency Division Multiplexing

1.(a)Briefly explain Low pass sampling theorem.

Or

(b)Explain low pass sampling theorem with suitable diagrams.

Sampling Theorem:-

" A band limited signal having no spectral components above f_m Hz can be determined uniquely by values sampled at uniform intervals of "

$$T_S < 1/2fm \text{ sec}$$

This theorem is also known as the uniform sampling theorem.

Nyquist Rate:- Nyquist Rate of sampling which gives the "minimum frequency needed to reconstruct the analog signal from sampled waveforms". $f_s \geq 2f_m$

Low pass sampling theorem

❖The implementation of a sampler is most commonly done with a SAMPLE and HOLD Circuit.

❖ In this operation a switch a switch and storage mechanism(for example, a transistor and a capacitor) is used to form a sequence of samples of analog input waveform.

❖ These samples look like a PAM waveform. The original analog waveform can be recovered from these PAM type samples simply by low pass filtering them.

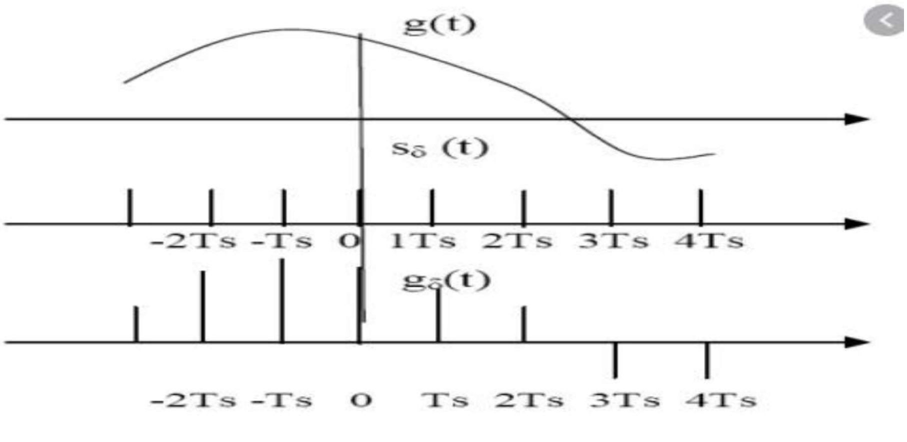


Fig : Sampling process

If we undersample (ie $f_s < f_{nyquist}$) then aliasing (ie overlapping of adjacent

spectrum replicates) occurs. The aliased spectral components appear in the frequency band between(fs-fm) and fm.

There are two ways

i) Prefiltering antialiasing filter

ii) Postfiltering antialiasing filters

These two methods make use of antialiasing filter.

In the first method called prefiltering antialiasing filter the analog signal itself is prefiltered so that the new maximum frequency $f_{m'}$ is reduced to fs/2 or less

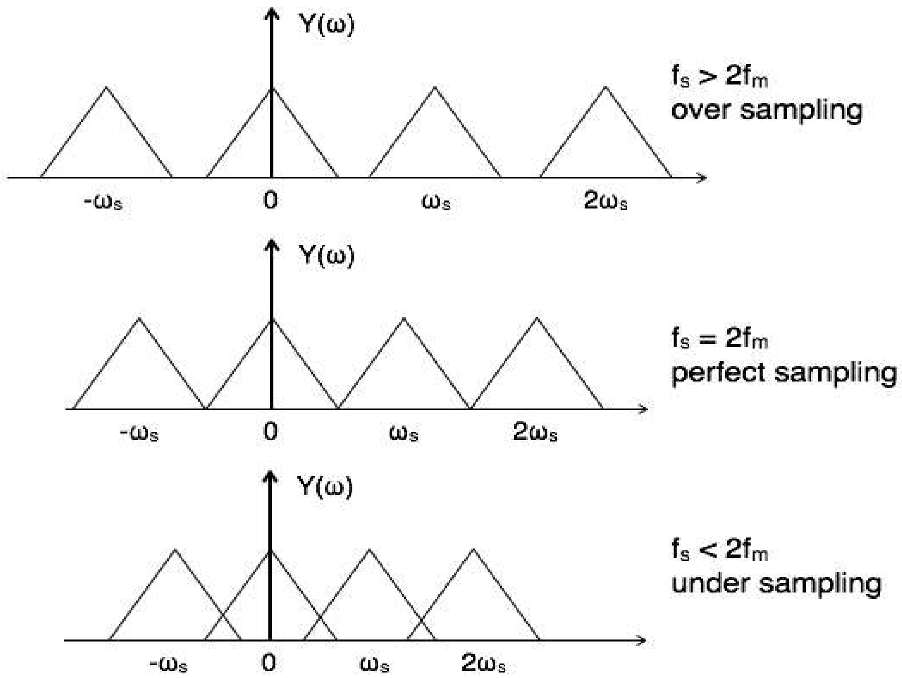
In the second method called post filtering antialiasing filter the aliased terms are eliminated after sampling with help of a low pass filter operation.

Disadvantage:- some information is lost due to filtering.

Transition Bandwidth:- All realizable filters required a non zero bandwidth for the transition between the pass band and the stop band commonly known as the transition bandwidth.

If we account for the 20% transition bandwidth the antialiasing filter we have a Nyquist sampling rate $f_s \geq 2.2f_m$

The large number of samples cab be filtered further by digital filters instead of analog filters.The steps required in A/D conversion for both the cases of undesampling and oversampling



Without oversampling

❖ The signal passes through a high performance analog lowpass antialiasing filter to limit its bandwidth. The antialiasing filter has a pass band equal to the signal bandwidth plus the transition bandwidth. The Nyquist Sampling rate $2f_m$ becomes $2f_m+f_t$.

- The filtered signal is sampled at the Nyquist rate for the approximately band limited signal obtained in the previous step.
- The samples are processed by an analog to digital converter that maps the continuous valued samples to a finite list of discrete output levels.

The sample rate at the output of the digital filter is reduced in proportion to the bandwidth reduction obtained by digital filter.

QUANTIZATION

2.(a)Explain the different types of quantization.

Or

(b).Explain uniform and non uniform quantization.

The conversion of the analog form of the signal to discrete form takes place in Quantiser.The sampled analog signal is still analog, because though the discretised in time, the signal amplitude can take any value.The quantizer forces the signal to take some discrete values from the continuous values.From the sampled signal $m_s(t)$ a new quantized signal $m_q(t)$ is created.$m_s(t)$ can take any value $m_q(t)$ can take only discrete values.

Types of Quantization

There are two types of Quantization - Uniform Quantization and Non-uniform Quantization.

The type of quantization in which the quantization levels are uniformly spaced is termed as a **Uniform Quantization**. The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a **Non-uniform Quantization**.

There are two types of uniform quantization. They are Mid-Rise type and Mid-Tread type. The following figures represent the two types of uniform quantization.

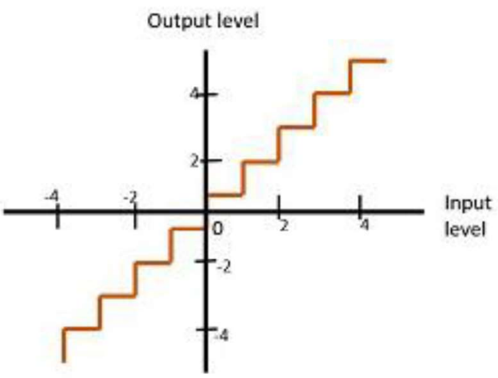


Fig 1 : Mid-Rise type Uniform Quantization

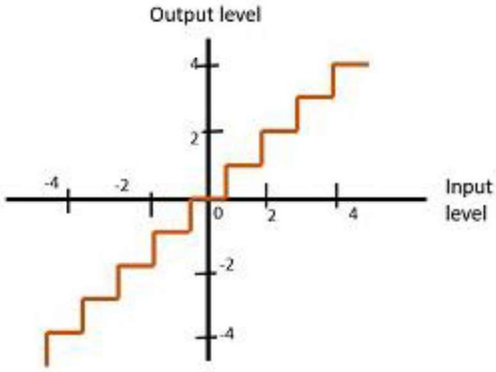


Fig 2 : Mid-Tread type Uniform Quantization

Figure 1 shows the mid-rise type and figure 2 shows the mid-tread type of uniform quantization.

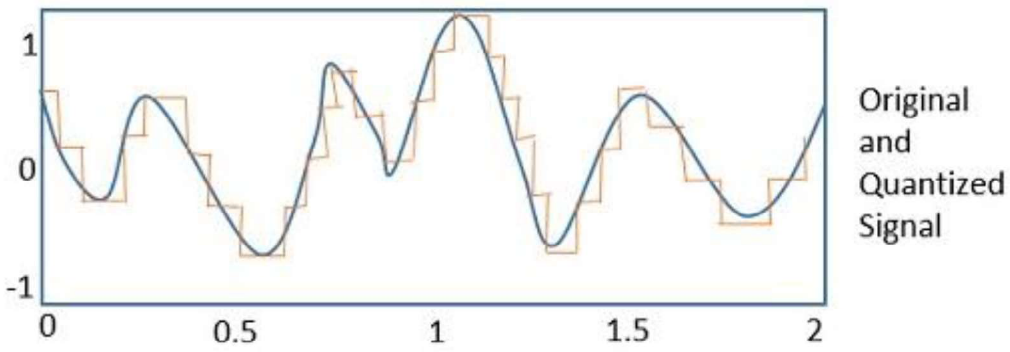
- The **Mid-Rise** type is so called because the origin lies in the middle of a raising part of the stair-case like graph. The quantization levels in this type are even in number.
- The **Mid-tread** type is so called because the origin lies in the middle of a tread of the stair-case like graph. The quantization levels in this type are odd in number.
- Both the mid-rise and mid-tread type of uniform quantizers are symmetric about the origin.

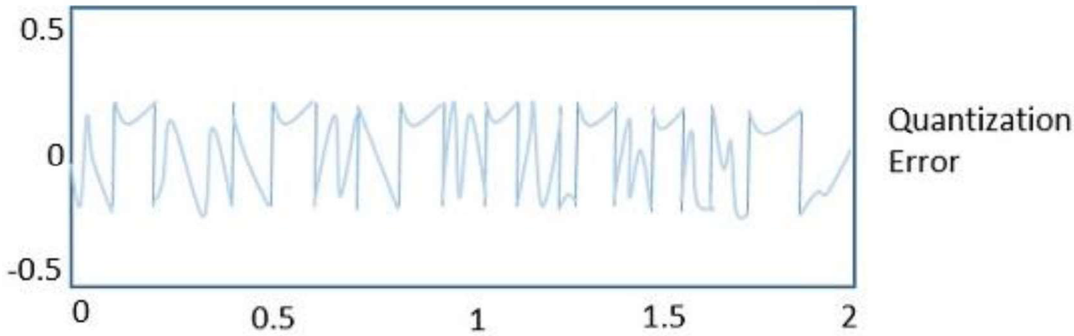
Quantization Error

For any system, during its functioning, there is always a difference in the values of its input and output. The processing of the system results in an error, which is the difference of those values.

The difference between an input value and its quantized value is called a **Quantization Error**. A **Quantizer** is a logarithmic function that performs Quantization (rounding off the value). An analog-to-digital converter (**ADC**) works as a quantizer.

The following figure illustrates an example for a quantization error, indicating the difference between the original signal and the quantized signal.





Quantization Noise

It is a type of quantization error, which usually occurs in analog audio signal, while quantizing it to digital. For example, in music, the signals keep changing continuously, where a regularity is not found in errors. Such errors create a wideband noise called as **Quantization Noise**.

Companding:- A more practical approach is to predistort the signal by an logarithmic compression characteristic and then put it to an uniform quantizer. This compressed and quantized signal is transmitted through the channel and can be undistorted at the receiver by the same algorithm. This process is known as companding

Pulse Amplitude Modulation

3.(a)Explain Natural PAM and Flat top PAM.

Or

(b).Briefly explain the generation and detection

In some PAM systems, the amplitude of each pulse is directly proportional to the instantaneous modulating-signal amplitude at the time the pulse occurs. In other PAM systems, the amplitude of each pulse is inversely proportional to the instantaneous modulating-signal amplitude at the time the pulse occurs

Generation of PAM

- Pulse amplitude modulation is the basic form of pulse modulation in which the signal is sampled at regular and each sample is made proportional to the amplitude of the modulating signal at the sampling instant.

- The Fig1 shows the generation of PAM signal from the sampler which has two inputs i.e. modulating signal and sampling signal or carrier pulse.

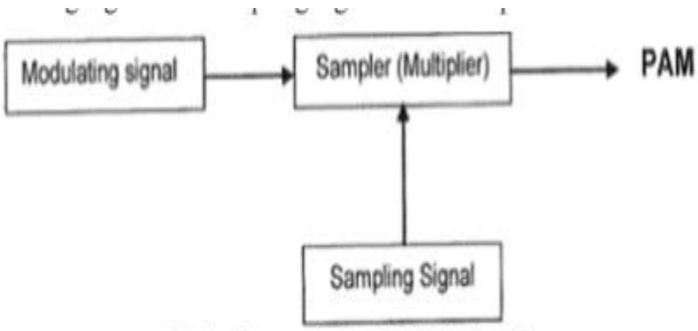


Fig1. Generation of PAM signal

Thus the amplitude of the signal is proportional to the modulating signal through which information is carried. This is Pulse amplitude modulation signal.

- Fig2 shows the spectrum of pulse amplitude modulated signal along with the message signal and the sampling signal which is the carrier train of pulses with the help of the waveform plotted in time domain.

- Pulse Modulation may be used to transmitting analog information, such as continuous speech signal or data

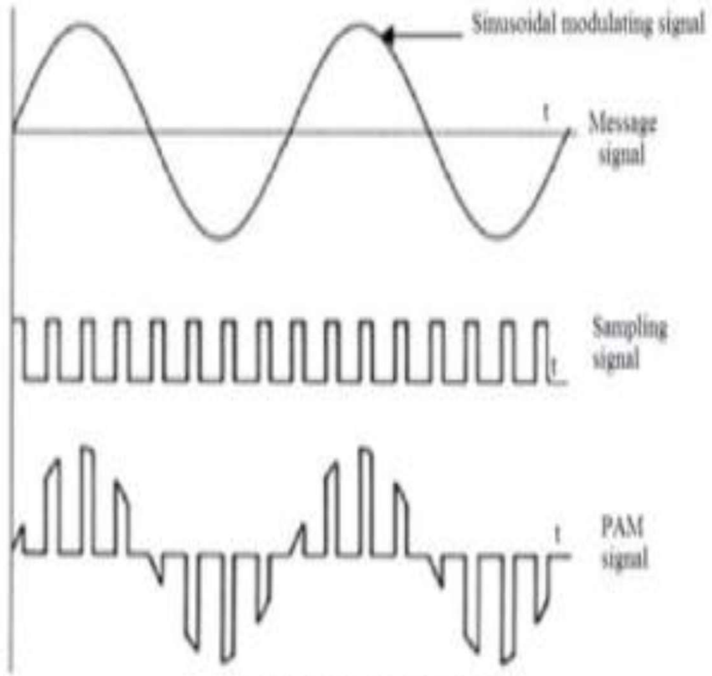


Fig2. Spectrum of PAM signal

Demodulation of PAM

For Demodulation of the Pulse Amplitude Modulated signal, PAM is fed to the low pass filter as shown in Fig3 below

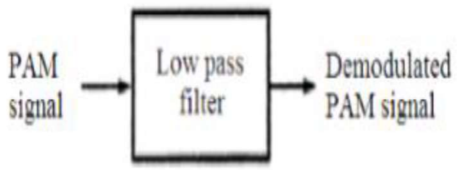


Fig3. PAM detector

• The low pass filter eliminates high frequency ripples and generates the demodulated signal which has its amplitude proportional to PAM signal at all time instant

. • This signal is then applied to an inverting amplifier to amplify its signal level to have the demodulated output with almost equal amplitude with the modulating signal.

• The Fig4 below shows the modulated and demodulated PAM signal.

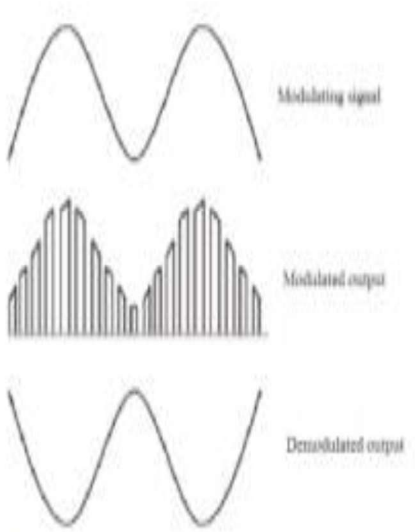


Fig4. Modulation and demodulation of PAM signal

Flat top PAM:

The PAM signal can be detected by simply passing it through a low pass filter. The sampling switch is closed for a short duration by a short pulse applied to the gate G of FET. During this period capacitor C is charged upto voltage is equal instantaneous value of incoming signal x(t). Now the sampling switch is opened and capacitor C holds the charge

The discharge switch is then closed by a pulse applied to gate G2 of other FET. Due to this the capacitor C is discharged to 0 volts. The discharge switch is then open and thus capacitor has no voltage. Hence the output of sample X hold circuit contains sequence of flat top samples.

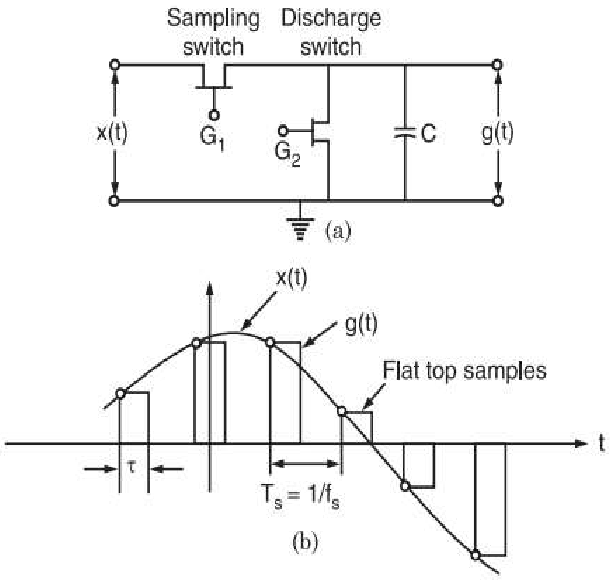


Fig.1 : (a) Sample and hold circuit generating flat top sampled PAM, (b) Waveforms of flat top sampled PAM

Line Coding

4.(a)Explain in detail about line coding.

Or

(b).With example,Explain in detail about the types of line coding.

A **line code** is the code used for data transmission of a digital signal over a transmission line. This process of coding is chosen so as to avoid overlap and distortion of signal such as inter-symbol interference.

Properties of Line Coding

Following are the properties of line coding

- As the coding is done to make more bits transmit on a single signal, the bandwidth used is much reduced.
- For a given bandwidth, the power is efficiently used.
- The probability of error is much reduced.
- Error detection is done and the bipolar too has a correction capability.
- Power density is much favorable.
- The timing content is adequate.

- Long strings of **1s** and **0s** is avoided to maintain transparency.

Types of Line Coding

There are 3 types of Line Coding

- Unipolar
- Polar
- Bi-polar

Unipolar Signaling

Unipolar signaling is also called as **On-Off Keying** or simply **OOK**.

The presence of pulse represents a **1** and the absence of pulse represents a **0**.

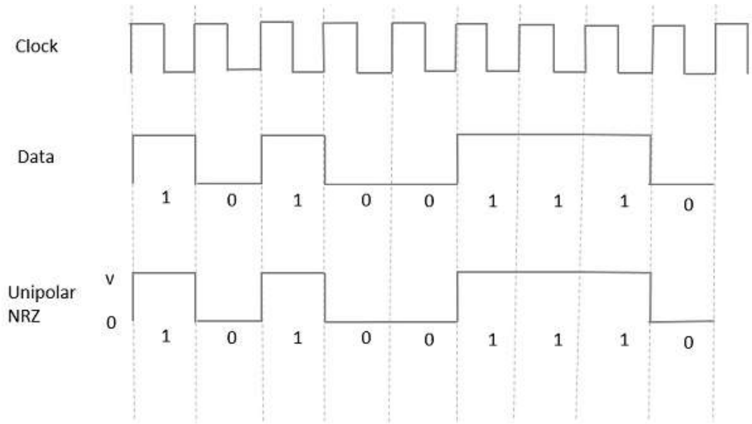
There are two variations in Unipolar signaling −

- Non Return to Zero (NRZ)
- Return to Zero (RZ)

Unipolar Non-Return to Zero (NRZ)

In this type of unipolar signaling, a High in data is represented by a positive pulse called as **Mark**, which has a duration T_0 equal to the symbol bit duration. A Low in data input has no pulse.

The following figure clearly depicts this.



Advantages

The advantages of Unipolar NRZ are

- It is simple.
- A lesser bandwidth is required.

Disadvantages

The disadvantages of Unipolar NRZ are

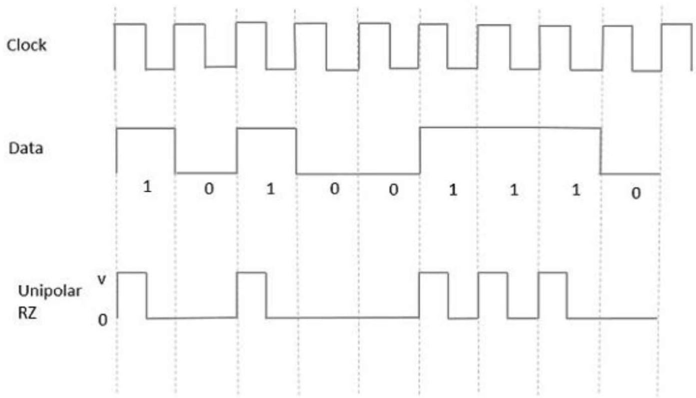
- No error correction done.

- Presence of low frequency components may cause the signal droop.
- No clock is present.
- Loss of synchronization is likely to occur (especially for long strings of **1s** and **0s**).

Unipolar Return to Zero (RZ)

In this type of unipolar signaling, a High in data, though represented by a **Mark pulse**, its duration T_0 is less than the symbol bit duration. Half of the bit duration remains high but it immediately returns to zero and shows the absence of pulse during the remaining half of the bit duration.

It is clearly understood with the help of the following figure.



Advantages

The advantages of Unipolar RZ are

- It is simple.
- The spectral line present at the symbol rate can be used as a clock.

Disadvantages

The disadvantages of Unipolar RZ are

- No error correction.
- Occupies twice the bandwidth as unipolar NRZ.
- The signal droop is caused at the places where signal is non-zero at 0 Hz.

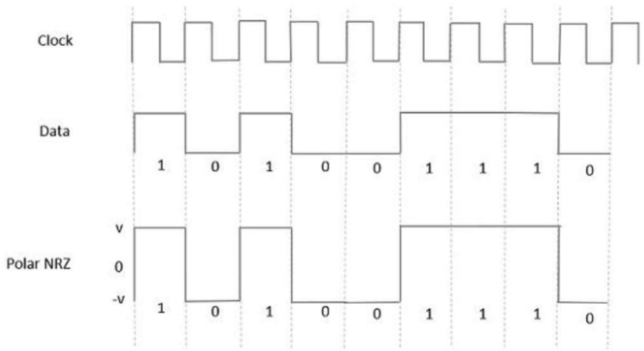
Polar Signaling

There are two methods of Polar Signaling. They are

- Polar NRZ
- Polar RZ

Polar NRZ

In this type of Polar signaling, a High in data is represented by a positive pulse, while a Low in data is represented by a negative pulse. The following figure depicts this well.



Advantages

The advantages of Polar NRZ are

- It is simple.
- No low-frequency components are present.

Disadvantages

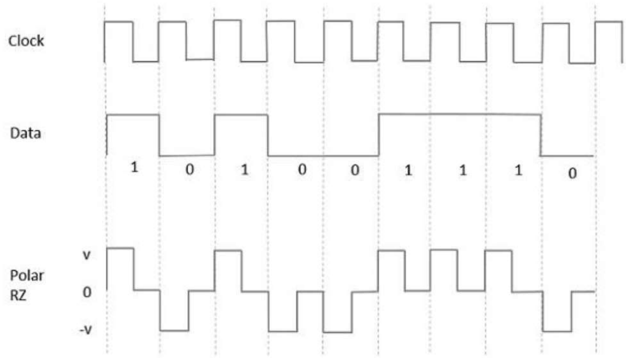
The disadvantages of Polar NRZ are

- No error correction.
- No clock is present.
- The signal droop is caused at the places where the signal is non-zero at **0 Hz**.

Polar RZ

In this type of Polar signaling, a High in data, though represented by a **Mark pulse**, its duration $\mathbf{T_0}$ is less than the symbol bit duration. Half of the bit duration remains high but it immediately returns to zero and shows the absence of pulse during the remaining half of the bit duration.

However, for a Low input, a negative pulse represents the data, and the zero level remains same for the other half of the bit duration. The following figure depicts this clearly.



Advantages

The advantages of Polar RZ are

- It is simple.
- No low-frequency components are present.

Disadvantages

The disadvantages of Polar RZ are

- No error correction.
- No clock is present.
- Occupies twice the bandwidth of Polar NRZ.
- The signal droop is caused at places where the signal is non-zero at **0 Hz**.

Bipolar Signaling

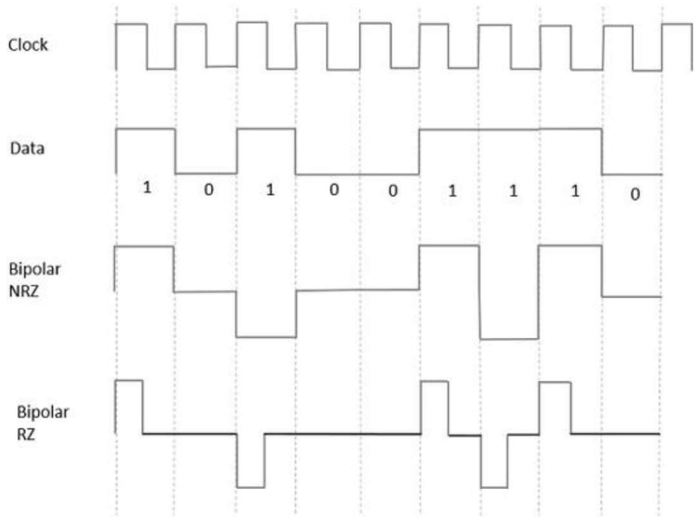
This is an encoding technique which has three voltage levels namely **+, -** and **0**. Such a signal is called as **duo-binary signal**.

An example of this type is **Alternate Mark Inversion (AMI)**. For a **1**, the voltage level gets a transition from + to – or from – to +, having alternate **1**sto be of equal polarity. A **0** will have a zero voltage level.

Even in this method, we have two types.

- Bipolar NRZ
- Bipolar RZ

From the models so far discussed, we have learnt the difference between NRZ and RZ. It just goes in the same way here too. The following figure clearly depicts this.



The above figure has both the Bipolar NRZ and RZ waveforms. The pulse duration and symbol bit duration are equal in NRZ type, while the pulse duration is half of the symbol bit duration in RZ type.

Advantages

Following are the advantages

- It is simple.
- No low-frequency components are present.
- Occupies low bandwidth than unipolar and polar NRZ schemes.
- This technique is suitable for transmission over AC coupled lines, as signal drooping doesn't occur here.
- A single error detection capability is present in this.

Disadvantages

Following are the disadvantages

- No clock is present.
- Long strings of data causes loss of synchronization.

PULSE CODE MODULATION

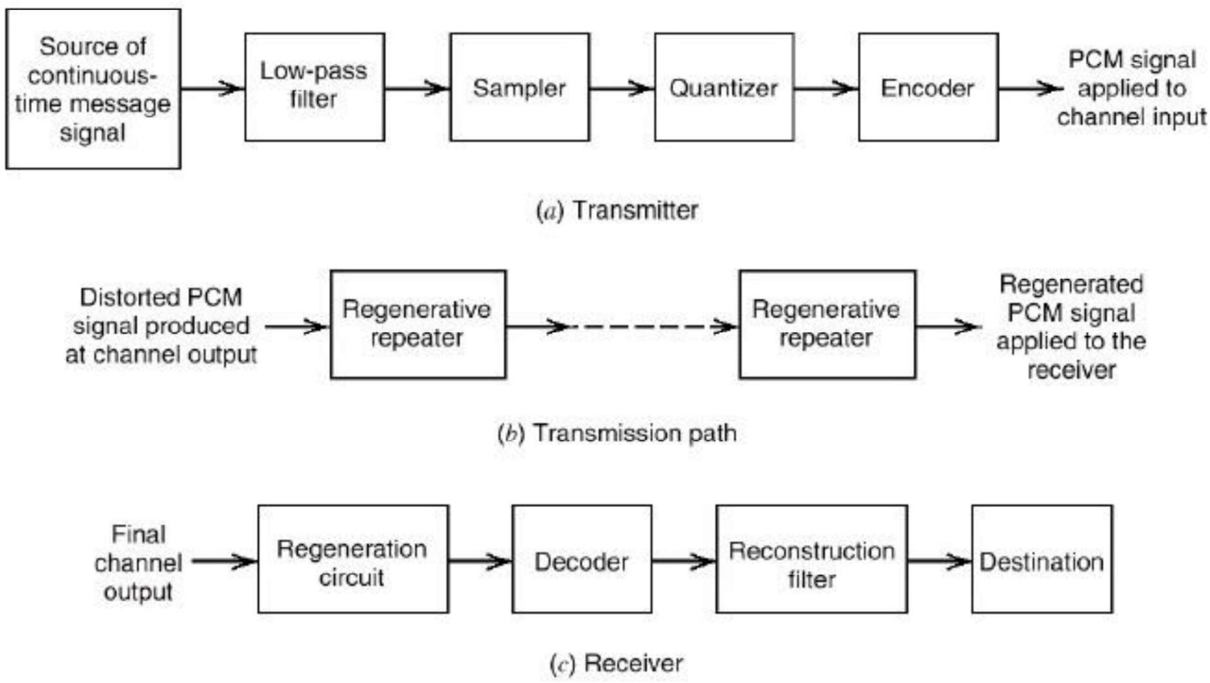
5.(a)With neat block diagram,Explain Pulse Code Modulation.

Or

(b).Briefly explain about PCM.

PCM is the only digitally encoded modulation technique used commonly for digital transmission. In PCM, the pulses are of fixed length and fixed amplitude. Presence of pulse is represented by binary '1' and absence of pulse is represented by binary '0'.

Fig.3.2 shows the simplified block diagram of a single channel, simplex (one way) PCM system



The Band pass filter limits the frequency of the analog input signal to the standard voice band frequency range of 300Hz to 3000 Hz.The sample and hold circuit periodically samples the analog input signal and converts those samples to a multi level PAM signal. The analog to digital converter (ADC) converts the PAM samples to parallel PCM codes.

The parallel to serial converter converts the parallel PCM codes into a serial binary data and placed on the transmission line as serial digital pulses. The serial to parallel converter in the receiver converts serial pulses received from the transmission line to parallel PCM codes. The (DAC) digital to analog converter converts the parallel PCM codes into PAM signals.

The Hold circuit is basically a Low pass filter that converts the PAM signals back to its original analog form. An integrated circuit that performs PCM encoding and decoding functions is called a codec (coder / decoder).

PCM SAMPLING

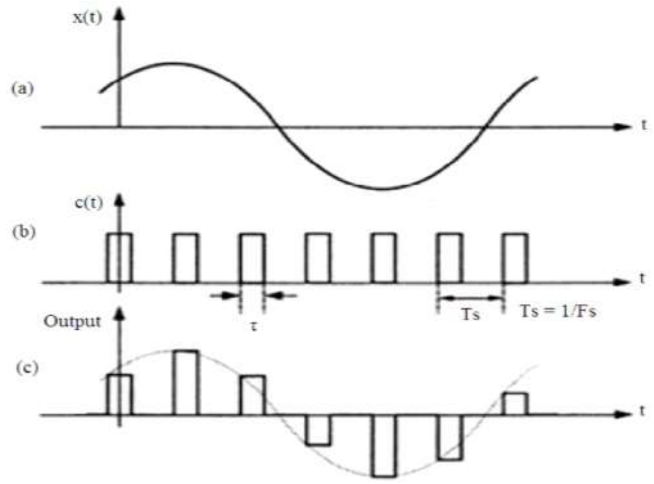
The function of sampling circuit in a PCM transmitter is to periodically sample analog signal and convert those samples to a constant amplitude pulses.

The sampling and hold circuit is used to sample analog signals.

There are two types of sampling natural sampling and flat top sampling.

In natural sampling, after sampling the top of the sample pulse has its shape similar to the shape of analog signals. So ADC find it difficult to convert the analog sample to PCM code.

So we go for flat top sampling.



(a) Modulating signal (b) sampling signal and (c) Flat top sampling spectrum

In flat top sampling, the input voltage is sampled with a narrow pulse and the sampled input voltage is held constant till the next sample is taken. So sample and hold circuit prefers flat top sampling.

But during flat top sampling the amplitude of sampled input may change which introduces an error known as aperture error. The magnitude of error depends on analog signal voltage and width of the sample pulse.

DIFFERENTIAL PULSE CODE MODULATION

6.(a)Explain in detail about DPCM transmitter and receiver.

Or

(b).Briefly explain about Differential Pulse Code Modulation with suitable diagrams

Redundant information in PCM:-

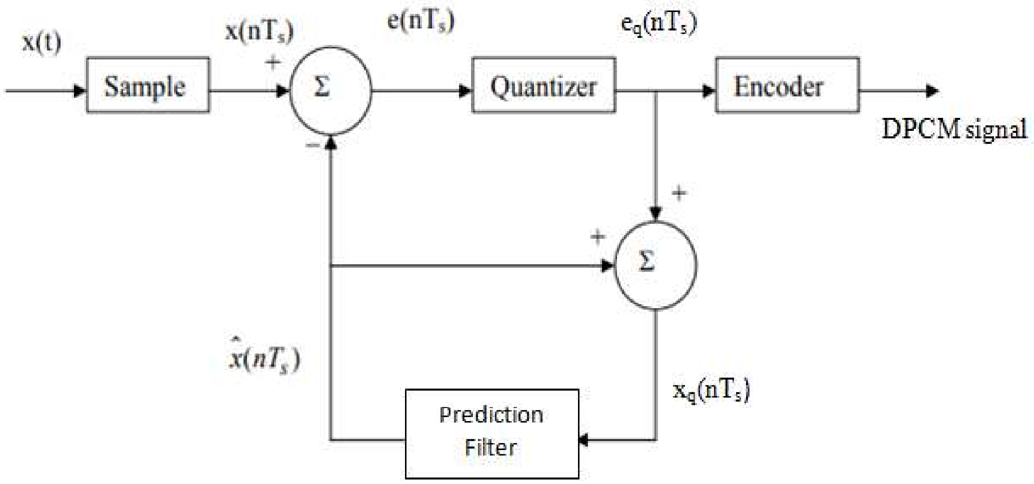
The samples of a signal are highly corrected with each other. This is because any signal does not change fast. That is its value from present sample to next sample does not differ with large amount. The adjacent samples of the signal carry the same information with little difference. When these samples are encoded by standard PCM system the resulting encoded signal contains redundant information.

Principle of DPCM:- If this redundancy is reduced then overall bit rate will decrease and number of bits required to transmit one sample will also reduced. This type of digital pulse modulation scheme is called differential pulse code modulation.

DPCM Transmitter:- The differential pulse code modulation works on the principle

of prediction. The value of the present sample is predicted from the past samples. The prediction may not be exact but it is very close to the actual sample value.

Fig shows the transmitter of DPCM system. The sampled signal is denoted by $x(nT_S)$ and the predicted signal is denoted by $x^{\wedge}(nT_S)$



The comparator finds out the difference between the actual sample value and the predicted sample value .This is called error and is denoted by $e(nT_S)$. It can be defined as

$$e(nT_S) = x(nT_S) - x^{\wedge}(nT_S)$$

Thus the error is the difference between unquatized input sample $x(nT_S)$ and prediction of it $x^{\wedge}(nT_S)$. The predicted value is produced by using a prediction filter. The quantizer output signal $e_q(nTS)$ and the previous prediction is added an given as input to prediction filter. This signal is called($x_q nT_S$).The quantiser output is written as

$$e_q(nT_S)= e(nT_S) + q(nT_S)$$

Here $q(nT_S)$ is the quantization error.The prediction filter $x_q(nT_S)$ is obtained by sum $x^{\wedge}(nT_S)$ and the quantiser output (ie) $x_q(nT_S)= x^{\wedge}(nT_S) + e_q(nTS)$

Putting the value of $e_q(nT_S)$ from equation (2)

$$xq(nT_S)= x^{\wedge}(nT_S) + e_q(nTS) + q(nT_S)$$

$$e(nT_S) = x(nT_S) - x^{\wedge}(nT_S)$$

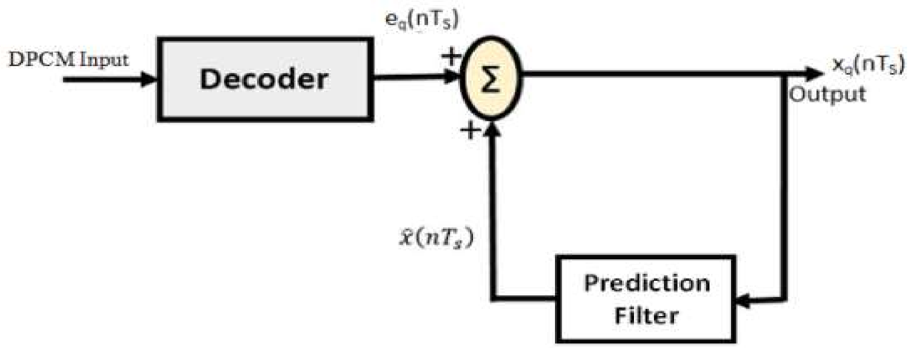
$$e(nT_S) + x^{\wedge}(nT_S) = x(nT_S)$$

putting the value of $e(nT_S) + x^{\wedge}(nT_S)$ in $x_q(nT_S)$

$$x_q(nT_S)= x(nT_S) + q(nT_S)$$

Thus the quantized version of the signal $x_q(nT_S)$ is the sum of the original sample value and the quantization error $q(nT_S)$.

DPCM Receiver:-



The decoder first reconstructs the quantized error from incoming binary signal. The prediction filter output and quantized error signals are summed up to give the quantized version of the original signal.

SNR Improvement in DPCM:-

Consider a signal m(t) of peak amplitude m_p applied to both the systems. The signal power in both the system are same. The quantiser in PCM quantises the signal m(t) whereas the quantiser in DPCM quantises the difference signal d(t).

Let the peak amplitude of the difference signal d(t) be d_p. If we use same number of levels L in both the cases the step size in DPCM is reduced by m_P/d_p. So the quantization noise reduces by a factor of $(m_P/d_p)^2$. So SNR increases.The processing gain is given by

$$G_P=\left(\frac{m_p}{d_p}\right)^2$$

The crest factor of the encoded signal are same for both PCM and DPCM. So the processing gain is

$$G_P=\frac{m^2}{d^2}$$

The SNR value for DPCM can be written as

$$SNR_{DPCM} = G_P\, SNR_{PCM}$$

$$SNR_{DPCM} = G_P(dB)+(4.77+6\lambda)dB$$

Predictor Gain

Speech G_P= 5.6dB (5-10 dB)

TV video G_P= 12dB

ADAPTIVE DIFFERENTIAL PULSE CODE MODULATION

7.(a).Explain in detail about ADPCM with necessary diagram.

Or

(b).Explain adaptive quantization and adaptive prediction in ADPCM

The design of coder involves two steps

- Removing redundancies from speech signal as far as possible
- Assigning the available bits to code the non redundant parts of the speech signals in a perceptually efficient manner.

A digital coding scheme that uses both the adaptive quantization and adaptive prediction is called adaptive differential pulse code modulation. The number of 8 bits per sample required in PCM is reduced to 4.Adaptive quantizer operates with a time varying step size $\Delta(nT_S)$ where T_S is the sampling period. The step size $\Delta(nT_S)$ is varied so as to much the variance σ_m^2 at the input sample m(n)

$$\Delta(n)=\emptyset\hat{\sigma}_m(n)$$

where, $\hat{\sigma}_m(n)=\sqrt{\sigma_m^2}$

$\hat{\sigma}_m(n)$ ➔ estimate or approximate value of standard deviation $\sigma(n)$

σ_m ⟶ varience

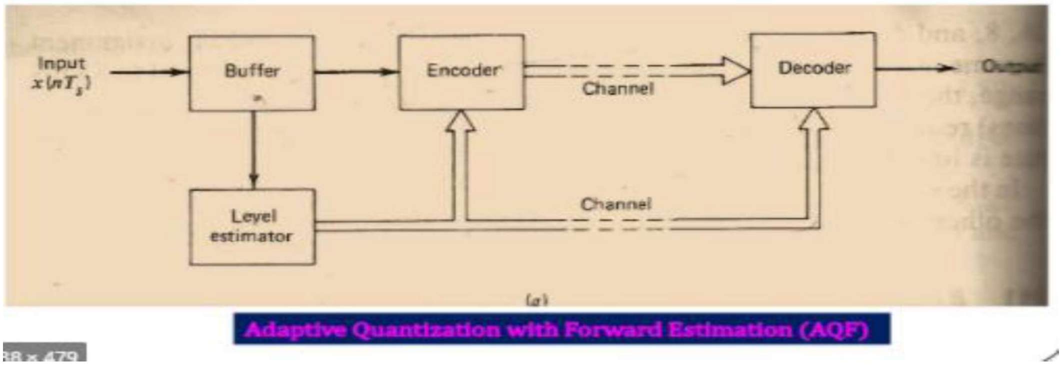
There are two ways to implement the above equation

i)**Adaptive quantization with forward estimation**:-Unquantised samples of input signals are used to derive forward estimates of $\sigma_m(n)$

Disadvantages:-

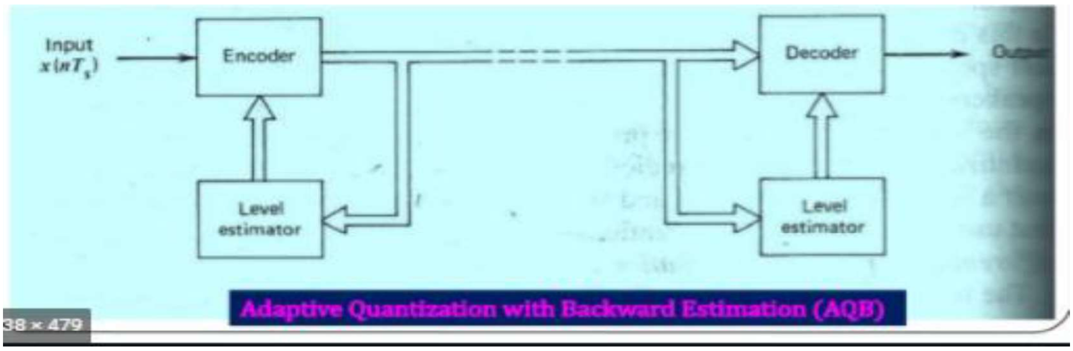
i) Processing delay of 16ms for speech.

ii) Complexity due to the presence of level estimation.



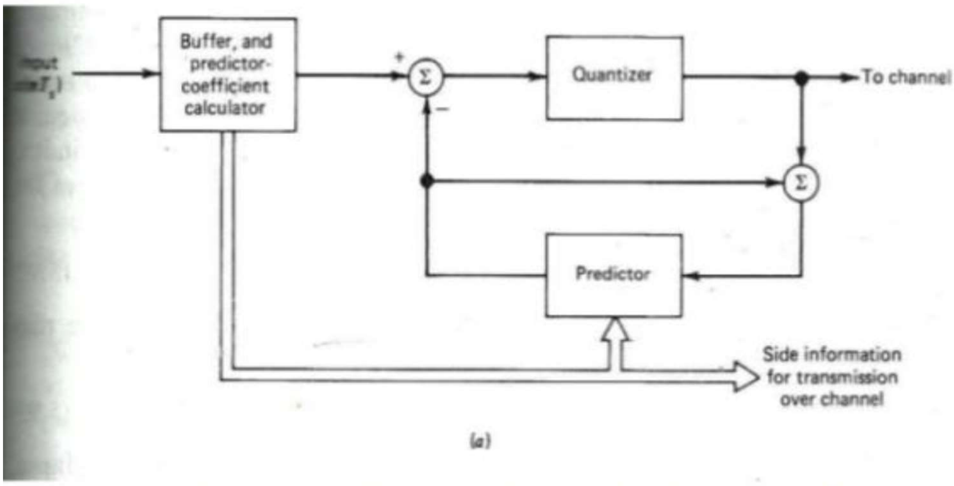
ii)**Adaptive quantization with backward estimation**:-

Samples of quantizer output are used to derive backward estimates of $\sigma_m(n)$.It is a non linear feedback system. The system is stable because the quantizer input, backward estimate and the corresponding step size $\Delta(nT_s)$ are bonded.



The use of adaptive prediction in ADPCM is justified because speech signals are inherently non stationary. There are two schemes for performing adaptive prediction.

1)**Adaptive Prediction with forward estimation**:The unquatised samples of the input signals are used to derive estimates of the predictor coefficients.



Adaptive prediction with forward estimation (APF)

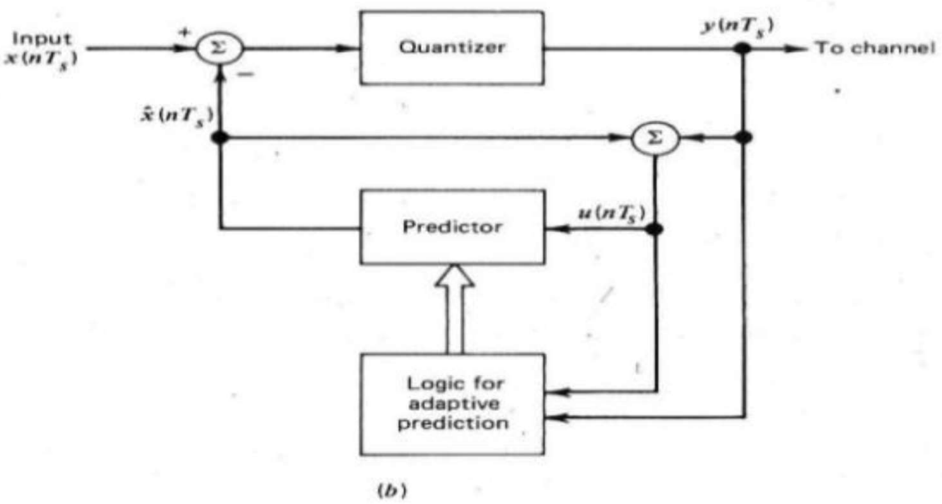
In APF scheme, N unquantised samples of the input speech are first buffered and then released after computation of M predictor coefficients that are optimized for the buffered segment of input samples. The choice of M involves a compromise between an adequate predictor gain and an acceptable amount of side information. The choice of buffer length N involves a compromise between the rate at which statistics of input speech signal change and the rate at which the information on predictor coefficient must be updated and transmitted to the receiver.

Disadvantages:-

- Side information, buffering and delay

These disadvantages are eliminated using APB scheme.

1) **Adaptive Prediction with backward estimation**: The samples of the quantizer output and the prediction error are used to derive estimates of predictor coefficient.APB can be used updated frequently for obvious reasons.



Adaptive prediction with backward estimation (APB)

DELTA MODULATION

8.(a)With neat diagram ,Explain in detail about DM transmitter and receiver.

Or

(b)Briefly explain about Delta modulation and its disadvantages.

Delta modulation transmits only one bit per sample. The present sample values compared with the previous sample value. The input signal x(t) is approximated to step signal by the delta modulator.The difference between the input signal x(t) and the staircase approximated signal confined two levels. (ie) +δ and – δIf the difference is positive then the signal is increaser by one step (ie) δ and if the difference is negative then approximated signal is decreased by δ.When the step size is reduced '0' is transmitted and if the step size is increased '1' is transmitted

The error between the sampled value of x(t) and approximated sample is given as

$$e(nT_S) = x(nT_S) - x^{\wedge}(nT_S)$$

Where $e(nT_S)$ = Error at present sample

$x(nT_S)$= Sampled signal of x(t)

$x^{\wedge}(nT_S)$ = Last sample approximation of the staircase waveform

Assume u(nT_S) as the present sample approximation of staircase output.

$$U[(n-1)T_S)\ ] = x^{\wedge}(nT_S)$$

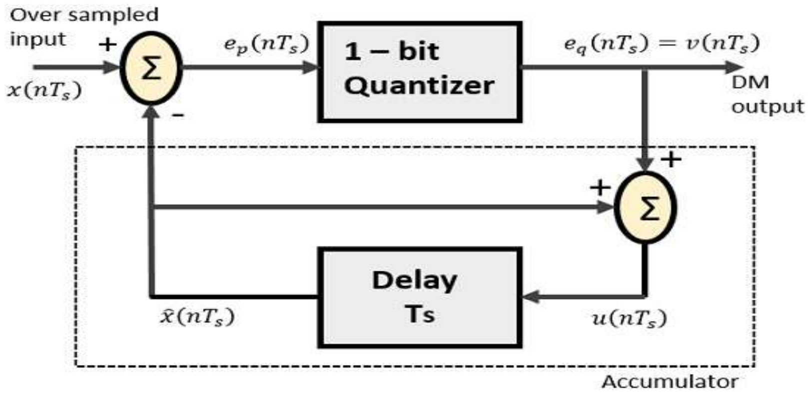
We have b(nT_S) $= +\delta$ if $x(nT_S) \geq x^{\wedge}(nT_S)$

$= -\delta$ if $x(nT_S) \geq x^{\wedge}(nT_S)$

If b(nTs)=+ δ binary 1 is transmitted

If b(nTs)=- δ binary 0 is transmitted

DM Transmitter



The summer in the accumulator adds quantizer output $(\pm\delta)$ with the previous sample approximation,

$$u(nTs) = u(nTs - Ts) + b(nTs)$$

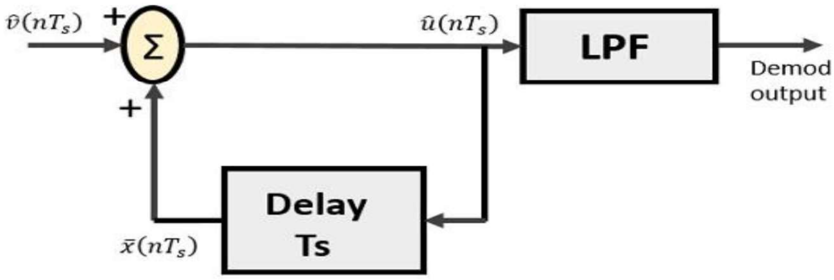
The present sample x(nT_s) and the previous sample are subtracted to get e(nT_s)

$$e(nTs) = x(nTs) - \hat{x}(nTs)$$

Depending on the sing of e(nT_s) one bit quantiser produces an output $\pm\delta$

$$\text{If } e(nTs) = \begin{cases} -1 \text{ then } b(nTs) = -\delta \\ +1 \text{ then } b(nTs) = +\delta \end{cases}$$

DM Receiver : All the receiver the accumulator and LPF are used



From the above diagram, we have the notations as

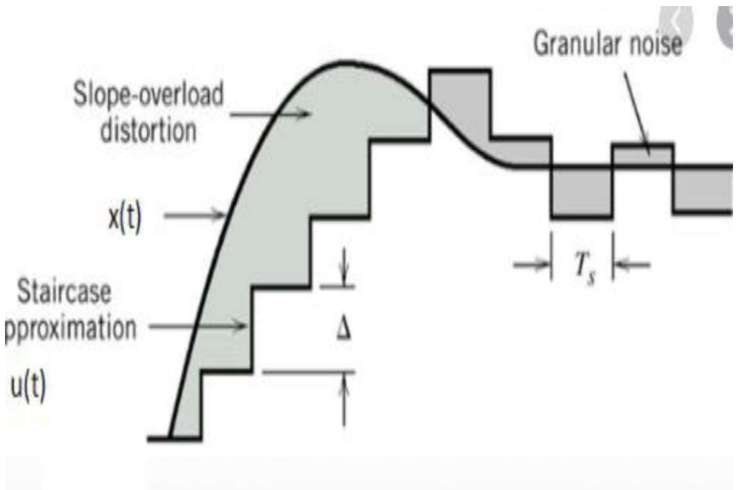
- v^(nTs)v^(nTs) is the input sample
- u^(nTs)u^(nTs) is the summer output
- x¯(nTs)x¯(nTs) is the delayed output

A binary sequence will be given as an input to the demodulator. The stair-case approximated output is given to the LPF.

Low pass filter is used for many reasons, but the prominent reason is noise elimination for out-of-band signals. The step-size error that may occur at the transmitter is called **granular noise**, which is eliminated here. If there is no noise present, then the modulator output equals the demodulator input.

Advantages :i)DM transmits only one but for one sample. Thus the signaling rate and transmission channel bandwidth is small .ii)The transmitter and receiver implementation is very much simple for DM. There is no A/D converter in DM.

Disadvantages :



Slope overload distortion :The distortion arises because of the large dynamic range of the input signal.The rate of rise of input signal x(t) is so high that the staircase signal cannot approximate it the step size δ become too small for staircase signal u(t) to follow the steep segment of x(t).There is a large error between the staircase approximated signal and the original signal. This is called *slope overload distortion*.

To reduce this error the step size should be increase when slop of x(t) is high.Since the step size of delta Modulator remains fixed its maximum or minimum slopes occur along straight lines. Therefore this modulator is called *linear delta Modulator.*

Granular Noise :

If the amplitude of the analog signal is almost constant we need small step size in order to reconstruct the original along signal. But here the step size is large and constant. Due to large step size it is difficult to reconstruct the original message signal. This error is referred to as *Granular noise*. To reduce granular noise the step size has to be decreased. So to remove slope overloading the step size has to be increased and to remove granular noise the step size has to be decreased.

Adaptive Delta Modulation

9.(a).With neat diagram,explain ADM transmitter and Receiver.

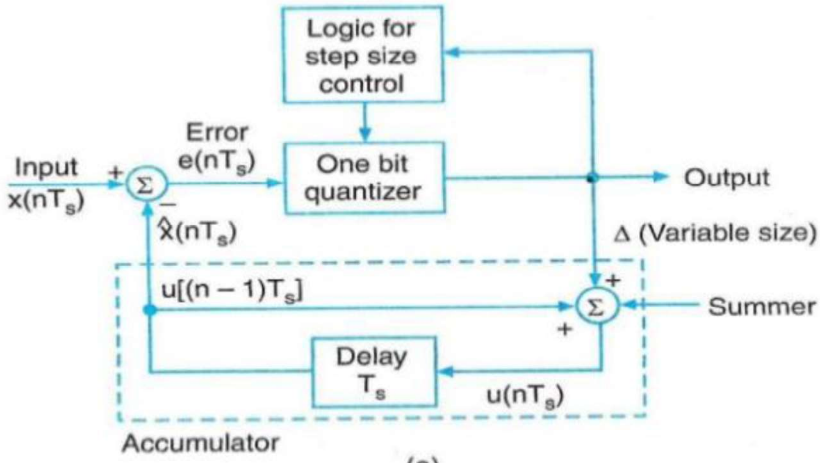
Or

(b).Explain the principle of Adaptive Delta Modulation.

Operating Principle :

To overcome the quantization error due to slope overload and granular noise, the step size (δ) is made adaptive to variations in the input signal x(t).Particularly in steep segment of the signal x(t) the step size is increased when the input is varying slowly, the step size is reduced. Then the method is called Adaptive delta modulation. The adaptive delta modulation can take continues changes in step size or discrete changes in step size.

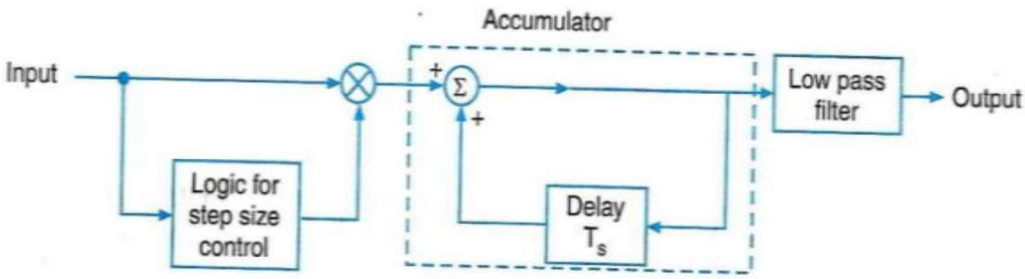
Transmitter and Receiver :The step size increases or decreases according to certain rule depending on one bit quantizer output. eg If one bit quantizer output is high(1), then step size may be doubled for next sample. If one bit quantizer output is low, then step size may be reduced one step.



(a)

In the receiver of adaptive delta modulator the first part generates the step size from each incoming bit. Exactly same process is followed as that in transmitter.

- ❖ The previous input and the present input decides the step size. It is then given to the accumulator which builds up staircase waveform. The low pass filter smoothens the staircase waveform to reconstruct the smooth signal.



Advantages of Adaptive delta Modulation :

ADM has certain advantages over delta modulation.

- ❖ Signal to noise ratio is better than delta modulation because of reduction in slope overload distortion and granular noise.
- ❖ Dynamic range of ADM is wide because of variable step size.
- ❖ Utilization of bandwidth is better than delta modulation.

Comparison between PCM Adaptive Delta Modulation and Differential Pulse Code Modulation

10.(a)Compare various Pulse modulation

Or

(b).Compare the different pulse modulation techniques.

Sl. No.	*Parameter*	*PCM*	*Delta Modulation (DM)*	*Adaptive Delta Modulation (ADM)*	*Differential Pulse Code Modulation (DPCM)*
1.	Number of bits	It can use 4, 8 or 16 bits per sample	It uses only one bit for one sample	Only one bit is used to encode one sample	Bitscan be more than one but are less than PCM
2.	Levels, step size	The number of levels depend on number of bits. Level size is fixed	Step size is fixed and cannot be varied	According to the signal variation, step size varies (Adaption)	Fixed number of levels are used
3.	Quantization error and distortion	Quantization error depends on number of levels used	Slope overload distortion and granular noise is present	Quantization error is present but other errors are absent	Slope overload distortion and quantization noise is present
4.	Bandwidth of transmission channel	Highest bandwidth is required since number of	Lowest bandwidth is required	Lowest bandwidth is required	Bandwidth required is lower than PCM

		bits are high			
5.	Feedback	There is no feed back in transmitter or receiver	Feedback exists in transmitter	Feedback exists	Feedback exists
6.	Complexity of notation	System is complex	Simple	Simple	Simple
7.	Signal of noise ratio	Good	Poor	Better than DM	Fair
8.	Area of applications	Audio and Video Telephony	Speech and images	Speech and images	Speech and video

Next part shows the comparison for voice encoding

Sl.No	*Parameter*	*PCM*	*DM*	*ADM*	*DPCM*
9.	Sampling rate kHz	8	64-128	48-64	8
10.	Bits/sample	7-8	1	1	4-6
11.	Bit rate	56-64	64-128	46-64	32-45

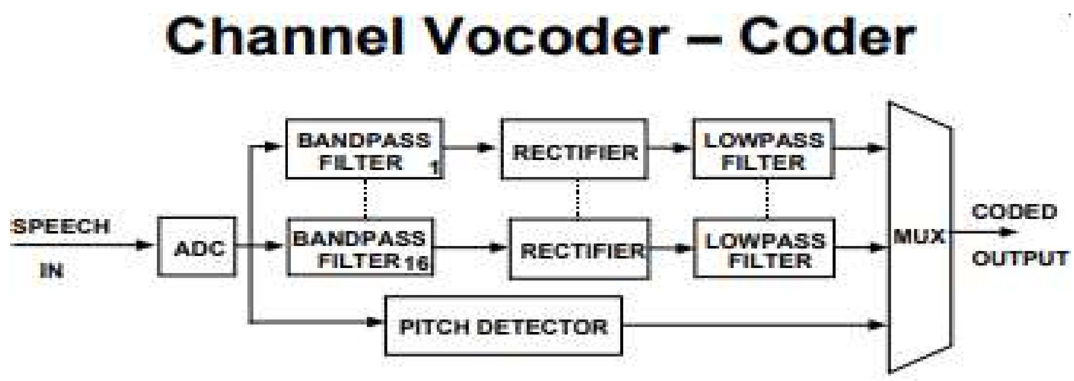
Channel Vocoder

11.(a).Explain in detail about Channel Vocoder

Or

(b).With neat block diagram,explain Channel Vocoder.

The channel vocoder splits speech into increasing, non-overlapping frequency sub-bands. The whole range covers the frequencies that humans can hear. Incoming speech is periodically sampled, typically every 20 ms. The channel vocoder uses a multiplexer to send two different categories of information on the same channel.



The best way to understand how multiplexers work is to consider them as sampling circuits. To multiplex two signals, we take a sample from the first signal and then the second. Samples are taken sequentially. The same cycle continues periodically. The sampling rate for all signals must be above the Nyquist limit. These samples may then be transmitted as a single signal. A de-multiplexer circuit on the receiving end can easily separate each signal since it knows the timing of the samples. This multiplexing scheme is called time division multiplexing, (TDM). The multiplexer of the channel vocoder operates in a similar fashion.

It enables two different groups of signals to be transmitted on the same wire without significant crossover. Once the band pass filters divide the speech into smaller frequency groups, each sub-band is rectified and low-pass filtered to determine the spectral envelope of the speech. They are then digitized and multiplexed for transmission. There are usually 16 sub-bands covering the whole of the audible frequency range. The speech is also analyzed for the pitch of the sample. For voiced sounds, the estimated frequency is multiplexed and transmitted.

For unvoiced sounds, since they do not have a pitch, only an indication of their existence is coded and transmitted. A pitch detector tries to distinguish between the voiced and unvoiced segments. It is quite easy to identify voiced segments with clear periodicity and unvoiced segments, which are non-periodic. However, it is very difficult to assess segments that fall between these two extremes. There are a number of voice detection algorithms developed, but none performs well in all applications. 5-9 In summary, the channel vocoder analyzes the incoming speech signal by three major categories:

• Speech spectral signal envelope

• Pitch

• Amplitude Extracted information is then multiplexed and transmitted.

Such analysis does not look to preserve the originality of the speech, but tries to compress it without losing the ability to understand it.

Frequency-Division Multiplexing

12.(a).Explain FDM

Or

(b).With block diagram,explain frequency division multiplexing.

Another important signal processing operation is *multiplexing,* whereby a number of independent signals can be combined into a composite signal suitable for transmission over a common channel. Voice frequencies transmitted over telephone systems, for example, range from 300 to 3100 Hz. To transmit a number of these signals over the same channel, the signals must be kept apart so that they do not interfere with each other, and thus they can be separated at the receiving end. This is accomplished by separating the signals either in frequency or in time. The technique of separating the signals in frequency is referred to as *frequency-division multiplexing* (FDM), whereas the technique of separating the signals in time is called *time-division multiplexing* (TDM).

A block diagram of an FDM system is shown in Figure The incoming message signals are assumed to be of the low-pass type, but their spectra do not necessarily have nonzero values all the way down to zero frequency. Following each signal input, we have shown a low-pass filter, which is designed to remove high-frequency components that do not contribute significantly to signal representation but are capable of disturbing other message signals that share the common channel. These low-pass filters may be omitted only if the input signals are sufficiently band limited initially. The filtered signals are applied

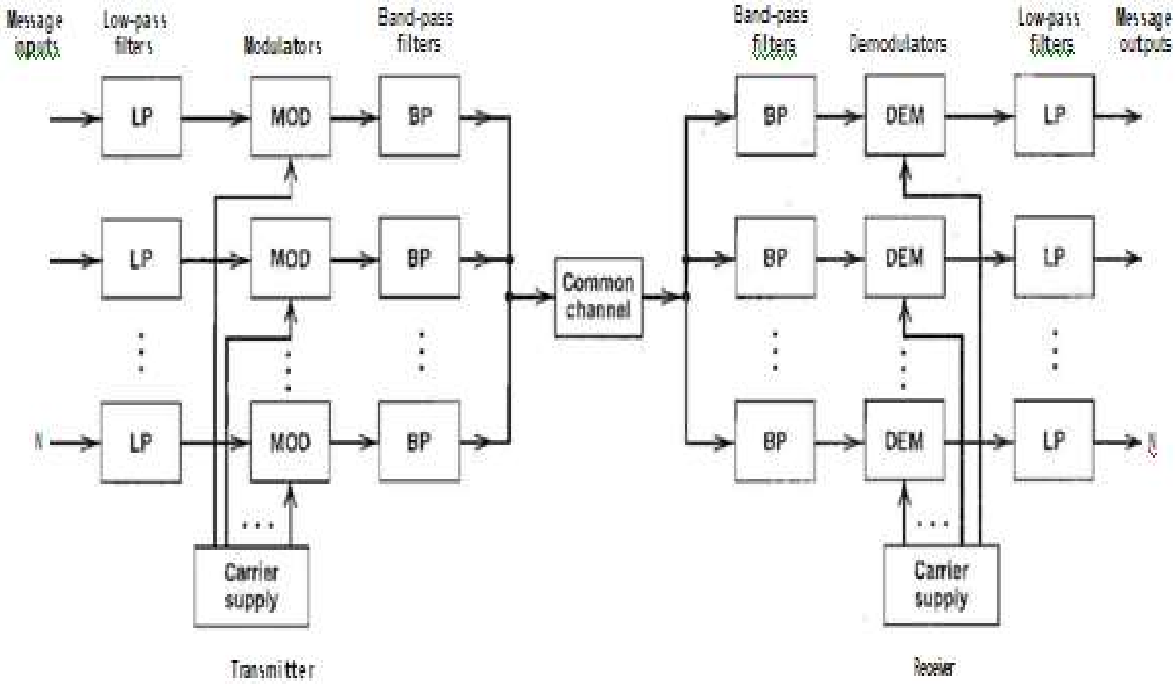


FIGURE 2.18 Block diagram of FDM system

to modulators that shift the frequency ranges of the signals so as to occupy mutually exclusive frequency intervals. The necessary carrier frequencies needed to perform these frequency translations are obtained from a carrier supply. For the modulation, we may use any one of the methods described in previous sections of this chapter. However, the most widely used method of modulation in frequency-division multiplexing is single side band modulation, which, in the case of voice signals, requires a bandwidth that is approximately equal to that of the original voice signal. In practice, each voice input is usually assigned a bandwidth of 4 kHz. The band-pass filters following the modulators are used to restrict the band of each modulated wave to its prescribed range.

The resulting band pass filter outputs are next combined in parallel to form the input to the common channel. At the receiving terminal, a bank of band-pass filters, with their inputs connected in parallel, is used to separate the message signals on a frequency-occupancy basis. Finally, the original message signals are recovered by individual demodulators. Note that the FDM system shown in Figure 2.18 operates in only one direction. To provide for two-way transmission, as in telephony, for example, we have to completely duplicate the multiplexing facilities, with the components connected in reverse order and with the signal waves proceeding from right to left.

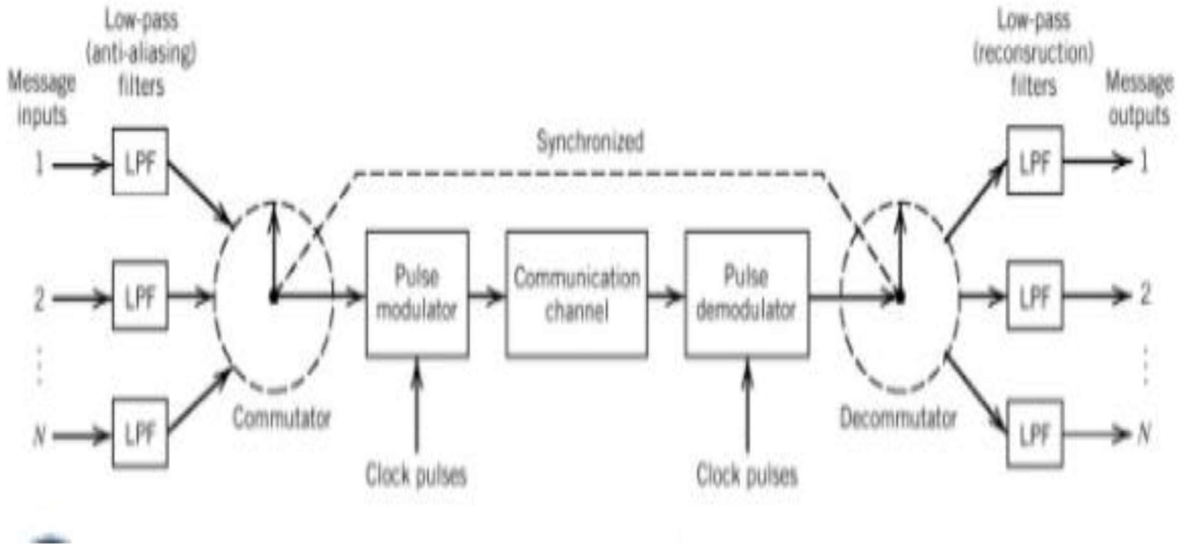
Time-Division Multiplexing (TDM)

13.(a).Explain TDM.

Or

(b).With neat block diagram,explain time division multiplexing.

In frequency division multiplexing, all signals operate at the same time with different frequencies, but in Time-division multiplexing all signals operate with same frequency at different times. This is a base band transmission system, where an electronic commutator sequentially samples all data source and combines them to form a composite base band signal, which travels through the media and is being demultiplexed into appropriate independent message signals by the corresponding commutator at the receiving end. The incoming data from each source are briefly buffered. Each buffer is typically one bit or one character in length. The buffers are scanned sequentially to form a composite data stream. The scan operation is sufficiently rapid so that each buffer is emptied before more data can arrive. Composite data rate must be at least equal to the sum of the individual data rates. The composite signal can be transmitted directly or through a modem. The multiplexing operation is shown in Fig.



Along with the sampled pulses, one synchronizing pulse is sent in each cycle. These data pulses along with the control information form a *frame*. Each of these frames contain a cycle of time slots and in each frame, one or more slots are dedicated to each data source. The maximum bandwidth (data rate) of a TDM system should be at least equal to the same data rate of the sources.
Synchronous TDM is called synchronous mainly because each time slot is preassigned to a fixed source. The time slots are transmitted irrespective of whether the sources have any data to send or not. Hence, for the sake of simplicity of implementation, channel capacity is wasted. Although fixed assignment is used TDM, devices can handle sources of different data rates. This is done by assigning fewer slots per cycle to the slower input devices than the faster devices.

Short Answers

1. What is meant by aliasing effect?

Aliasing effect takes place when sampling frequency is less than Nyquist rate.Under such condition, the spectrum of the sampled signal overlaps with itself. Hence higher frequencies take the form of lower frequencies. This interference of the frequency components is called as aliasing effect.

2. State Sampling theorem.

A bandlimited signal of finite energy, which has no frequency components higher than W Hz, may be completely recovered from the knowledge of its samples taken at the rate of 2W samples per second.

3. Mention the merits of DPCM.

1 .Bandwidth requirement of DPCM is less compared to PCM.

2. .Quantization error is reduced because of prediction filter

3.Numers of bits used to represent one sample value are also reduced compared to PCM.