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Internal Assesment Test - III

Sub:	Communication Theory							Code:	21EC44
Date:	09/ 09/ 2023	Duration:	90 mins	Max Marks:	50	Sem:	4^{th}	Branch:	ECE
Answer Any FIVE FULL Questions									

		Marks	OBE	
			CO	RBT
1.	Derive the expression for figure of merit of DSBSC receiver.	[10]	CO3	L2
2.	Derive the expression for figure of merit of AM receiver.	[10]	CO3	L2
3.	With the help of block diagram and waveforms explain generation and detection of PPM	[10]	CO4	L2
4	With neat diagram explain the different components of a PCM system	[10]	CO5	L2

PTO

5	What are Line codes? Give representations for any 3 line codes used in digital modulation schemes. Represent the binary data: 10011101 in i) Unipolar NRZ ii) Bipolar NRZ iii) Unipolar RZ iv) Unipolar RZ v) Split phase formatting.	[10]	CO5	L3
6	Derive the expression for signal to noise ratio of a uniform quantoizer.	[10]	CO5	L3
7	With the help of necessary diagrams explain Delta modulation. With relavent example explain granular noise and slope overload distortion.	[10]	CO5	L3
8	Explain TDM with a neat block diagram.	[10]	CO4	L2

Solution

1. DSBSC Noise

2.2) NOISE IN DSB-SC RECEIVERS

- A DSB-SC signal is given by

$$s(t) = m(t)c(t) \quad (1)$$

where,
$m(t)$ = message signal and let us assume that the message signal power is 'P' watts
$c(t) = A_c cos2\pi f_c t$ = carrier signal and the power of the carrier signal is $\frac{A_c^2}{2}$

Hence,

$$s(t) = A_c m(t) cos2\pi f_c t \quad (2)$$

- The combination $s(t) + w(t)$ is applied to a bandpass filter, the BPF is actually a narrow- BPF such that $f_c \gg B_T$,
- After passing from BPF, wideband noise $w(t)$ gets converted into narrowband noise $n(t)$
- The filtered signal $x(t)$ available for demodulation is defined by

$$x(t) = s(t) + n(t)$$

$$x(t) = A_c m(t) cos2\pi f_c t + n(t) \quad (3)$$

- The power of the noise $n(t)$ is given by $N_o W$, where W is the bandwidth of message signal

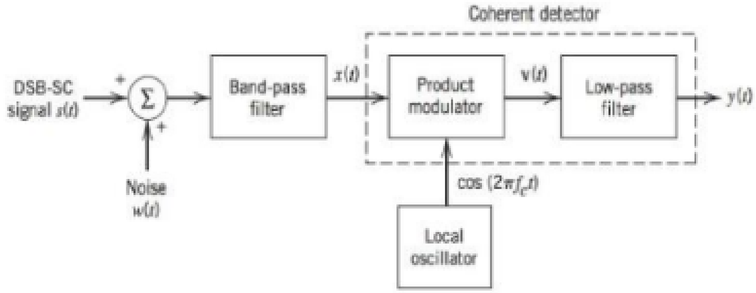
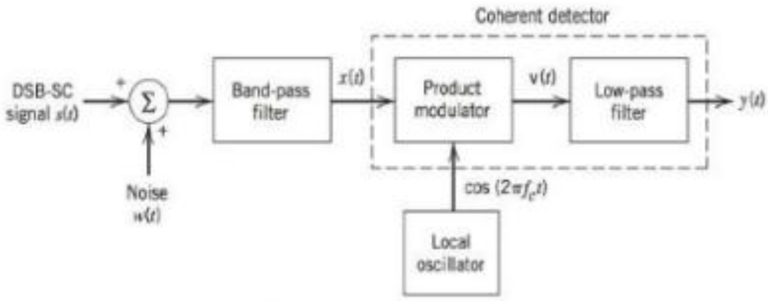


Fig. Model of DSB-SC receiver using coherent detection.

- we define **the *channel signal-to-noise ratio*,**

$$(SNR)_C = \frac{\text{average power of the modulated signal}}{\text{the average power of noise in the message bandwidth}}$$

$$(SNR)_C = \frac{\frac{A_c^2}{2}P}{N_o W} = \frac{A_c^2 P}{2N_o W} \quad (4)$$



In the coherent detecter the incoming signal x(t) is multiplied by the locally generated carrier signal to produce v(t) which is given by

$$\Rightarrow v(t) = x(t) cos2\pi f_c t$$

$$\Rightarrow v(t) = (s(t) + n(t)) cos2\pi f_c t \quad (5)$$

$$\because n(t) = n_I(t) cos2\pi f_c t - n_Q(t) sin2\pi f_c t \quad (6)$$

$$\Rightarrow v(t) = (A_c m(t) cos2\pi f_c t + n_I(t) cos2\pi f_c t - n_Q(t) cos2\pi f_c t) cos2\pi f_c t \quad (7)$$

After passing from a LPF, all the higher frequency terms will be eliminated, the output is given by

$$y(t) = \underbrace{\frac{A_c m(t)}{2}}_{\text{Demodulated Signal}} + \underbrace{\frac{n_I(t)}{2}}_{\text{Noise}} \quad (8)$$

The power of the demodulated signal $\frac{A_c m(t)}{2}$ is $\frac{A_c^2 P}{4}$

The power of the noise $\frac{n_I(t)}{2}$ is $\frac{N_o W}{2}$

- The *output signal-to-noise ratio*,

$$(SNR)_O = \frac{\text{average power of the demodulated message signal}}{\text{the average power of the noise}}$$

measured at the receiver output.

$$(SNR)_O = \frac{\frac{A_c^2 P}{4}}{\frac{N_o W}{2}}$$

$$(SNR)_O = \frac{A_c^2 P}{2N_o W} \quad (9)$$

Finally we need to find out 'Figure of Merit' of DSB-SC as

$$FOM = \frac{(SNR)_O}{(SNR)_C}$$

$$\boxed{FOM = 1} \quad (10)$$

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2. AM Receiver Noise

2.3) NOISE IN AM RECEIVERS

- An AM signal is given by

$$s(t) = A_c(1 + k_a m(t))cos2\pi f_c t \quad (1)$$

where,
$\boldsymbol{m(t)}$ = message signal and let us assume that the message signal power is **'P'** watts
$\mathbf{c(t)} = A_c cos2\pi f_c t$ =carrier signal and
the average power of the carrier signal is $\frac{A_c^2}{2}$

$$s(t) = A_c cos2\pi f_c t + A_c k_a m(t)cos2\pi f_c t \quad (2)$$

So average power in $s(t) = \frac{A_c^2}{2} + \frac{A_c^2 k_a^2 P}{2} = \frac{A_c^2}{2}(1 + k_a^2 P)$

- The combination $s(t) + w(t)$ is applied to a bandpass filter, the BPF is actually a narrow- BPF such that $f_c \gg B_T$,
- After passing from BPF, wideband noise $w(t)$ gets converted into narrowband noise $n(t)$
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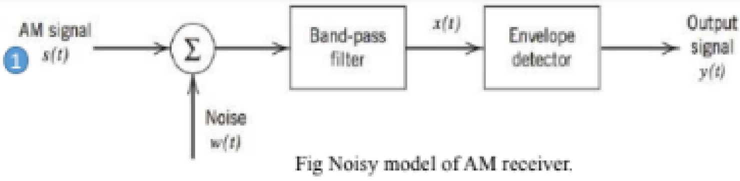

Fig Noisy model of AM receiver.

- The power of the noise $n(t)$ is given by $\boldsymbol{N_o W}$, where W is the bandwidth of message signal
- we define **the *channel signal-to-noise ratio***,

$$(SNR)_C = \frac{\textit{average power of the modulated signal}}{\textit{the average power of noise in the message bandwidth}}$$

$$(SNR)_C = \frac{\frac{A_c^2}{2}(1 + k_a^2 P)}{N_o W} = \frac{A_c^2(1 + k_a^2 P)}{2N_o W} \quad (4)$$

2.

2.3) NOISE IN AM RECEIVERS

- An AM signal is given by

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the average power of the carrier signal is $\frac{A_c^2}{2}$

$$s(t) = A_c cos2\pi f_c t + A_c k_a m(t)cos2\pi f_c t \quad (2)$$

So average power in $s(t) = \frac{A_c^2}{2} + \frac{A_c^2 k_a^2 P}{2} = \frac{A_c^2}{2}(1 + k_a^2 P)$

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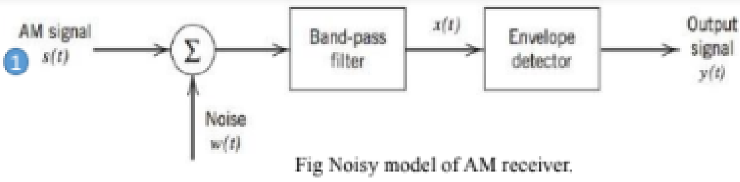

Fig Noisy model of AM receiver.

- The power of the noise $n(t)$ is given by $\boldsymbol{N_o W}$, where W is the bandwidth of message signal
- we define **the *channel signal-to-noise ratio***,

$$(SNR)_C = \frac{\textit{average power of the modulated signal}}{\textit{the average power of noise in the message bandwidth}}$$

$$(SNR)_C = \frac{\frac{A_c^2}{2}(1 + k_a^2 P)}{N_o W} = \frac{A_c^2(1 + k_a^2 P)}{2N_o W} \quad (4)$$

3. PPM

* PULSE-POSITION MODULATION :

* In pulse-duration modulation (PDM), the samples of the message signal are used to vary the duration of the individual pulses. This form of modulation is also referred to as Pulse-Width modulation or pulse-length modulation.

* In PPM, the position of a pulse relative to its unmodulated time of occurance is varied in accordance with the message signal as shown in fig(6)(d) for the case of sinusoidal modulation.

* GENERATION OF PPM WAVES :

The PPM signal which is generated is shown in fig(7)(a). The message signal m(t) is first converted into a PAM signal by means of a sample and Hold circuit, generating a staircase waveform u(t), which is shown in 8(b) for the message signal m(t) shown in fig 8(a).

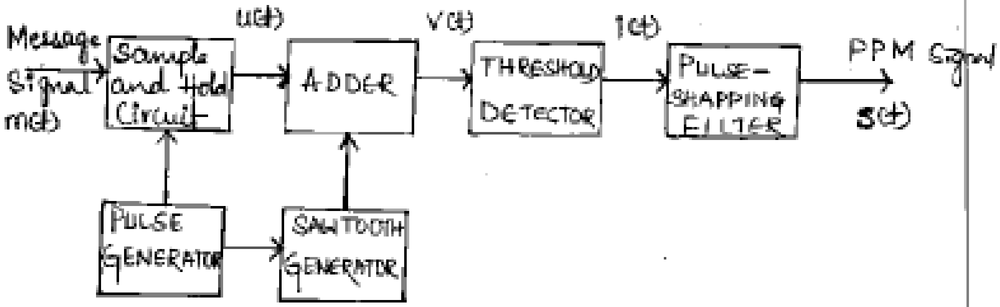

Fig 7(a) : Block diagram of PPM generator.

* Next, the signal u(t) is added to a sawtooth wave, yielding the combined signal v(t). The combined signal v(t) is applied to a threshold detector that produces a very narrow pulse each time v(t) crosses zero in the -ve going direction. The resulting sequence of "impulses" i(t) is shown in fig (8)(e). Finally, the PPM signal s(t) is generated by using this sequence of impulses to excite a filter whose impulse response is defined by the standard pulse g(t).

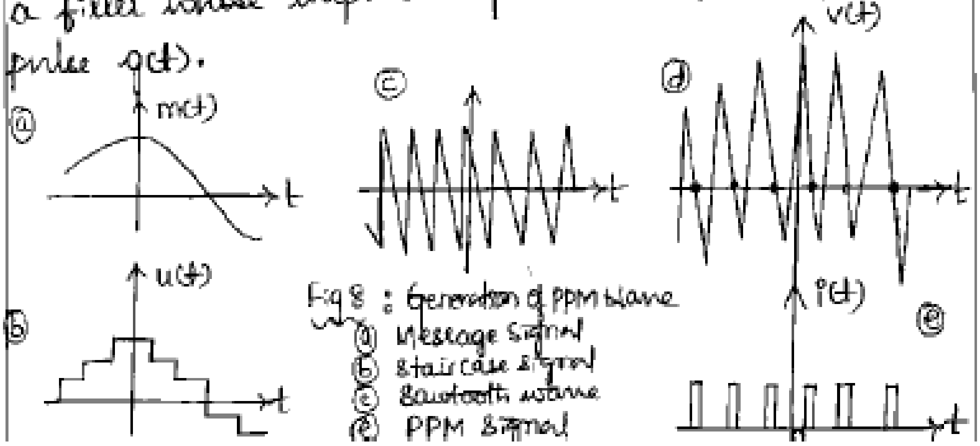

Fig 8 : Generation of PPM wave
ⓐ Message Signal
ⓑ Staircase signal
ⓒ Sawtooth wave
ⓔ PPM signal

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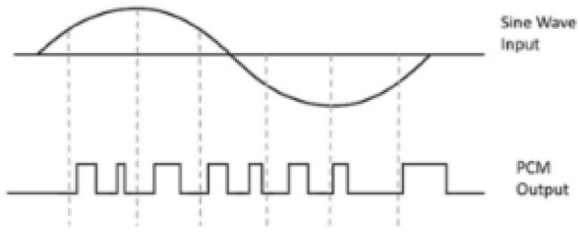
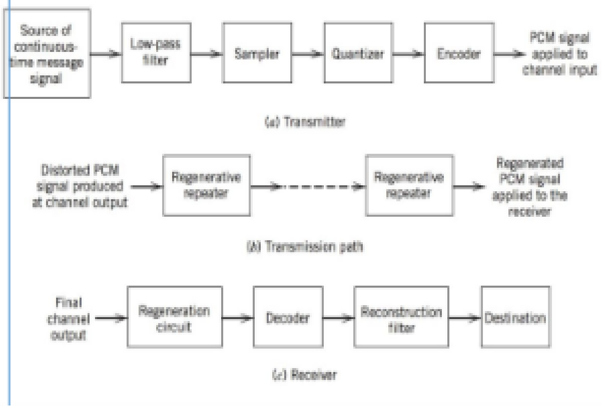
4. PCM

4).PULSE-CODE MODULATION

- A signal is pulse code modulated to convert its analog information into a binary sequence, i.e., **1s** and **0s**. The output of a PCM will resemble a binary sequence. The following figure shows an example of PCM output with respect to instantaneous values of a given sine wave.
- PCM is the most basic form of digital pulse modulation.In pulse-code modulation (PCM) a message signal is represented by a sequence of coded pulses, which isaccomplished by representing the signal in discrete form in both time and amplitude.
- The basic operations performed in the transmitter of a PCM system are sampling, quantizing, and encoding,
- The basic operations in the receiver are regeneration of impaired signals, decoding, and reconstruction of the train of quantized samples
- The quantizing and encoding operations are usually performed in the same circuit, which is called an analog-to-digital converter.





4.1)SAMPLING

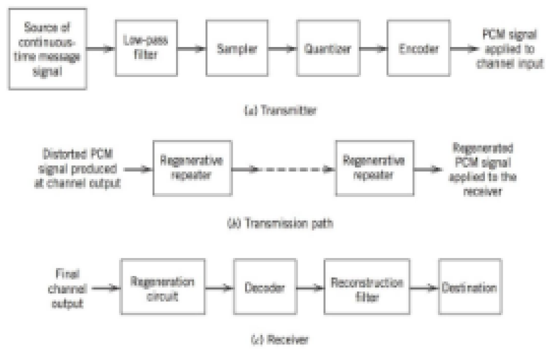
The incoming message signal is sampled with a train of narrow rectangular pulses, to ensure perfect reconstruction of the message signal at the receiver, the sampling rate must be greater than twice the highest frequency component W of the message signal in accordance with the sampling theorem.

$$f_s \geq 2W$$

$f_s = Samping\ frequency$ and W= bandwidth of message signal

4.2) QUANTIZATION

The sampled version of the message signal is then quantized, thereby providing a new representation of the signal that is discrete in both time and amplitude.
Quantizers can be of a uniform or non-uniform type.
The uniform quantizer can be characteristise as midtread or midrise type.
The use of a non-uniform quantizer is equivalent to passing the baseband signal through a compressor and then applying the compressed signal to a uniform quantizer.
The non-uniform quantizer requires to apply **companding**, it is a joint term use for the combination of a compressor and an expander.



4.3) ENCODING

After the processes of sampling and quantization we have limited discrete samples in time and amplitude, but not in the form best suited to transmission over a line or radio path.
Further encoding process is required to covert these discrete set of sample values to a more appropriate form of signal suitable for transmission over the channnel
Encoding process ensures transmitted signal more robust to noise, interference, and other channel degradations
Representing these discrete set of values as a particular arrangement of discrete events is called a code. One of the discrete events in a code is called a code element or symbol. For example, the presence or absence of a pulse is a symbol
Suppose that, in a binary code, each code word consists of R bits:Then, using such a code, we may represent a total of 2^R distinct numbers.

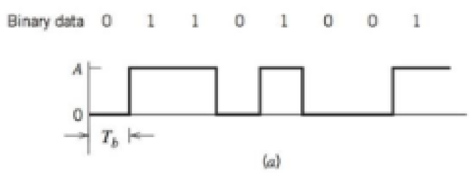
4.3.1) LINE CODES

It is in a line code that a binary stream of data takes on an electrical representation. Any one of several line codes can be used for the electrical representation of a binary data stream

4.3.1) (i) Unipolar Nonreturn-to-Zero (NRZ) Signaling.

In this line code, symbol 1 is represented by transmitting a pulse of amplitude A for the duration of the symbol, and symbol 0 is represented by switching off the pulse, as in fig. A

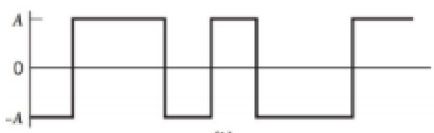
This line code is also referred to as on-off signaling. A disadvantage of on-off signaling is the waste of power due to the transmitted DC level.



(a)

4.3.1) (ii) Polar Nonreturn-to-Zero (NRZ) Signaling.

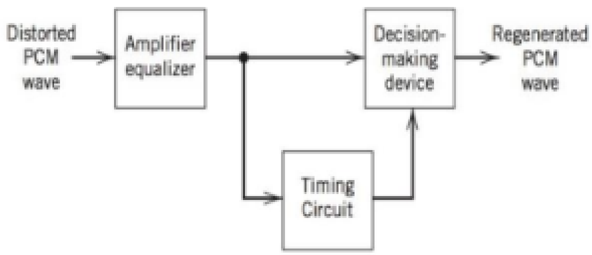
In this second line code, symbols 1 and 0 are represented by transmitting pulses of amplitudes +A and -A, respectively, as illustrated in Figure B This line code is relatively easy to generate and is more power-efficient than its unipolar counterpart.



(b)

4.4) REGENERATION

- The most important feature of any digital system lies in the ability to control the effects of distortion and noise produced by transmitting a digital signal through a channel.
- This capability is accomplished by reconstructing the signal by means of a chain of regenerative repeaters located at sufficiently close spacing along the transmission route.
- As illustrated in the block diagram, three basic functions are performed by a regenerative repeater: equalization, timing , and decision making.
- The equalizer shapes the received pulses so as to compensate for the effects of amplitude and phase distortions produced by the transmission characteristics of the channel.
- The timing circuitry provides a periodic pulse train, derived from the received pulses, for sampling the equalized pulses at the instants of time where the signal-to-noise ratio is a maximum.
- The sample so extracted is compared to a predetermined threshold in the decision-making device. In each bit interval a decision is then made whether the received symbol is a 1 or a 0 on the basis of whether the threshold is exceeded or not. If the threshold is exceeded, a clean new pulse representing symbol 1 is transmitted to the next repeater. Otherwise, another clean new pulse representing symbol 0 is transmitted.
- In this way, the accumulation of distortion and noise in a repeater span is completely removed



In practice, however, the regenerated signal departs from the original signal for two main reasons:
1. The unavoidable presence of channel noise and interference causes the repeater to make wrong decisions occasionally, thereby introducing bit errors into the regenerated signal.
2. If the spacing between received pulses deviates from its assigned value, a jitter is introduced into the regenerated pulse position, thereby causing distortion.

4.5) DECODING

The first operation in the receiver is to regenerate (i.e., reshape and clean up) the received pulses one last time. These clean pulses are then regrouped into code words and decoded (i.e., mapped back) into a quantized PAM signal. The decoding process involves generating a pulse the amplitude of which is the linear sum of all the pulses in the code word, with each pulse being weighted by its place value $(2^0, 2^1 2^2, 2^3, \ldots, 2^{R-1})$ in the code, where R is the number of bits per sample.

4.6) FILTERING

The final operation in the receiver is to recover the message signal wave by passing the decoder output through a low-pass reconstruction filter whose cutoff frequency is equal to the message bandwidth W. Assuming that the transmission path is error free, the recovered signal includes no noise with the exception of the initial distortion introduced by the quantization process.

4.8) MULTIPLEXING

- In applications using PCM, it is natural to multiplex different message sources by time division, whereby each source keeps its individuality throughout the journey from the transmitter to the receiver.
- This individuality accounts for the comparative ease with which message sources may be dropped or reinserted in a time-division multiplex system.
- As the number of independent message sources is increased, the time interval that may be allotted to each source has to be reduced, since all of them must be accommodated into a time equal to the reciprocal of the sampling rate.
- This in turn means that the allowable duration of a code word representing a single sample is reduced. However, pulses tend to become more difficult to generate and to transmit as their duratio is reduced.
- Furthermore, if the pulses become too short, impairments in the transmission medium begin to interfere with the proper operatio of the system.

✓ Accordingly, in practice, it is necessary to restrict the number of independent message sources that can be included within a time-division group.

5. Line codes
6. SNR of uniform quantizer

2.1) Uniform Quantizer

- In a uniform quantizer, the representation levels are uniformly spaced; otherwise, the quantizer is non-uniform.
- The quantizer characteristic can also be of midtread or midrise type. Midtread and midrise are two types types of uniform quantizers
- The input-output characteristic of a uniform quantizer of the midtread type, which is so called because the origin lies in the middle of a tread of the staircase like graph Fig.A
- The corresponding input-output characteristic of a uniform quantizer of the midrise type, in which the origin lies in the middle of a rising part of the staircase like graph Fig.B

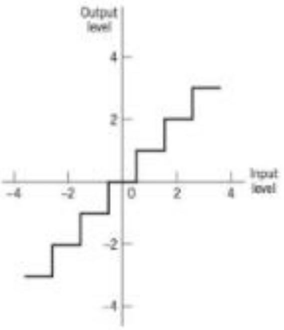


Fig.A Midtread type uniform quantizer

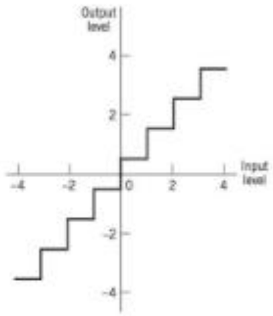


Fig.B Midrise type uniform quantizer

3).QUANTIZATION NOISE

The use of quantization introduces an error defined as the difference between the input signal m and the output signal v. This error is called quantization noise.

Typical variation of the quantization noise as a function of time, assuming the use of a uniform quantizer of the midtread type

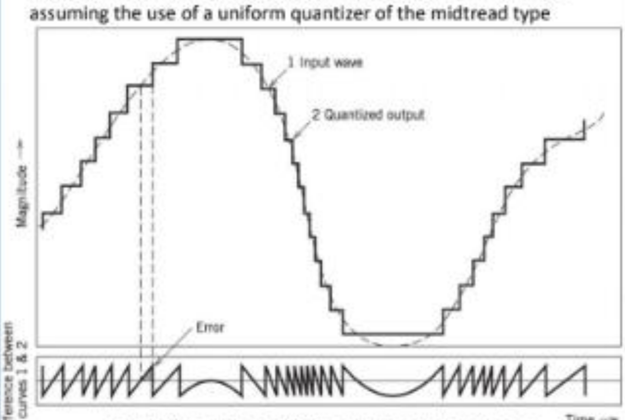


Fig.C Illustration of the quantization process

Let the quantization error be denoted by the random variable Q of sample value q.

$$\boxed{q = m - v} \quad (1)$$

or, correspondingly,

$$Q = M - V \quad (2)$$

m be the sample value of a zero-mean random variable M.

v be the sample value of a zero-mean discrete random variable V

Quantization error be denoted by the random variable Q of sample value q

Consider then an input m of continuous amplitude is in the range $(-m_{max}, m_{max})$

The step-size of the quantizer is given by

$$\Delta = \frac{2m_{max}}{L} \quad (3)$$

where L is the total number of representation levels

Let R denote the number of bits per sample used in the construction of the binary code

then, $\quad L = 2^R \quad (4)$

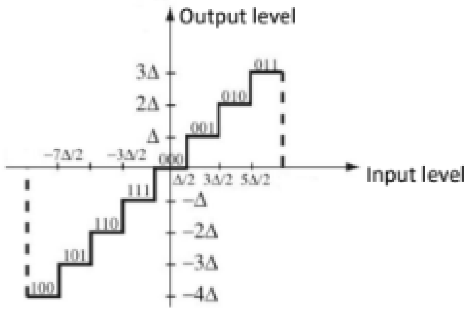
equivalently, $\quad R = \log_2 L \quad (5)$

For a uniform quantizer, the quantization error Q will have its sample values bounded by $-\Delta/2 \leq q \leq \Delta/2$

Assuming that the quantization error Q is a uniformly distributed random variable, we may thus express the probability density function of the quantization error Q as follows:

$$f_Q(q) = \begin{cases} 1/\Delta; & -\Delta/2 \leq q \leq \Delta/2 \\ 0; & elsewhere \end{cases} \quad (5)$$

with the mean of the quantization error being zero, its variance σ_Q^2 is the same as the mean-square value



$$\sigma_Q^2 = \int_{-\Delta/2}^{\Delta/2} q^2 f_Q(q)\,dq = E[Q^2] \quad (6)$$

$$\sigma_Q^2 = \int_{-\Delta/2}^{\Delta/2} q^2 \, {}^1/_\Delta \, dq = E[Q^2] \qquad \boxed{\sigma_Q^2 = \Delta^2/_{12}} \quad (7)$$

$$\sigma_Q^2 = \frac{\left(\frac{2m_{max}}{L}\right)^2}{12} \Rightarrow \sigma_Q^2 = \frac{\left(\frac{2m_{max}}{2^R}\right)^2}{12} \quad \boxed{\Rightarrow \sigma_Q^2 = \frac{1}{3} m_{max}^2 2^{-2R}} \quad (8)$$

Let P denote the average power of the message signal m(t).We may then express the output signal-to-noise ratio of a uniform quantizer as

$$(SNR)_o = \frac{P}{\sigma_Q^2} = \left(\frac{3P}{m_{max}^2}\right) 2^{2R} \quad (9)$$

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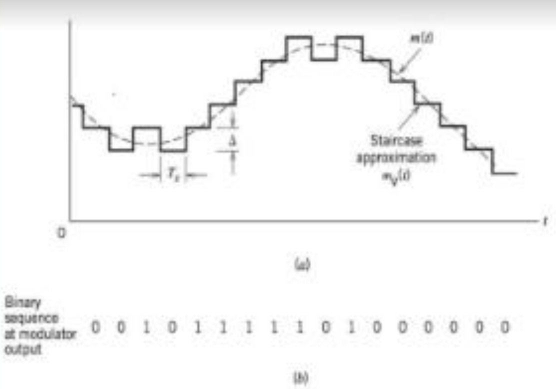
7. Delta Modulation

6) DELTA MODULATION

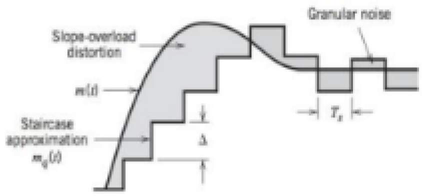
- The problem with PCM is congestion, as the number of quantization level increases, number of bits/ sample increases, hence it creates unnecessary load



- In Delta modulation, an analog input is approximated by a staircase function that moves up and down by one Quantization level (Δ) at each signal interval T_s, so it can be referred as **1 bit quantizer**
- In delta modulation (DM), an incoming message signal is oversampled (i.e., at a rate much higher than the Nyquist rate) to purposely increase the correlation between adjacent samples of the signal
- The difference between the input $m(t)$, and the approximation $m_q(t)$, is quantized into only two levels, namely, ± Δ, corresponding to positive and negative differences, respectively.
- Thus, if the approximation falls below the signal at any sampling epoch, it is increased by Δ. If, on the other hand, the approximation lies above the signal, it is diminished by Δ.
- The principal virtue of delta modulation is its simplicity. It may be generated by applying the sampled version of the incoming message signal to a digital encoder that involves a comparator, quantizer, and accumulator interconnected

8.

Delta modulation is subject to two types of quantization error: (1) slope overload distortion and (2) granular noise



(1) slope overload distortion

- For fair approximation of analog input into digital output through staircase sequence via 1 bit quantization, the step size must be compared with maximum slope in the messahe signal
- If we consider the maximum slope of the original input waveform $m(t)$,it is clear that in order for the sequence of samples $\{m_q(nT_s)\}$ to increase as fast as the input sequence of samples $\{m(nT_s)\}$ in a region of maximum slope of $m(t)$, we require that the condition

$$\frac{\Delta}{T_s} \geq \max\left|\frac{dm(t)}{dt}\right|$$

be satisfied.

- If the above condition is not followed, and

$$\frac{\Delta}{T_s} \ll \max\left|\frac{dm(t)}{dt}\right|$$

we find that the step-size Δ is too small for the staircase approximation $m_q(t)$ to follow a steep segment of the input waveform $m(t)$, with the result that $m_q(t)$ falls behind $m(t)$.

- This condition is called **slope overload**, and the resulting quantization error is called **slope-overload distortion (noise).**

(2) Granular noise

In contrast to slope-overload distortion, granular noise occurs when the step-size Δ is too large relative to the local slope characteristics of the input waveform m(t),

$$\frac{\Delta}{T_s} \gg \max\left|\frac{dm(t)}{dt}\right|$$

thereby causing the staircase approximation $m_q(t)$ to hunt around a relatively flat segment of the input waveform

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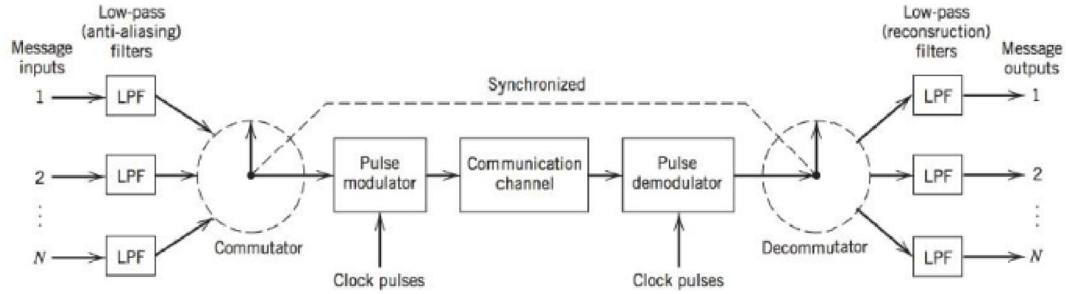
9. TDM

- An important feature of the sampling process is a **conservation of time.**
- The transmission of the message samples engages the communication channel for only a fraction of the sampling interval on a periodic basis
- So, some of the time interval between adjacent samples is cleared for use by other independent message sources on a time-shared basis
- A *time-division multiplex* (TDM) *system,* enables the joint utilization of a common communication channel by a transmission of multiple independent messages without mutual interference among them



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Transmitter

- The concept of TDM is illustrated by the block diagram, each input message signal is first restricted in bandwidth by a low-pass pre-alias filter
- The lowpass filter outputs are then applied to a *commutator,* which is usually implemented using electronic switching circuitry
- The function of the commutator is twofold:

1) to take a narrow sample of each of the N input messages at a rate f_s that is slightly higher than $2W$,
where W is the cutoff frequency of the pre-alias filter, and
(2) to sequentially interleave these N samples inside the sampling interval T_s

- Following the commutation process, the multiplexed signal is applied to a pulse modulator, the purpose of which is to transform the multiplexed signal into a form suitable for transmission over the common channel

Receiver

- At the receiving end of the system, the received signal is applied to a *pulse demodulator,* which performs the reverse operation of the pulse modulator.
- The narrow samples produced at the pulse demodulator output are distributed to the appropriate low-pass reconstruction filters by means of a *decommutator,* which operates in *synchronism* with the commutator in the transmitter.

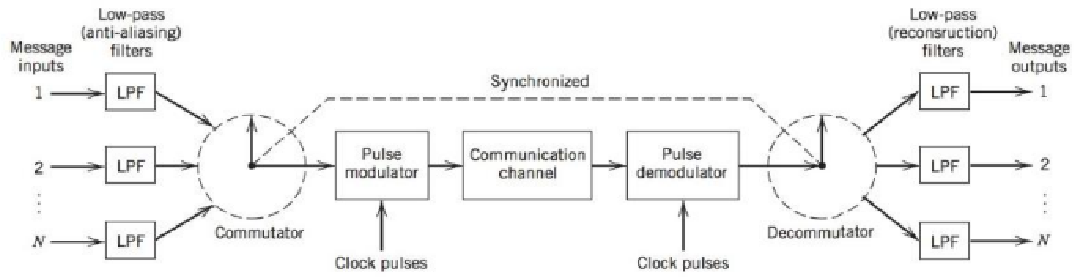

Fig. Block diagram of TDM system