كلية المستقبل الجامعة

قسم هندسة تقنيات الحاسبات

المرحلة الثالثة

المادة : اتصالات رقمية

Q1- What are the advantages and disadvantages of digital systems?

Ans:

Advantages:

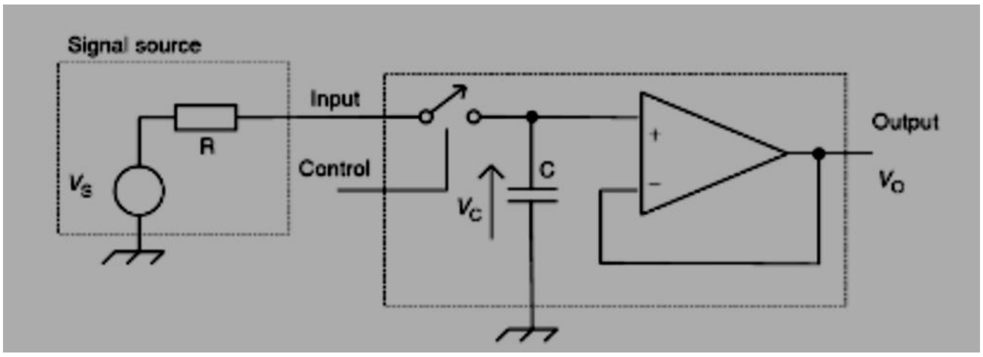
1. Hardware is more flexible.
2. Lower cost compared with analog system.
3. Easier and more efficient to multiplex several digital signals.
4. Can combine different signal types data, voice, text, etc..
5. Can use packet switching.
6. Encryption and privacy techniques are easier to implement.
7. Better overall performance.

Disadvantages

1. Require reliable synchronization.
2. Requires A/D conversions at high rate.
3. In general requires larger bandwidth.

Q2- Explain with draw the method that used for PAM signals generation?

Ans: Sample and hold circuit (and explain principle of circuit work)



The switch closes only when that particular channel is to be sampled. If the source impedance **r** is small, the capacitor voltage changes to the input voltage within the time τ that switch is closed.

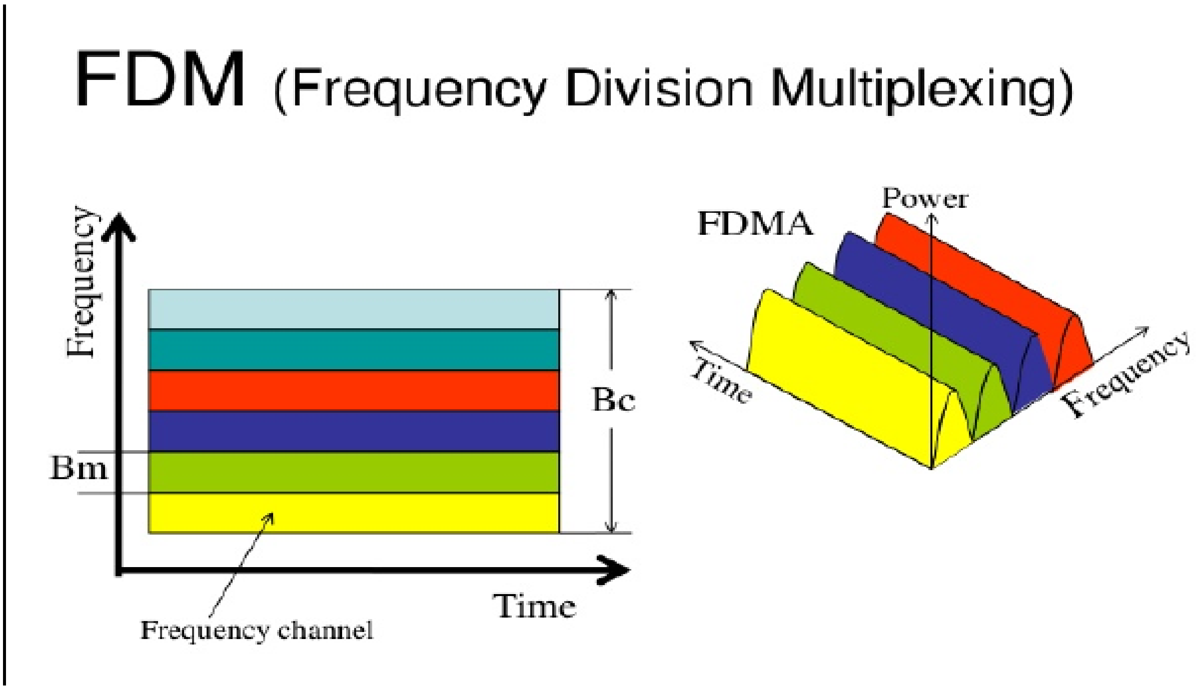
The load impedance **R** is arranged to be high so that the capacitor retains the voltage level until the switch is closed again. Therefore the sample and hold circuit accepts only those values of the input which occur at the sampling times and then holds them until the next sampling time.

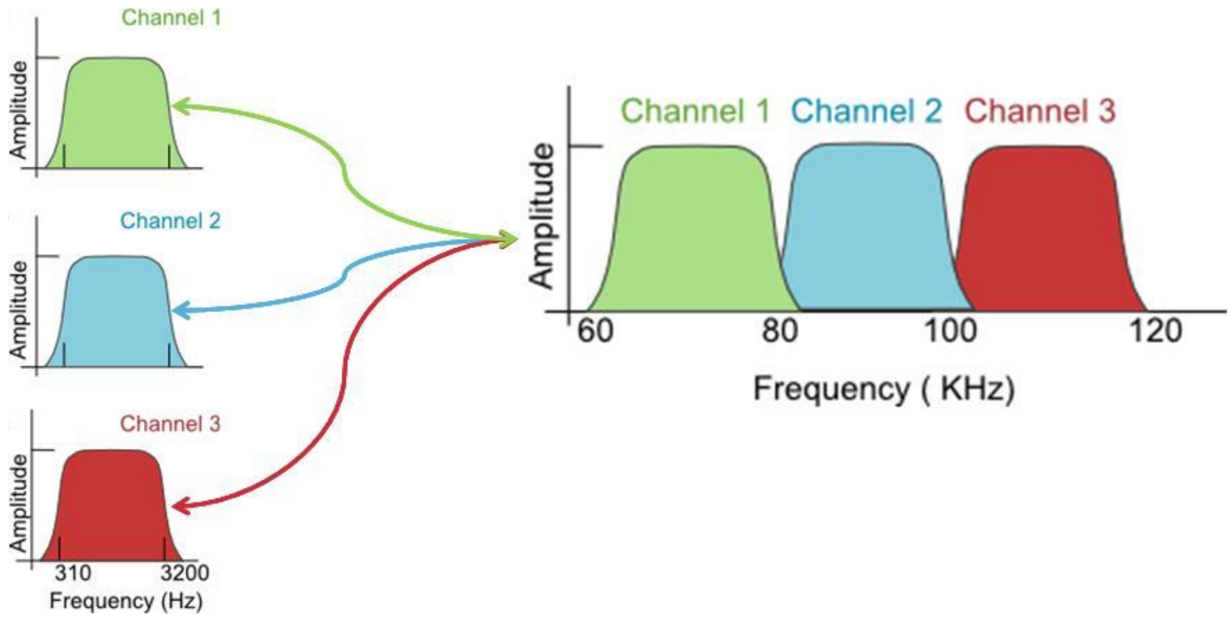
Q3- There is two common methods are used for multiplexing, define them then explain with draw the types of TDM technique?

Ans:

1. Frequency division multiplexing (FDM):

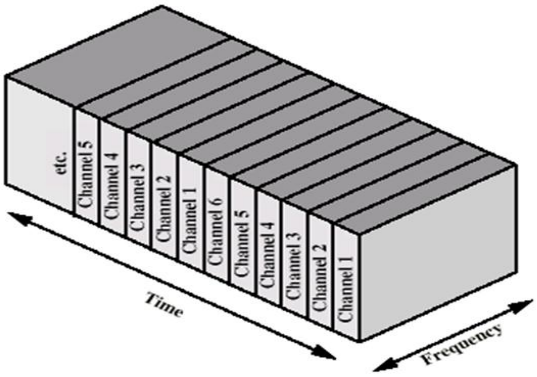
In FDM multiple sources that originally occupied the same frequency spectrum are each converted to a different frequency band and transmitted simultaneously over a single transmission medium. FDM is an analog multiplexing scheme.





2. **<u>Time Division Multiplexing (TDM)</u>**

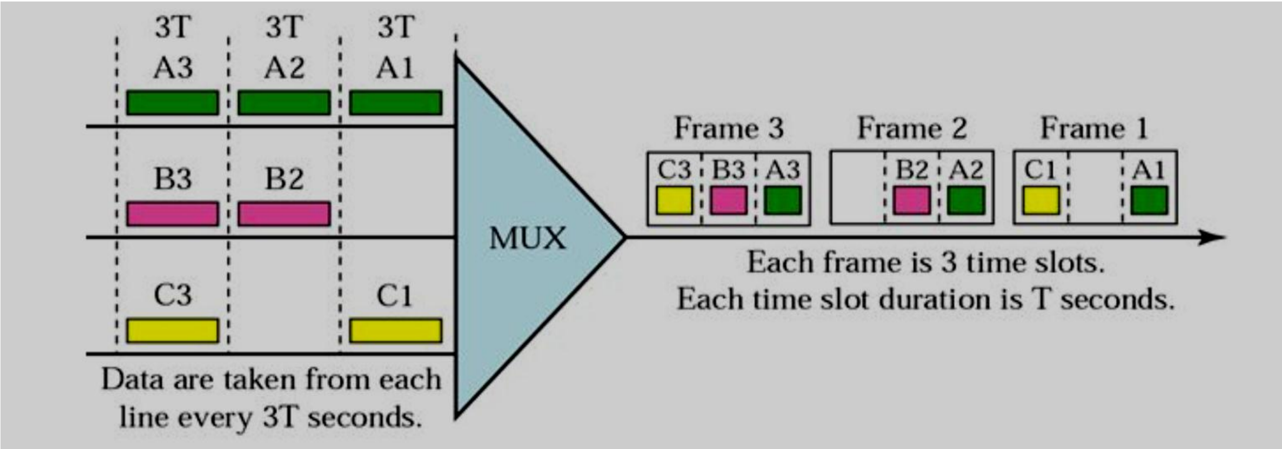
With TDM system, transmission from multiple sources occurs on the same transmission medium but not at the same time. Transmission from various sources is interleaved in time domain. The unused time regions between slot assignments, called guard times, act as buffer zone to reduce interference.



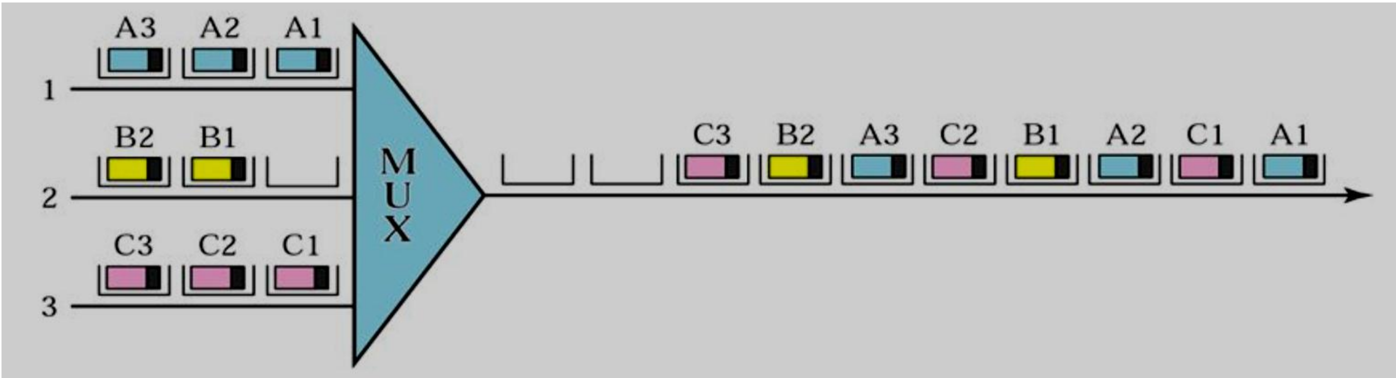
Time is segmented in to intervals called frames. Each frame is further partitioned in to assignable user time slots.

Types of TDM:

1. ***Fixed-assignment TDM***: In fixed assignment TDM scheme, any slot that has no data to send during a particular frame, that slot is wasted.



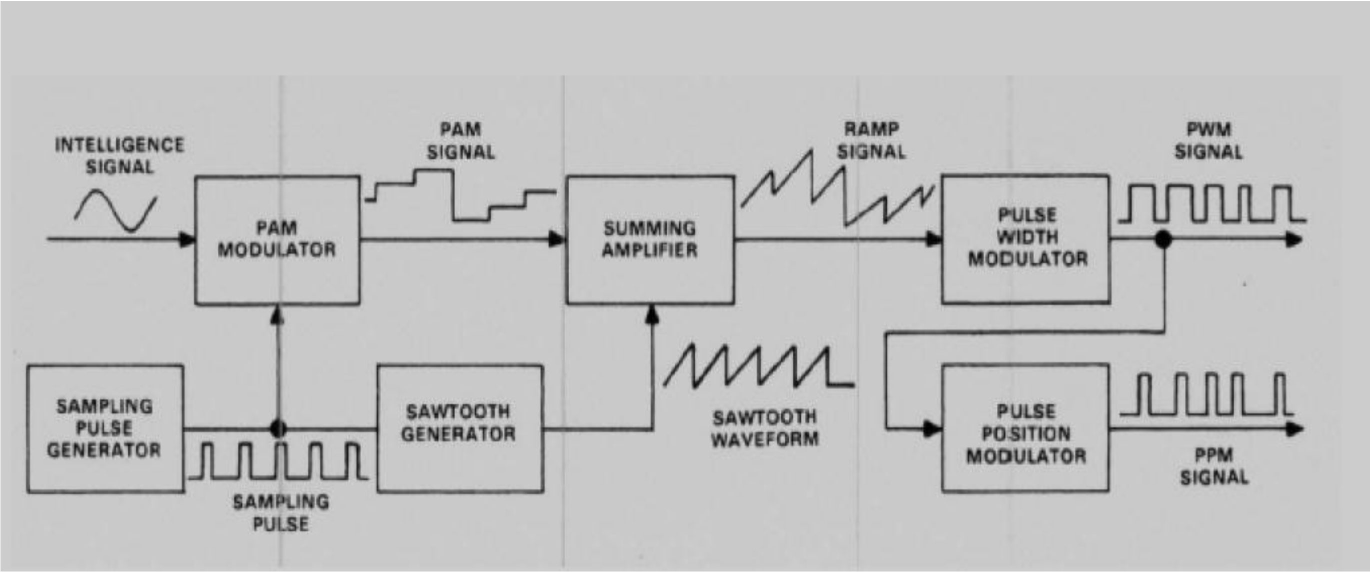
2. ***Dynamic assignment TDM***: In dynamic assignment TDM scheme, dynamic slot assignment used that enables any available slot to take place to be sent during a particular frame.



Q4- Explain with draw the method that used for PWM signals generation?

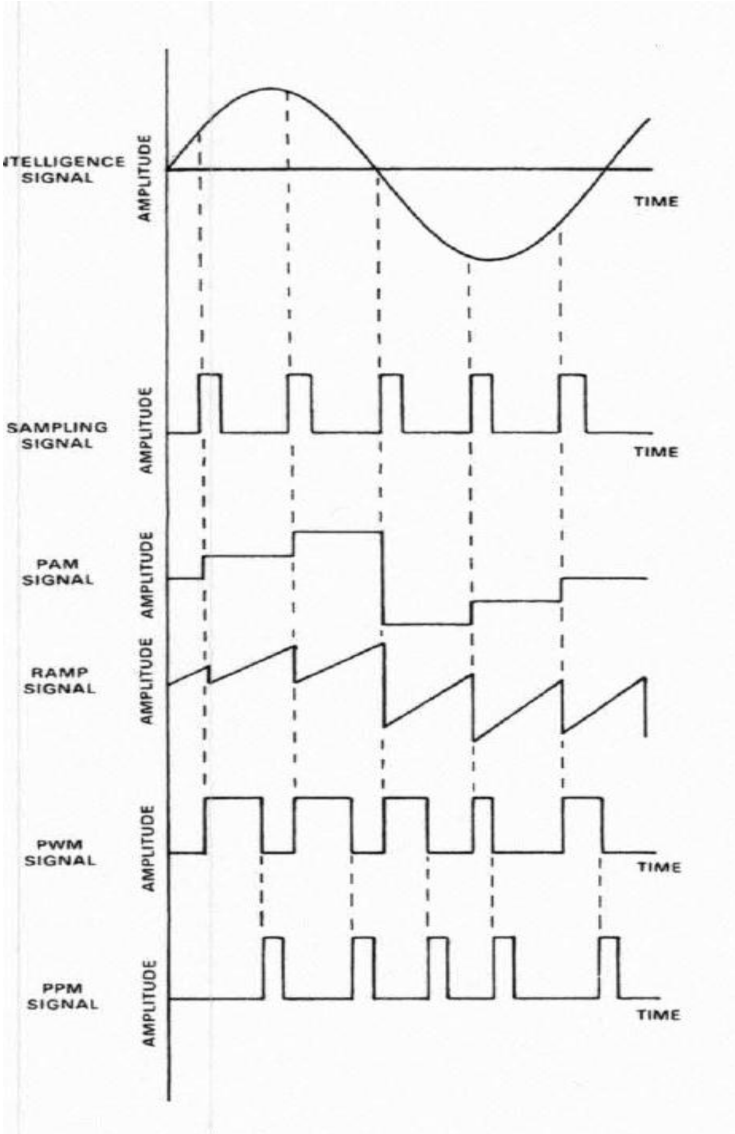
Q5- Explain with draw the method that used for PPM signals generation?

Same Answer:



The ramp generator produces a precision ramp voltage which has peak to peak amplitude slightly larger than the maximum amplitude range of the input signals. This ramp voltage is the basis for the amplitude to timing conversion and therefore must be accurately known. The comparator is a high gain amplifier intended for two stated operation. If input signal is higher than a preset reference level, the output is held in one state (i.e. a given voltage level). Whenever the input signal level is less than the reference level, the output is held in the other state. Which output state is present, then, depends upon whether the input is above and below the threshold (reference level) of the comparator.

A convenient way to generate PPM is to use PWM waveform generated above and then trigger a constant width pulse generation those edge of the PWM waveform with a negative slope.



INTELLIGENCE SIGNAL
AMPLITUDE
TIME
SAMPLING SIGNAL
AMPLITUDE
TIME
PAM SIGNAL
AMPLITUDE
RAMP SIGNAL
AMPLITUDE
PWM SIGNAL
AMPLITUDE
TIME
PPM SIGNAL
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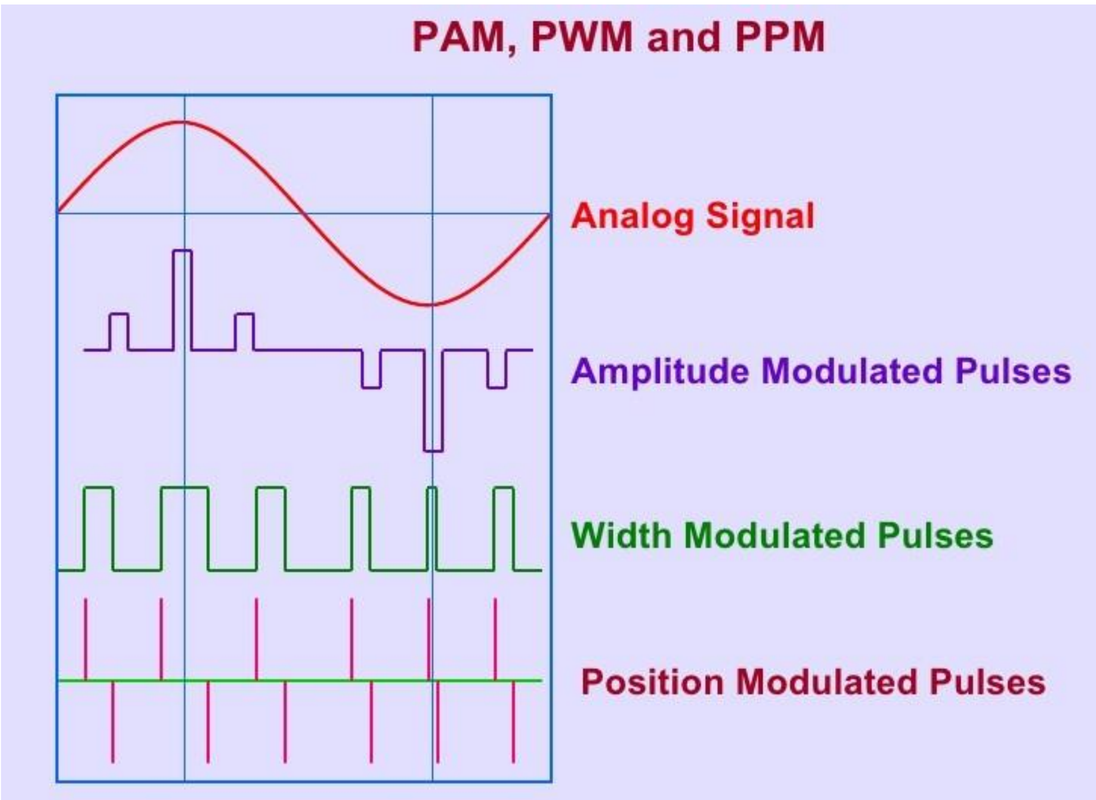
TIME



PAM, PWM and PPM
Analog Signal
Amplitude Modulated Pulses
Width Modulated Pulses
Position Modulated Pulses

Q6- Derive the following Sampling equation:

$$X_s(f) = X(f) \otimes X_\delta(f) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} X(f - nf_s)$$

Ans:

$$x_\delta(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

Let us choose $T_s = \frac{1}{2f_m}$, so that Nyquist rate is just satisfied.

Using shifting property of the impulse function the $x_s(t)$, shown in Fig. (e), can be given by

$$x_s(t) = x(t)x_\delta(t) = \sum_{n=-\infty}^{\infty} x(t)\delta(t - nT_s)$$

$$X_\delta(f) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} \delta(f - nf_s)$$

The convolution with an impulse function simply shifts the original function, as follows:

$$X(f) \otimes \delta(f - nf_s) = X(f - nf_s)$$

The Fourier transform of the sampled waveform, $X_s(f)$, can be given by:

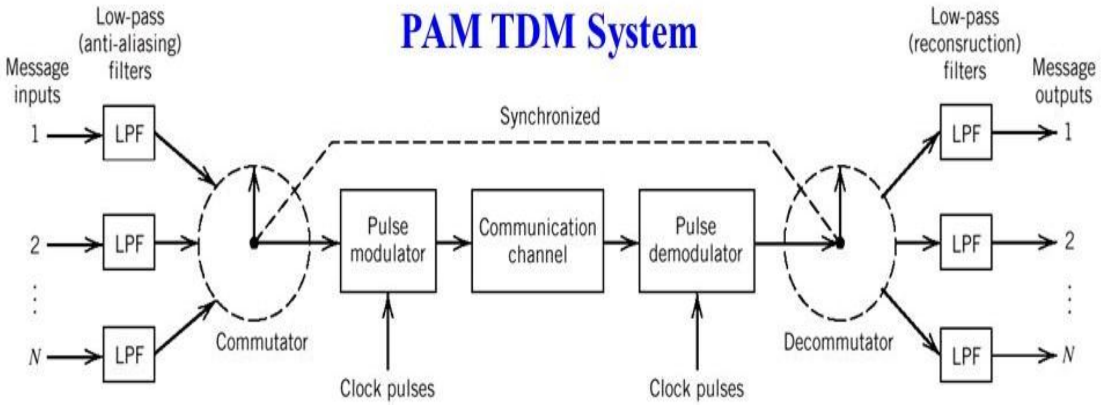
$$X_s(f) = X(f) \otimes X_\delta(f) = X(f) \otimes [\frac{1}{T_s} \sum_{n=-\infty}^{\infty} \delta(f - nf_s)]$$

$$= \frac{1}{T_s} \sum_{n=-\infty}^{\infty} X(f - nf_s)$$

Q7/ Draw and explain PAM/TDM System?

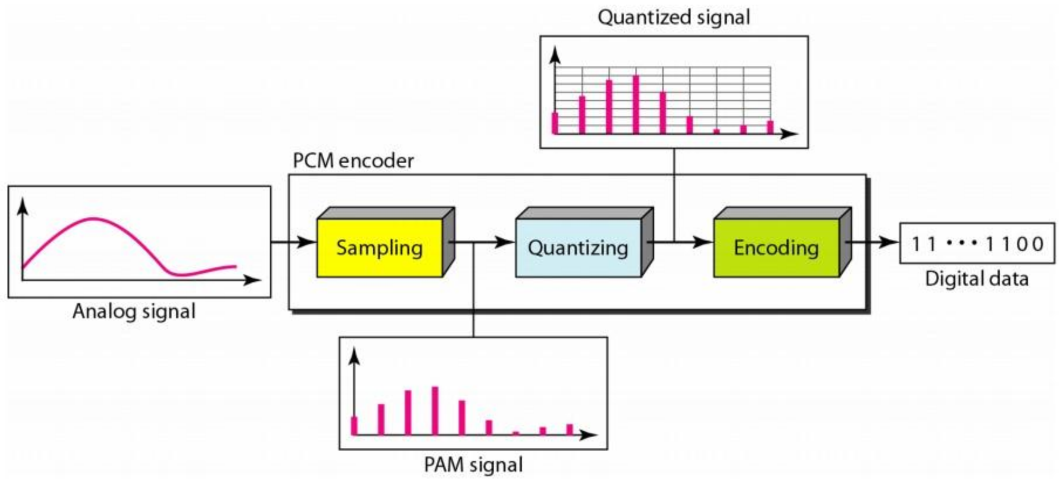
Answer:

Suppose we wish to time multiplexed two signals using PAM. Let us assume that both input signal $f_1(t)$ and $f_2(t)$ are low pass, and band limited to 3KHz. The sampling theorem states that each must be sampled at a rate no less than 6KHz. This requires a 12KHz minimum clock rate for the two channel system. Figure below shows the block diagram of PAM/TDM system.



Q8/ Draw the steps required in PCM generation.

Answer:



Q9/Ten voice channels each of bandwidth (B.W) =3.2 KHz are sequentially sampled at 8 KHz and TDM'ed.

(a) What is the system bandwidth (B.W).

(b) If TDM'ed signal is PCM'ed using 8-level quantization, find bit rate (Rb)

Answer:

(a) Without guard band

$$T_s = \frac{1}{f_s} \; \frac{1}{8KHz} = 125\,\mu\text{sec}.$$

∵ 10 voice channels, ∴ 10 samples

$$\therefore \text{ Necessary B.W} = \frac{10}{125*10^{-6}} = 80KHz$$

(b) $k = \log_2 L = \log_2 8 = 3 \frac{bit}{sample}$

$$\therefore R_b = 80*10^3 \frac{sample}{\text{sec}.} * 3 \frac{bit}{sample} = 240 \frac{bit}{\text{sec}.}$$

Q10/

The information in an analog waveform, with maximum frequency f_m=3 KHz, is to be transmitted over M level PCM system, where number of pulse level M=16. The quantization distortion is specified not to exceed $\pm 1\%$ of the peak-to-peak analog signal.

(a) What is the minimum number of bits/sample, or bits/PCM word that should be used in this PCM system.

(b) What is the minimum required sampling rate, and what is the resulting bit transmission rate.

(c) What is the PCM pulse or symbol transmission rate.

Answer:

Note:- in this example we are considered with two types of levels, the number of quantization levels (L), and the 16 level of the multilevel PCM pulses (M).

(a) By using

$$L \geq \frac{1}{2p} \text{ levels}$$

$$\therefore k \geq \log_2 \frac{1}{2p} \text{ bits}$$

where L=number of quantization level, k=number of bits, and p=fraction of peak-to-peak analog voltage.

$$\therefore k \geq \log_2 \frac{1}{2*0.01} = \log_2 50 = 5.6$$

$$\therefore k = 6 \Rightarrow L = 2^k = 64$$

The number of bit/samples =k =6

(b) $f_s=2f_m=6000$ sample/second

$\therefore$ bit transmission rate $R_b \geq kf_s$

$$\therefore R_b = 6*6000 = 36000 \text{ bit/sec}$$

(c) Since multilevel pulses are to be used with $M=2^m=16$

$\therefore m=4$ bit/symbol

$\therefore$ The bit stream will be partitioned into groups of 4-bits to form a new 16-level PCM digit.

$\therefore$ Symbol transmission rate (R_s)

$$R_s = \frac{3600}{4} = 9000 \frac{symbol}{sec}$$

Q11/ (a) Find the minimum required bandwidth for the base-band transmission of 4-level PCM pulses sequence having a data rate of Rb=2400 bit/sec. if the system transfer characteristic consists of a raised cosine spectrum with 100% excess bandwidth (r=1).

(b) The same PCM sequence is modulated on to a carrier wave, so that the base-band spectrum is shifted and centered at frequency f_0. Find the minimum required DSB bandwidth for transmitting the modulated PCM sequence. Assume that the system transfer characteristic is the same in part (a).

Answer:

(a) $\because$ M=2^m, since M=4, $\therefore$ m=2 bit/symbol

$\therefore$ Pulse or symbol rate

$$R_s = \frac{R_b}{m} = \frac{2400 bit/\sec.}{2bit/symbol} = 1200 \frac{symbol}{\sec.}$$

$\therefore$ Minimum bandwidth $= \dfrac{(1+r)R_s}{2} = \dfrac{(1+1)*1200}{2} = 1200 Hz$

(b) $W_{DSB} = (1+r)R_s = (1+1)*1200 = 2400 Hz$

Q12/ Draw and explain Delta Modulator(DM) modulator and demodulator generation.

Answer: The comparator computes the difference between two inputs. The quantizer consists of hard limiter with an input/output relation that is scaled version of the signum function. The accumulator increments the approximation by a step Δ in positive or negative direction, depending on the algebraic sign of the error signal ***e(nT$_s$)***.

Demodulation is subjected to two types of error:-

(1) Slop over load direction.

(2) Granular noise.

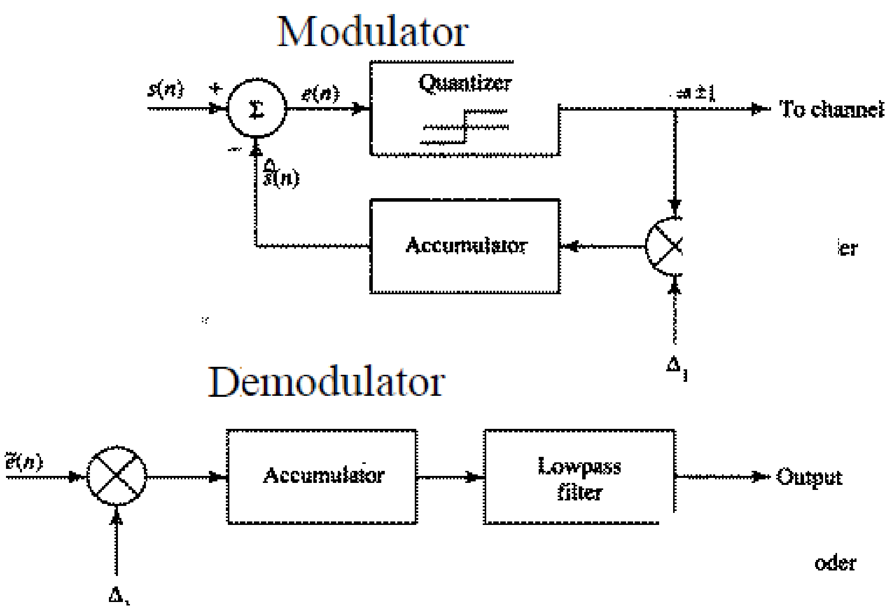
Equation (c) may be observed as a digital equivalent of integration in the sense that it represents the accumulation of positive and negative increments of magnitude Δ, also, denoting the quantization error by ***q(nT$_s$)***, as shown by

$$m_q(nT_s) = m(nT_s) + q(nT_s)$$

From equation (a) may be observed

$$e(nT_s) = m(nT_s) - m(nT_s - T_s) - q(nT_s - T_s)$$

Thus, except for the quantization error ***q(nT$_s$-T$_s$)***, the quantizer



Q13/

A Delta modulator is used to encode speech signal band-limited to 3KHz with sampling frequency 100 KHz. For ±1 volt peak signal voltage, find Minimum step size to avoid slope overloading.

Solution:-

For DM system, if input signal $f(t) = b\cos\omega_m t$.

(a) $\therefore \left|\frac{df(t)}{dt}\right|_{max} = b2\pi f_m$

if step size used in DM system=a

$$\therefore f_s > \frac{2\pi f_m}{a/b}$$

$$\therefore a \geq \frac{2\pi f_m b}{f_s}$$

$$\frac{2\pi * 3 * 10^3 * 1}{100*10^3} = 0.188V$$ minimum step size.

Q14/ Channel 1 of two channels PAM system handles 8KHz signal. Channel 2 handles 10 KHz signals. The two channels are sampled at equal intervals of time using very narrow pulses at the lowest frequency that is theoretical adequate. The sampled signals are time multiplexed and passed through a LPF before transmission.

(1)What is the minimum clock frequency of the PAM system?
(2)Find Bx and Tx.
(3)What is the minimum cut off frequency of LPF used before transmission that will preserve the amplitude information on the output pulses?
(4) What would be the minimum bandwidth if these channels were frequency multiplexed, using AM technique and SSB technique?

(1)

$$f_{s1} = 2 * f_{m1}$$

$$f_{s1} = 2 * 8 = 16KHz$$

$$f_{s2} = 2 * f_{m2}$$

$$f_{s2} = 2 * 10 = 20KHz$$

In order to sample channel 2 adequately

$$f_s = f_{s2} = 20KHz$$

$\therefore$ The minimum clock rate $= n * f_s$

$$= 2 * 20 = 40KHz$$

(2)

$$T_s = \frac{1}{f_s} = \frac{1}{20KHz} = 50\mu\text{sec}$$

□ $n = 2$

$$T_x = \frac{T_s}{n} = \frac{50}{2} = 25\mu\text{sec}$$

□ $B_x \geq \dfrac{1}{2T_x}$

$\therefore B_x = 20KHz$

3.Frequency of LPF =20KHz

(4) For AM

$$\min BW. = 2(f_{m1} + f_{m2}) = 2(8+10) = 36KHz$$

For SSB

$$\min BW. = f_{m1} + f_{m2} = 8 + 10 = 18KHz$$

Q15/ Channel 1 of Six channels PAM system handles 1KHz signal. Channel 2 handles 2 KHz signals, channel 3 handles 3 KHz, channel 4 handles 6 KHz and channel 5 handles 6 KHz and channel 6 handles 25 KHz. The six channels are sampled at equal intervals of time using very narrow pulses at the lowest frequency that is theoretical adequate. The sampled signals are time multiplexed and passed through a LPF before transmission.

a. What is the minimum clock frequency of the PAM system?
b. Find Tx and Bx?
c. What is the minimum cut off frequency of LPF used before transmission that will preserve the amplitude information on the output pulses?
d. What would be the minimum bandwidth if these channels were frequency multiplexed, using AM technique and SSB technique?

Answer:

In order to sample channel 6 adequately:

Fs max=fs6=50KHz

Fclock=nfsmax=6*50=300KHz

Ts=1/fs=1/50K=20 µs

Tx=Ts/6=20 µs/6=3.33 µs

Bx=1/2Tx=150.015KHx

B.w AM=2fm total=2(1+2+3+6+6+25)=86KHz

B.w SSB=fm total=(1+2+3+6+6+25)=43 KHz

Q16/ Channel 1 of four channels PAM system handles 8KHz signal. Channel 2 handles 10 KHz signals, channel 3 handles 15KHz, and channel 4 handles 20KHz. The four channels are sampled at equal intervals of time using very narrow pulses at the lowest frequency that is theoretical adequate. The sampled signals are time multiplexed and passed through a LPF before transmission.

a. What is the minimum clock frequency of the PAM system?
b. Find Tx and Bx?
c. What is the minimum cut off frequency of LPF used before transmission that will preserve the amplitude information on the output pulses?
d. What would be the minimum bandwidth if these channels were frequency multiplexed, using AM technique and SSB technique?

Q17/ The information in an analog waveform, with maximum frequency fm=1 KHz, is to be transmitted over M level PCM system, where number of pulse level M=64. The quantization distortion is specified not to exceed 1.5% of the peak-to-peak analog signal.

(a)What is the minimum number of bits/sample, or bits/PCM word that should be used in this PCM system.

(b)What is the minimum required sampling rate, and what is the resulting bit transmission rate.

(c) What is the PCM pulse or symbol transmission rate?

Q18/ Two low pass signals, each band limited 4KHz, are to be time multiplexed into a single channel using PAM. Each signal is impulse sampled at a rate 10KHz. The time multiplexed signal waveform is filtered by an ideal LPF before transmission.

(a) What is minimum clock frequency of the system?

(b) What is the minimum cut off frequency of the LPF?

(c) In the receiver side, determine the minimum and maximum acceptable bandwidth of the LPF used in retrieving the analog signal?

Ans. (a) 20KHz (b) 10KHz (c) 4KHz, 6KHz.

Q19/ The information in an analog waveform, with maximum frequency fm=6 KHz, is to be transmitted over M level PCM system, where number of pulse level M=256. The quantization distortion is specified not to exceed 3% of the peak-to-peak analog signal.

(a)What is the minimum number of bits/sample, or bits/PCM word that should be used in this PCM system.

(b)What is the minimum required sampling rate, and what is the resulting bit transmission rate.

(c) What is the PCM pulse or symbol transmission rate?

Q20/ A Television signal with a bandwidth of 4.2 MHz is transmitted using binary PCM. The number of quantization levels is 1024. Assume that the signal is sampled at the rate of 20% above nequist rate, find:

a) Minimum number of bits/sample. b) Transmission bandwidth.

c) Final bit rate. d) Output signal to quantization noise ratio.

Q21/ The bandwidth of TV video plus audio signal are 4.5 MHz. If the signal is converted to 64- level PCM bit stream with 512 quantization levels, assume that the signal is sampled at the rate of 20% above nyquist rate, determine:

a) Code word length per PCM. b) Transmission bandwidth.

b) Final bit rate. d) Output signal to quantization noise ratio.

e) Number of bits per symbol. f) Symbol rate.

Q22/ A compact disk (CD) records audio signals digitally by using 32- level PCM. Assume the audio signal bandwidth to be 15 KHz.

a. What is Nyquist rate?
b. If the Nnyquist samples are quantized into L=65536 levels and then binary coded, determine the number of binary digits required to encode a sample.
c. Determine the bit rate and symbol rate at Nyquist rate.
d. For practical reasons, the signals are sampled at rate above Nequist rate at 44100 samples per second. If L=65536, determine number of bits per samples required to encode the signal and transmission bandwidth of encoded signal then find resulting bit rate and symbol rate.

Q23/A Delta modulator is used to encode speech signal band-limited to 6 KHz with sampling frequency 400 KHz. For b=±1 volt peak signal voltage, find Minimum step size to avoid slope overloading. Let m(t)=b* cos wmt.

Q24/ A delta modulator system is designed to operate at five times the nyquist rate for a signal with 3 KHz bandwidth. Determine the maximum amplitude of a 2 KHz input sinusoid for which the delta modulator does not have slope over load. Quantization step size is 250 mV. Let m(t)=b* cos wmt.

Q25/ A Delta Modulation (DM) system is designed to operate at 3 times the nyquist rate for a signal with a 3 KHz bandwidth. The quantization step size is 250 mV.

Let m(t)=b* $\sin wmt$.

a) Determine the maximum amplitude of a 1 KHz input sinusoid for which the delta modulator does not show slope overload.
b) Determine the post filtered output SNR for the signal of part (a).

Q26/ A Delta Modulation (DM) system is tested with a 10 KHz sinusoidal signal with a 1 V peek to peek at the input. It sampled at 10 times the nyquist rate. Let m(t)=b* $\cos wmt$.

a) What is the step size required to prevent slop overload?
b) What is the corresponding SNR?

Q27 and Q28/

➡ **Example 2.3.6 :** *A DM system is designed to operate at 3 times the nyquist rate for a signal with a 3 kHz bandwidth. The quantizing step size is 250 mV.*

(i) Determine the maximum amplitude of a 1 kHz input sinusoid for which the delta modulator does not show slope overload.

(ii) Determine the postfiltered output SNR for the signal of part (i).

[May/June-2004, 8 Marks]

Solution : Given data

Bandwidth, W = 3 kHz

Signal frequency, f_m = 1 kHz

$\therefore$ Nyquist rate $= 2f_m = 2\times 1\ kHz = 2\ kHz$

Sampling frequency, $f_s = 3\times nyquist\ rate$

$= 3\times 2\,kHz = 6$ kHz

Step size, δ = 250 mV

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

$$= \frac{1}{6000}$$

(i) To obtain signal amplitude

Slope overload distortion does not occur if,

$$A_m \leq \frac{\delta}{2\pi f_m T_s}$$

Putting values in above equation,

$$A_m \leq \frac{250 \times 10^{-3}}{2\pi \times 1000 \times \frac{1}{6000}}$$

$$\leq 0.2387 \text{ V.}$$

Thus maximum amplitude is 238.7 mV.

(ii) To obtain signal to noise ratio

Signal to noise ratio of delta modulator is given by equation 2.3.12 as,

$$\frac{S}{N} = \frac{3}{8\pi^2 W f_m^2 T_s^3}$$

$$= \frac{3}{8\pi^2 \times 3000 \times (1000)^2 \times \frac{1}{(6000)^3}}$$

$$= 4.56 \times 10^{-4} = -33.41 \text{ dB}$$

Here note that signal to noise ratio is very poor since sampling frequency is low.

➠ **Example 2.3.7 :** *A DM system is tested with a 10 kHz sinusoidal signal with 1V peak to peak at the input. It is sampled at 10 times the Nyquist rate.*

(i) What is the step size required to prevent slope overload ?

(ii) What is the corresponding SNR ?

Solution : Given data

Signal frequency, $f_m = 10$ kHz

Signal amplitude, $A_m = \frac{1V}{2} = 0.5V$

Nyquist rate $= 2 \times f_m = 2 \times 10 kHz = 20\ kHz$

Sampling frequency, $f_s = 10 \times \text{Nyquist rate}$

$= 10 \times 20 kHz = 200 kHz$

(i) To obtain step size

From equation 2.3.4 we have,

$$A_m \leq \frac{\delta}{2\pi f_m T_s}$$

Under this condition slope overload will not occur. From above equation step size will be,

$$\delta \geq 2\pi f_m T_s A_m$$

Putting values in above equation,

$$\delta \geq 2\pi \times 10{,}000 \times \frac{1}{200 \times 10^3} \times 0.5$$

$$\geq 0.157 \text{ V}$$

Thus the step size greater than 157 mV will prevent the slope overload.

(ii) To obtain signal to noise ratio

Signal to noise ratio of delta modulation system is given by equation 2.3.12 as,

$$\frac{S}{N} = \frac{3}{8\pi^2 W f_m^2 T_s^3}$$

$$\frac{S}{N} = \frac{3}{8\pi^2 W f_m^2 T_s^3}$$

This is post filtered signal to noise ratio. In this example value of 'W' is not given. Hence we will calculate signal to noise ratio from equation 2.3.7 and equation 2.3.10 as,

$$\frac{S}{N} = \frac{\frac{\delta^2}{8\pi^2 f_m^2 T_s^2}}{\frac{\delta^2}{3}}$$

$$= \frac{3}{8\pi^2 f_m^2 T_s^2}$$

Putting values in above equation.

$$\frac{S}{N} = \frac{3}{8\pi^2 \times (10{,}000)^2 \times \frac{1}{\left(200 \times 10^3\right)^2}}$$

$$= 15.2 = 11.8 \text{ dB.}$$