MUMBAI UNIVERSITY

Principles Of Communication Engineering

SEMESTER 4 MAY 2018 – Choice Based

Q.1 Solve the following.

a) Distinguish between narrowband and wideband FM. [4M]

Ans :

Narrowband FM	Wideband FM
1.Modulation index is less than 1	1.Modulation is greater than 1.
2.Frequency Deviation = 5kHz	2. Frequency Deviation =75kHz
3.Modulating frequency = 3kHz	3. Modulating frequency = 30Hz to 15kHz
4.Supression of noise is very less.	4.Noise is more suppressed.
5.Applications : FM mobile communication,short range ship to shore communication,etc	5.Applications: Entertainment broadcasting,high quality music transmission,etc

b) What is companding ? [4M]

Ans : i) Companding is the process of compression and then expansion.

ii) With companded system, the higher amplitude analog signals are compressed (amplified less than lower amplitude signals) prior to transmission and then expanded (amplified more than the lower amplitude signals) in the receiver.

A-law Companding Technique

iii) Uniform quantization is achieved at A = 1, where the characteristic curve is linear and no compression is done.

iv) A-law has mid-rise at the origin. Hence, it contains a non-zero value.

v) A-law companding is used for PCM telephone systems.

μ-law Companding Technique

vi) Uniform quantization is achieved at $\mu = 0$, where the characteristic curve is linear and no compression is done.

vii) μ-law has mid-tread at the origin. Hence, it contains a zero value.

viii) μ-law companding is used for speech and music signals.μ-law is used in North America and Japan.

c) Why AGC is required in AM receivers? **[4M]**

Ans :i) AGC is a departure from linearity in AM radio receivers.

ii) The AGC circuit keeps the receiver's output level from fluctuating too much by detecting the overall strength of the signal and automatically adjusting the gain of the receiver to maintain the output level within an acceptable range

iii) In simple AGC is a system which will change the overall gain of the receiver automatically, this is done in order to keep the receiver output constant even when the signal strength at the input of the receiver is changing.

iv) In AGC system a dc voltage (AGC bias) is derived from the detector. This AGC bias is thus proportional to the strength of received signal.

v) AGC bias is applied to a selected number of RF and IF amplifiers and mixer stage.

Vi) The transconductance and hence the gain of the devices connected in these stages is dependent on the applied AGC bias.

d) Explain aliasing error and aperture effect. **[4M]**

Ans : i) Aliasing effect take place when sampling frequency is less than Nyquist rate under such condition, the spectrum of the sampled signal overlaps with itself.

ii) Hence higher frequency components are called aliasing effect.

iii) The aliased signal will appear at a predictable frequency in the Fourier spectrum. For example, given a sampling frequency of 200Hz (Nyquist frequency = 100Hz), a digitized 101Hz signal will appear at 99Hz, while a 200Hz signal will appear at 0Hz or DC. A 201Hz signal will look like a 1Hz signal, and so on.

iv) In flat top sampling,due to the lengthening of the sample, amplitude distortion as well as adelay of T/2 was introduced.This distortion is referred to as Aperture effect.

v)Aperture effect can be corrected by connecting an equalizer in cascade with the low passreconstruction filter. This equalizer has the effect of decreasing the in-band loss of reconstructionfilter as the frequency increases in such a manner as to compensate for the aperture effect

e) Explain various types of noise affecting communication system. [4M]

Ans : i) Noise : Noise is an unwanted signal which interferes with the original message signal and corrupts the parameters of the message signal.

ii) Noise limits the operating range of the systems

Noise affects the sensitivity of receivers

iii) External Source

This noise is produced by the external sources which may occur in the medium or channel of communication, usually.

This noise cannot be completely eliminated. The best way is to avoid the noise from affecting the signal.

A) Atmospheric noise (due to irregularities in the atmosphere).
B) Extra-terrestrial noise, such as solar noise and cosmic noise.
C) Industrial noise

iv) Internal Source

This noise is produced by the receiver components while functioning. The components in the circuits, due to continuous functioning, may produce few types of noise. This noise is quantifiable.

A proper receiver design may lower the effect of this internal noise.

A) Thermal agitation noise (Johnson noise or Electrical noise).
B) Shot noise (due to the random movement of electrons and holes).
C) Transit-time noise (during transition).
D) flicker, resistance effect and mixer generated noise.

Q.2 a) What are the drawbacks of delta modulation?Explain adaptive delta moldulation in detail. [10M]

Ans :

A]Drawbacks of delta modulation :

1.Slope overload distortion :

i) This distortion arises because of large dynamic range of the input signal.

ii)The rate of rise of input signal x(t) is so high that the staircase signal can not approximate it, the step size 'Δ' becomes too small for staircase signal u(t) to follow the step segment of x(t).

iii) Hence, there is a large error between the staircase approximated signal and the original input signal x(t).This error is called as slope overload distortion.

2.Granular Noise :

iv) Granular or Idle noise occurs when the step size is too large compared to small variation in the input signal.

v) This means that for very small variations in the input signal, the staircase signal is changed by large amount (Δ) because of large step size.

vi) When the input signal is almost flat , the staircase signal u(t) keeps on oscillating by ±Δ around the signal.

vii) The error between the input and approximated signal is called granular noise.

B] Adaptive Delta Modulation :

i) Adaptive delta modulation technique is used to avoid drawbacks of delta modulation i.e granular noise and slope overload distortion .

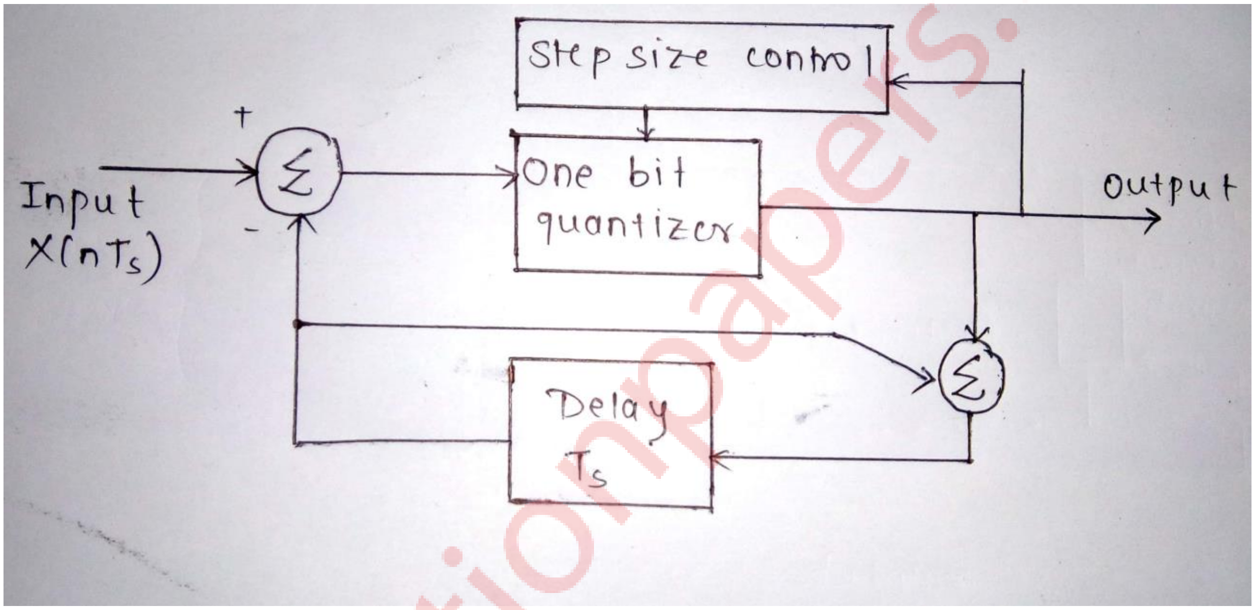
ii) This Modulation method is similar to Delta modulation except that the step size is variable according to the input signal in Adaptive Delta Modulation whereas it is a fixed value in delta modulation.

iii) In Adaptive Delta Modulation, the step size of the staircase signal is not fixed and changes depending upon the input signal.

iv) Here first the difference between the present sample value and previous approximation is calculated.

v) This error is quantized i.e. if the present sample is smaller than the previous approximation, quantized value is high or else it is low. The output of the one-bit quantizer is given to the Logic step size control circuit where the step size is decided.

vi) The transmitter circuit consists of a summer, quantizer, Delay circuit, and a logic circuit for step size control.



vii)The baseband signal X(nTs) is given as input to the circuit. The feedback circuit present in the transmitter is an Integrator. The integrator generates the staircase approximation of the previous sample.

viii) At the summer circuit, the difference between the present sample and staircase approximation of previous sample e(nTs) is calculated.

ix)This error signal is passed to the quantizer, where a quantized value is generated. The step size control block controls the step size of the next approximation based on either the quantized value is high or low. The quantized signal is given as output.

x) At the receiver end Demodulation takes place. The receiver has two parts. First part is the step size control. Here the received signal is passed through a logic step size control block, where the step size is produced from each incoming bit.

xi) Step size is decided based on present and previous input. In the second part of the receiver, the accumulator circuit recreates the staircase signal.

xii)This waveform is then applied to a low pass filter which smoothens the waveform and recreates the original signal.

b) What is signal multiplexing? Explain TDM & FDM in detail. [10M]

Ans :

A. Signal Multiplexing :

-Signal multiplexing is a process in which multiple signals can be transmitted over the same communication medium simultaneously.

-If the analog signals are multiplexed, then it is called as analog multiplexing. Similarly, if the digital signals are multiplexed, then it is called as digital multiplexing.

-A number of signals were combined to send through a single cable. The process of multiplexing divides a communication channel into several number of logical channels, allotting each one for a different message signal or a data stream to be transferred. The device that does multiplexing can be called as Multiplexer or MUX.

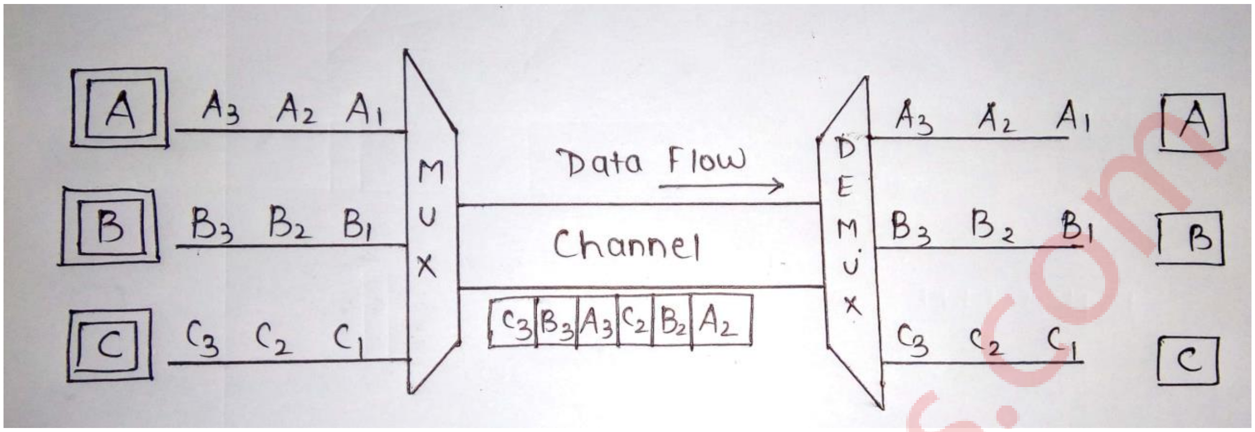
B. TDM – Time Division Multiplexing :-

-Time division multiplexing (FDM) is a technique of multiplexing, where the users are allowed the total available bandwidth on time sharing basis. Here the time domain is divided into several recurrent slots of fixed length, and each signal is allotted a time slot.

- TDM is digital multiplexing technique. In TDM, the channel/link is not divided on the basis of frequency but on the basis of time.

-Total time available in the channel is divided between several users.

-Each user is allotted a particular a time interval called time slot or time slice during which the data is transmitted by that user.



-Thus each sending device takes control of entire bandwidth of the channel for fixed amount of time.

-In TDM the data rate capacity of the transmission medium should be greater than the data rate required by sending or receiving devices.

-Thus each signal will be transmitted for a very short time. One cycle or frame is said to be complete when all the signals are transmitted once on the transmission channel.

-The TDM system can be used to multiplex analog or digital signals, however it is more suitable for the digital signal multiplexing. The TDM signal in the form of frames is transmitted on the common communication medium.

Advantages of TDM :
1. Full available channel bandwidth can be utilized for each channel.
2. Intermodulation distortion is absent.
3. TDM circuitry is not very complex.
4. The problem of crosstalk is not severe.

Disadvantages of TDM :

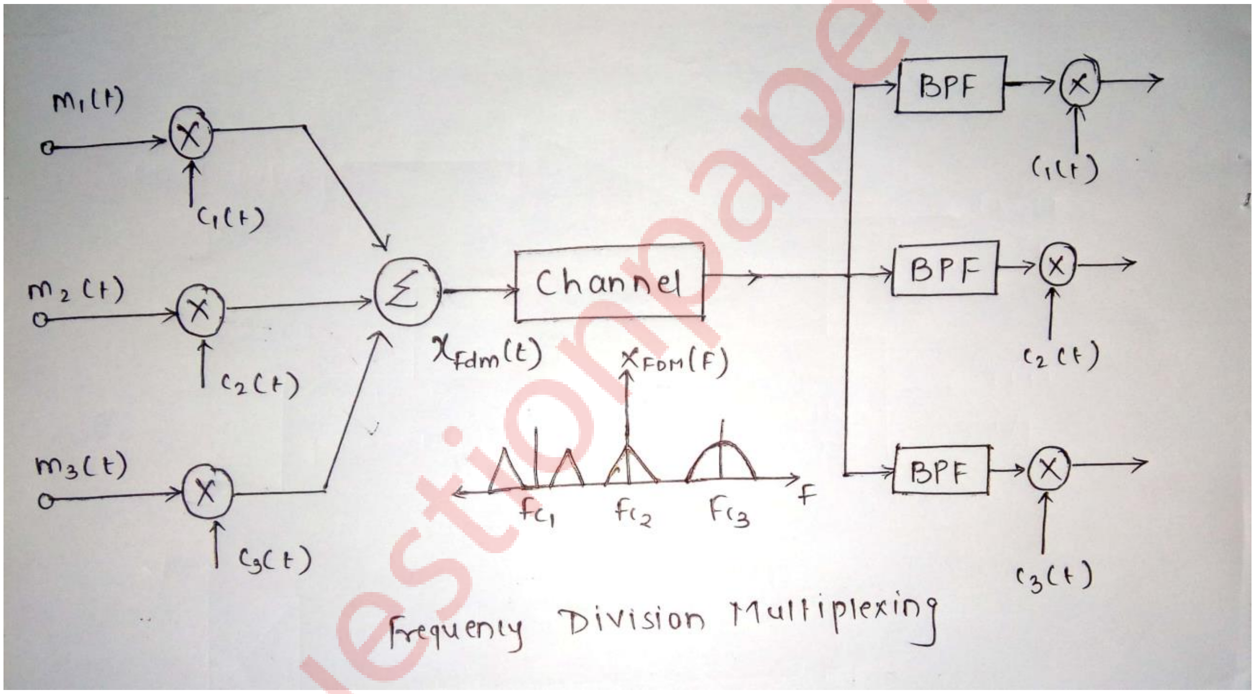
1. Synchronization is essential for proper operation.
2. Due to slow narrowband fading, all the TDM channels may get wiped out.

C. FDM – Frequency Division Multiplexing :

-Frequency-Division Multiplexing (FDM) is a scheme in which numerous signals are combined for transmission on a single communications line or channel.

-It is analog multiplexing technique. Each signal is assigned a different frequency (sub channel) within the main channel. its requires channel synchronization. FDM multiplexing technique is based on orthogonality of sinusoids.

-FDM requires that the bandwidth of a link should be greater than the combined bandwidths of the various signals to be transmitted. Thus each signal having different frequency forms a particular logical channel on the link and follows this channel only.



-These channels are then separated by the strips of unused bandwidth called guard bands. These guard bands prevent the signals from overlapping

-A typical analog connection via a twisted pair telephone line requires approximately three kilohertz (3 kHz) of bandwidth for accurate and reliable data transfer.

-Twisted-pair lines are common in households and small businesses. But major telephone cables, operating between large businesses, government agencies, and municipalities, are capable of much larger bandwidths

Advantages of FDM:

1. A large number of signals (channels) can be transmitted simultaneously.

2. FDM does not need synchronization between its transmitter and receiver for proper operation.

3. Demodulation of FDM is easy.

4. Due to slow narrow band fading only a single channel gets affected.

Disadvantages of FDM:

1. The communication channel must have a very large bandwidth.

2. Intermodulation distortion takes place.

3. Large number of modulators and filters are required.

4. FDM suffers from the problem of crosstalk.

5. All the FDM channels get affected due to wideband fading.

Q.3 a) State and prove sampling theorem for low pass bandlimited signal. [10M]

Ans :

i) A continuous time signal can be represented in its samples and can be recovered back when sampling frequency fs is greater than or equal to the twice the highest frequency component of message signal. i. e.

$$f_s \geq 2f_m$$

ii) Consider a continuous time signal x(t). The spectrum of x(t) is a band limited to fm Hz i.e. the spectrum of x(t) is zero for $|\omega|>\omega_m$.

iii) Sampling of input signal x(t) can be obtained by multiplying x(t) with an impulse train δ(t) of period Ts. The output of multiplier is a discrete signal called sampled signal which is represented with y(t) .

iv) The process of sampling can be explained by the following mathematical expression:

Sampled signaly(t)=x(t).δ(t)......(1)

Sampled signal y(t)=x(t).δ(t) ......(1)

v) The trigonometric Fourier series representation of δδ(t) is given by

$$\delta(t) = a0 + \sum_{n=1}^{\infty}(ancosn\omega st + bnsinn\omega st) \quad ......(2)$$

Where $a0 = 1/Ts \int_{-T/2}^{T/2} \delta(t)dt = 1Ts\delta(0) = 1/Ts$

$an = 1/Ts \int_{-T/2}^{T/2} \delta(t)\cos(n\omega_s)\, dt = 2/T$

$bn = 1/Ts \int_{-T/2}^{T/2} \delta(t)\sin(n\omega_s)dt = 0$

$$\delta(t) = 1/T + \sum_{n=1}^{\infty}\left(\frac{2}{T}\right).\cos(n\omega_s t)$$

But we know that

y(t)=x(t).δ(t)

$$\therefore y(t) = x(t).[\,1/T + \sum_{n=1}^{\infty}\left(\frac{2}{T}\right).\cos(n\omega_s t)]$$

$$\therefore y(t) = \frac{1}{T}[\,x(t) + \sum_{n=1}^{\infty}(2.x(t)).\cos(n\omega_s t)]$$

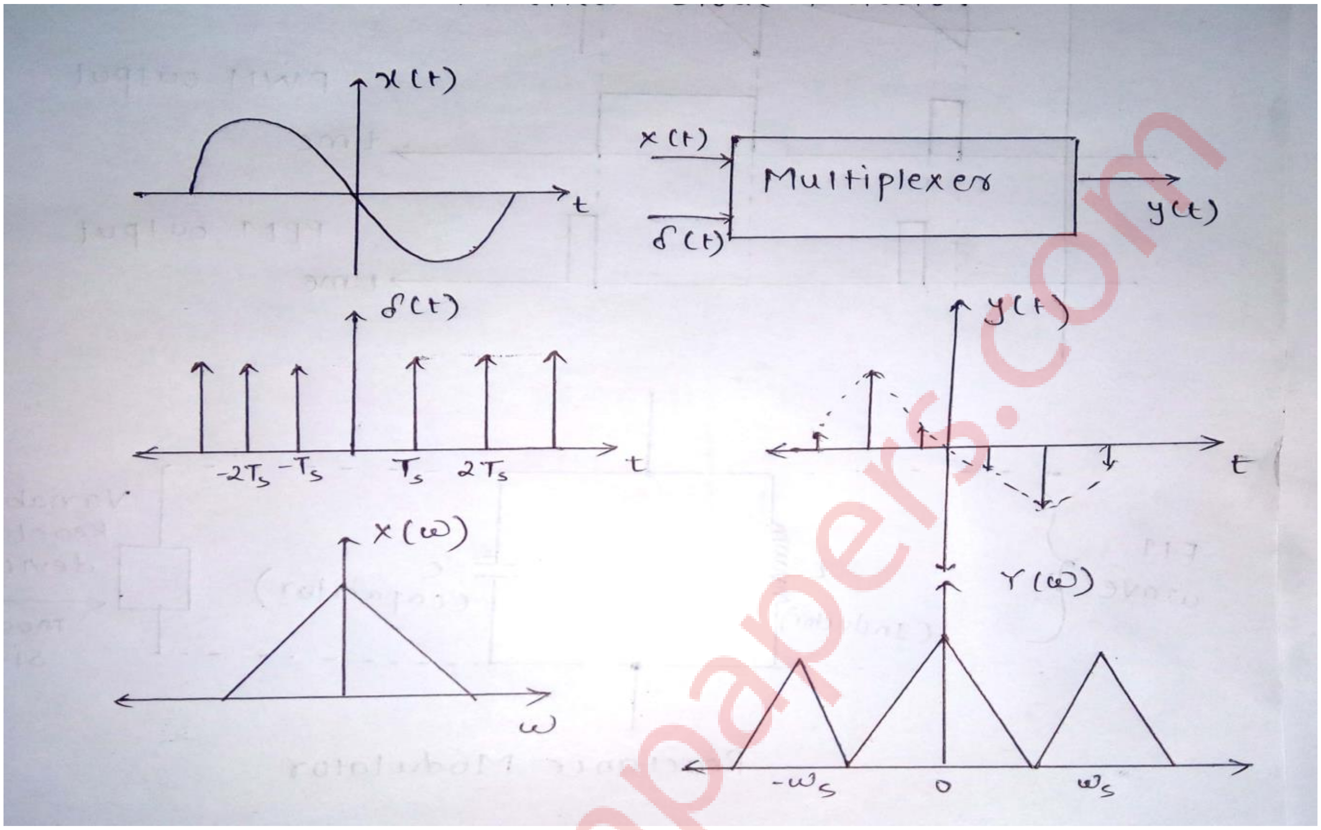
$$\therefore y(t) = \frac{1}{T}[\,x(t) + 2.\cos(\omega_s t).x(t) + 2.\cos(2\omega_s t).x(t) + 2.\cos(3\omega_s t).x(t) + ...]$$

Taking fourier transform ,

$$\therefore Y(w) = \frac{1}{T}[\,X(w) + X(w - ws) + X(w + ws) + \cdots]$$

$$\boxed{\therefore Y(w) = \frac{1}{T}[\,\sum_{-\infty}^{\infty} X(w - n.ws)]}$$

Where n=0,±1,±2,...



b) Explain practical diode detector with suitable diagrams. [10M]

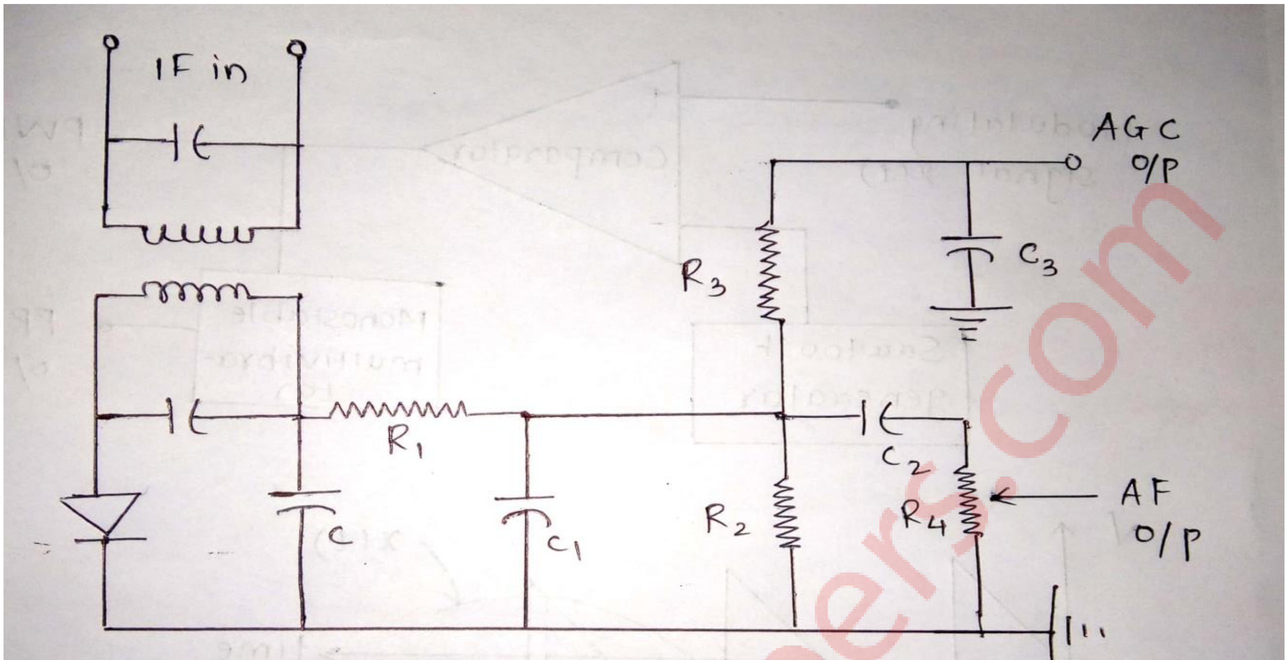
Ans :

i)A detector is used in receiver circuits to recognize the presence of signals. Typically a diode or similar device is used as a detector.

ii)The diode detector is the simplest and most basic form of amplitude modulation, AM signal detector and it detects the envelope of the AM signal.

iii)A number of additions have been made to the Simple Diode Detector, and its practical version.

iv) The circuit operates in the following manner. The diode has been reversed, so that now the negative envelope is demodulated. This has no effect on detection, but it does ensure that a negative AGC voltage will be available.



Practical Diode Detector.

v)The resistor R of the basic circuit has been split into two parts (R1 and R2) to ensure that there is a series dc path to ground for the diode, but at the same time a low-pass filter has been added, in the form of R1 – C1. This has the function of removing any RF ripple that might still be present.

vi)Capacitor C2 is a coupling capacitor, whose main function is to prevent the diode dc output from reaching the volume control R4.

vii)Although it is not necessary to have the volume control immediately after the detector, that is a convenient place for it.

viii)The combination R3 – C3 is a low-pass filter designed to remove AF components, providing a dc voltage whose amplitude is proportional to the carrier strength, and which may be used for automatic gain control.

ix)It can be seen that the dc diode load is equal to R1 + R2, whereas the audio load impedance Zm is equal to R1 in series with the parallel combination of R2, R3 and R4, assuming that the capacitors have reactances which may be ignored.

x)This will be true at medium frequencies, but at high and low audio frequencies Zm may have a reactive component, causing a phase shift and distortion as well as an uneven frequency response.

xi) Advantages : low cost,easy to implement,does not require any setup.

Disadvantages : Distortion,selective fading,less sensitivity.

Q.4 a) What are different methods of FM generation?Explain reactance modulator in detail. [10M]

Ans :

1. Different methods of FM generation :

[A] Direct Method :

-- Reactance modulator

-- Varactor diode modulator

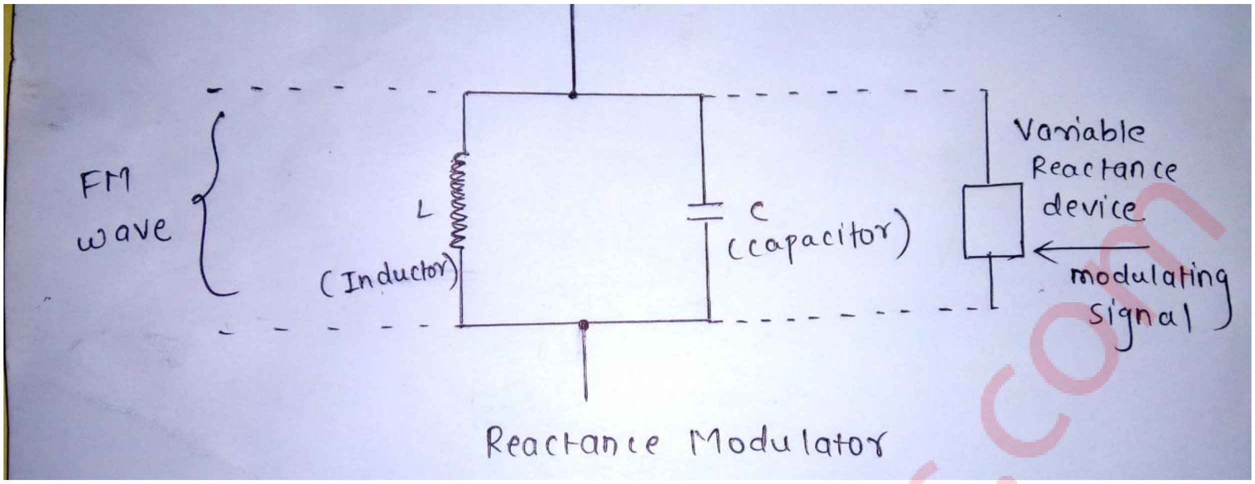
[B] Indirect Method

-- Armstrong method

2.Reactance Modulator : -

i)A reactance modulator changes the frequency of the tank circuit of the oscillator by changing its reactance.

ii) This is accomplished by a combination of a resistor, a condenser, and a vacuum tube (the modulator) connected across the tank circuit of the oscillator and so adjusted as to act as a variable inductance or capacitance.



iii)The net result is to change the resonant frequency of the LC circuit by amounts proportional to the instantaneous a.f. voltages applied to the grid of the modulator tube, without changing the resistance of the LC circuit or the amplitude of the oscillations.

iv)The voltages supplied to both the modulator and oscillator must be carefully stabilized to prevent undesired frequency changes.

v)The speech does not have to deliver any power and need supply only a small output voltage, say 10 or 15 volts.

vi)A diode, R-C coupled, will be sufficient even with a sensitive microphone and a high-powered oscillator. The frequency change of LC per volt change on the a.f. grid of the modulator tube will be greater when C1, , is made smaller. The blocking condenser C2 has a comparatively high value, and hence offers but small reactance to r.f. currents.

vii)The radio-frequency voltages which are developed across the tank in the oscillator circuit also appear across the RC1 circuit and across the parallel 6L7 modulator tube.

viii) The resistance r has been replaced by the internal resistance of the modulator tube .

ix) The voltage drop across C1 is 90° out of phase with the tank voltage. It is applied to the control grid of the 6L7 whose r.f. plate current responds in the same phase. Thus this current is made to lag 90° behind the tank voltage.

x) The r.f. plate current flows through the tank circuit and, combined with the current therein, is equivalent to a new current whose phase differs from the normal value just as though an additional reactance (not resistance) had been connected in with L and C. This, of course, changes the frequency of the LC circuit and hence of the transmitter. When a.f. is fed into the modulator tube, it causes proportionate changes in the r.f. plate current and hence in the equivalent reactance of the LC circuit.

b) Explain how PPM is generated using PWM ? [10M]

Ans :

i) Pulse position modulation : The amplitude and the width of the pulse remains constant. The time when the pulse occurs is varied in accordance with the modulating signal.

ii)Pulse Width Modulation : The amplitude of the pulse is maintained constant but the width of each pulse is varied according to the modulating signal.

iii)Saw tooth generator: – The saw tooth generator is conneted to the inverting terminal of the operational amplifier (Opamp). The Op-amp is used in comparator mode.

iv)Modulating signal: – The modulating signal is given as input to the non inverting terminal of the Op-amp as comparator.

v)PWM & PPM generation PWM generation: – The output of the comparator is zero except when modulating signal waveform exceeds the sawtooth wave, when the output is high.

vi) PPM generation: – The output of the comparator (i.e. PWM) is given as input to negative edge triggered monostable pulse generator. – On the trailing edge (negative going edge) of the PWM signal produces a short pulse of fixed duration. This output is PPM signal.

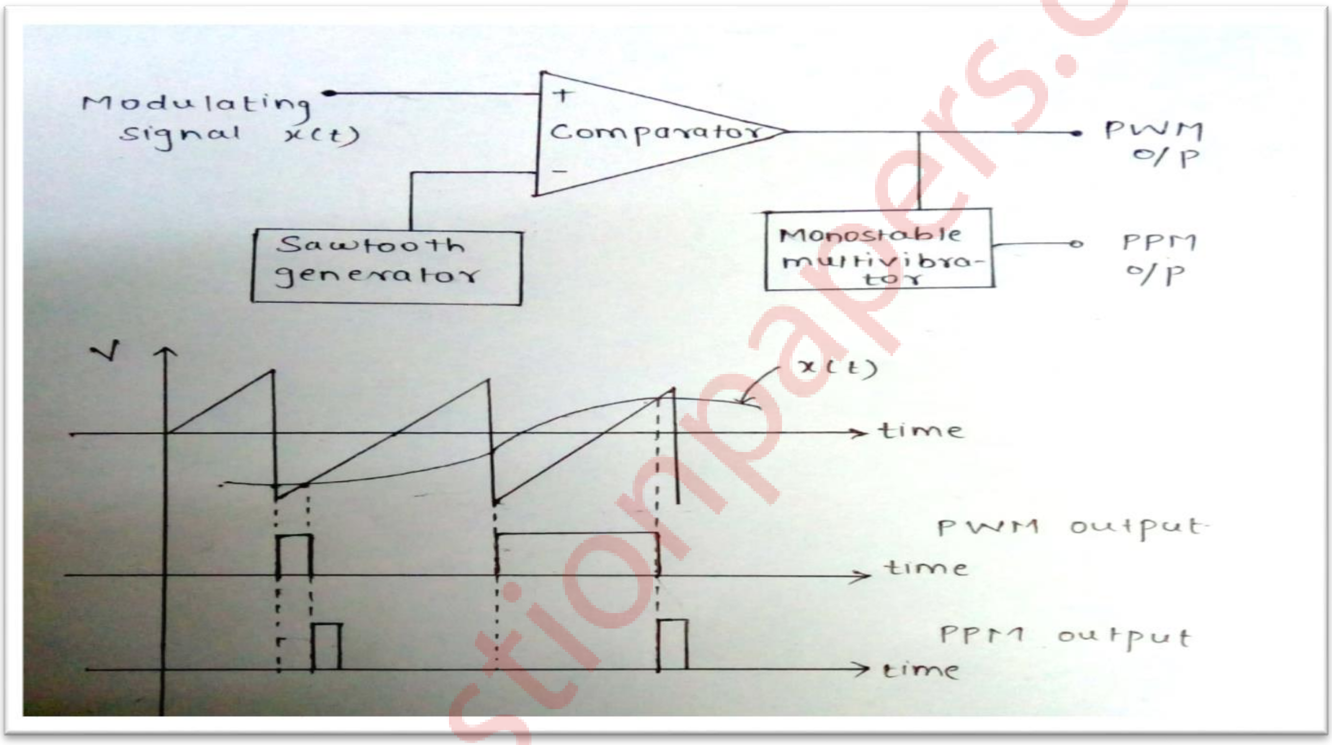
vii) Advantage: – Less noise interference due to constant amplitude. – Signal and noise separation is very easy. – Due to constant pulse width & amplitudes, transmission power for each pulse is same.

vii) Disadvantage – Synchronization between transmitter & receiver is required.

viii) Applications of PPM :

R/C transmitters,R/C receivers ,Autopilot/Stabilization system,PCTx

ix) PPM and PWM are used to send analog signals, not digital signals. They are analog protocols.



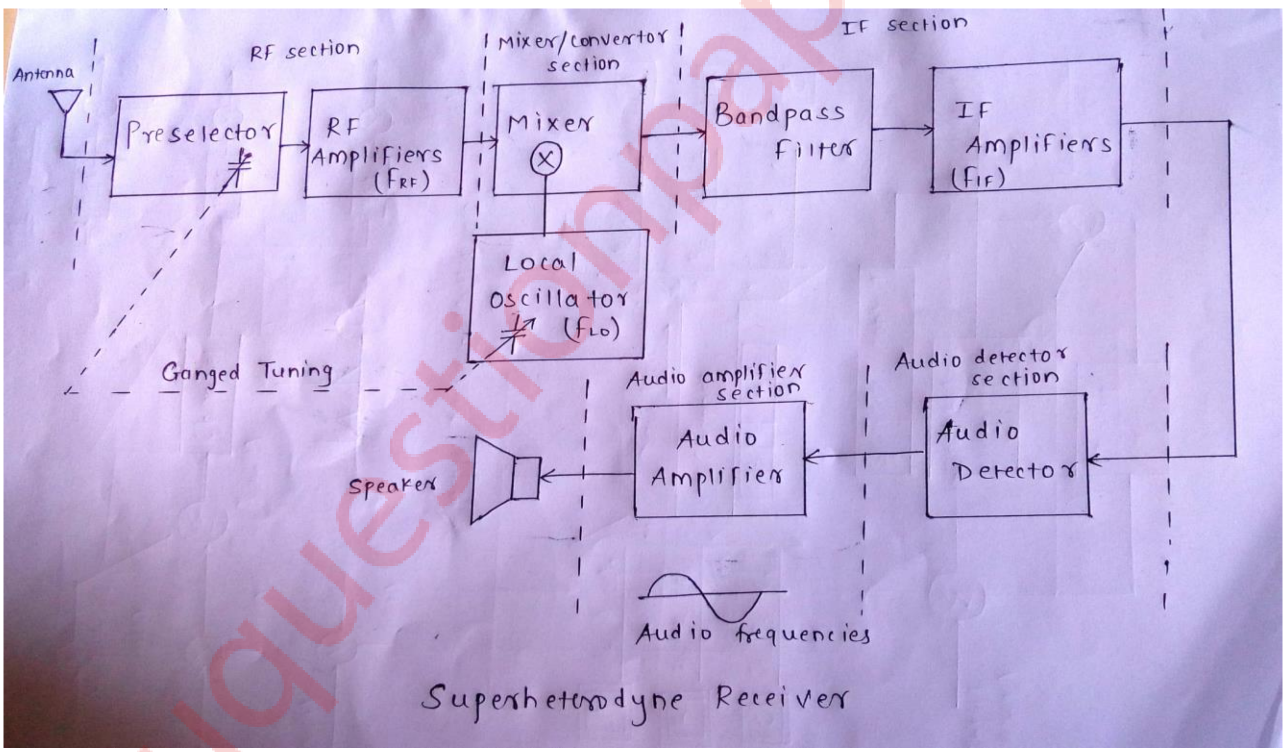
x) Role of monostable multivibrator : When triggered , a pulse of of predefined duration is produced.The circuit then returns to its stable state and produces no more output until triggered.In PPM generation the pulse is generated whenever the PWM pulse is high. And the duration of PPM pulse generated by monostable multivibrator is constant for each triggered pulse.

Q.5 a) Explain superheterodyne receiver . [10M]

Ans :

i)Heterodyne means to mix two frequencies together in a nonlinear device or to translate one frequency to another using nonlinear mixing. Essentially, there are five sections to a superheterodyne receiver: RF section, mixer/convertor section, IF section, detector section and the audio amplifier section.

ii)RF Section – The RF Section generally consists of a preselector and an amplifier stage. The preselector is a broad-tuned band pass filter with an adjustable center frequency used to reject unwanted radio frequency and to reduce the noise bandwidth. The RF amplifier amplifies the signal and also determines the sensitivity of the receiver.



iii)Mixer/ Converter Section – It consists of a radio-frequency oscillator and a mixer. The choice of oscillator depends on the stability and accuracy desired. Mixer/converter is a non-linear device that is used to convert radio frequency to intermediate frequencies. Although the frequencies are changed, the shape of the envelope, the bandwidth and the original information contained in the envelope remains unchanged.

iv)IF Section – It consists of a series of bandpass filters and IF amplifiers. Most of the receiver gain and selectivity is achieved in the IF section. The IF is always lower than the RF because it is easier and less expensive to construct high-gain, stable amplifiers for low frequency signals. IF amplifiers are also less likely to oscillate when compared to RF amplifiers.

v)Detector Section – It converts the IF signals back to the original source information. The detector can be as simple as a single diode or as complex as a PLL or balanced demodulator.

vi)Audio amplifier section – It amplifies the information signal to a required level. It comprises of several cascaded audio amplifiers and one or more speakers. The number of amplifiers depends on the audio signal power desired.

Advantages :

1. As high frequency is down converted to IF, there is no possibility of oscillations.

2. The bandwidth remains constant over the entire frequency range.

3. Better selectivity as no adjacent channels are picked due to variation in bandwidth.

b) Explain VSB transmission. [10M]

Ans :

i)Vestigial sideband (VSB) is a type of amplitude modulation (AM) technique (sometimes called VSB-AM) that encodes data by varying the amplitude of a single carrier frequency .

ii)Portions of one of the redundant sidebands are removed to form a vestigial sideband signal - so-called because a vestige of the sideband remains.

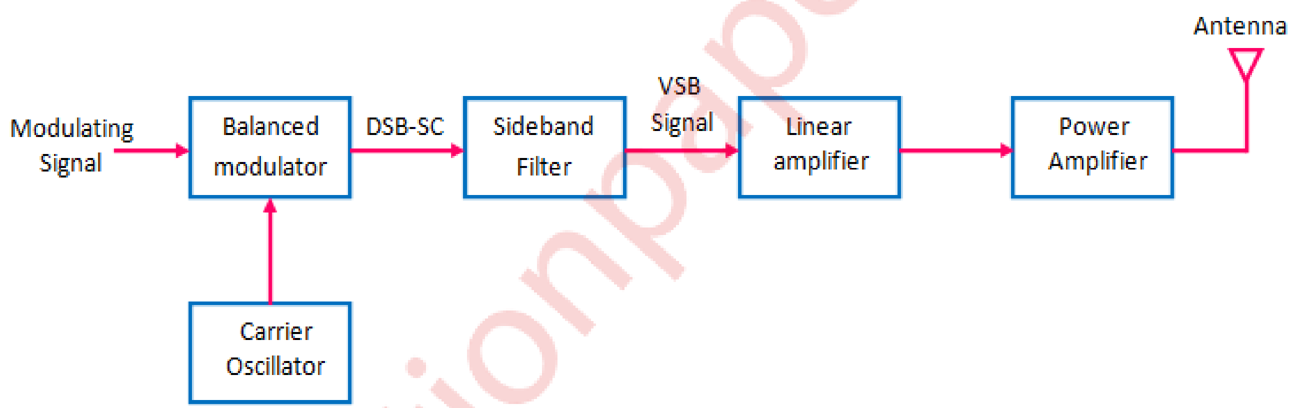
iii)In AM, the carrier itself does not fluctuate in amplitude. Instead, the modulating data appears in the form of signal components at frequencies slightly higher and lower than that of the carrier.

iv)These components are called sidebands . The lower sideband (LSB) appears at frequencies below the carrier frequency; the upper sideband (USB) appears at frequencies above the carrier frequency.

v)The actual information is transmitted in the sidebands, rather than the carrier; both sidebands carry the same information.

vi)Because LSB and USB are essentially mirror images of each other, one can be discarded or used for a second channel or for diagnostic purposes.

vii)VSB transmission is similar to single-sideband (SSB) transmission, in which one of the sidebands is completely removed. In VSB transmission, however, the second sideband is not completely removed, but is filtered to remove all but the desired range of frequencies.



viii) Advantages :-

--Highly efficient.

--Reduction in bandwidth.

--Filter design is easy as high accuracy is not needed.

--The transmission of low frequency components is possible, without difficulty.

--Possesses good phase characteristics.

ix)Disadvantages

--Bandwidth when compared to SSB is greater.

--Demodulation is complex.

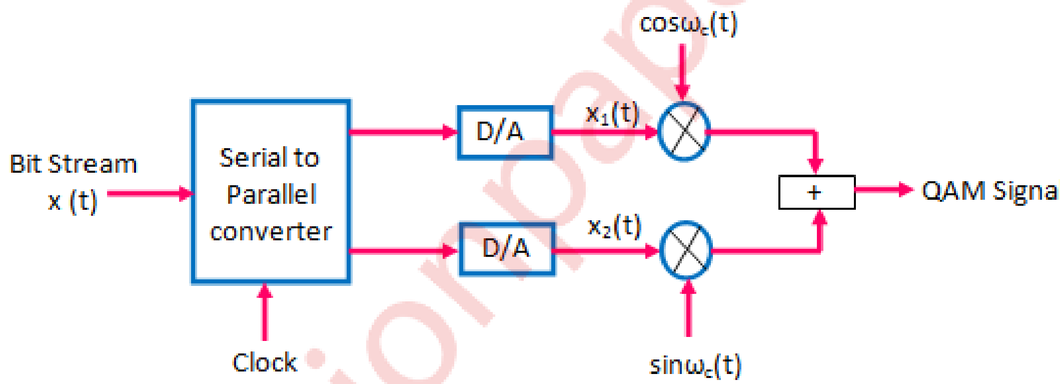
Q.6 Write Short note on - [20M]

1.Quadrature amplitude modulation :

i)Quadrature Amplitude Modulation, QAM is a signal in which two carriers shifted in phase by 90 degrees (i.e. sine and cosine) are modulated and combined.

ii)As a result of their 90° phase difference they are in quadrature and this gives rise to the name. Often one signal is called the In-phase or "I" signal, and the other is the quadrature or "Q" signal.

iii)The resultant overall signal consisting of the combination of both I and Q carriers contains of both amplitude and phase variations.



iv)In view of the fact that both amplitude and phase variations are present it may also be considered as a mixture of amplitude and phase modulation.

v)Use of quadrature amplitude modulation comes from the fact that a straight amplitude modulated signal, i.e. double sideband even with a suppressed carrier occupies twice the bandwidth of the modulating signal. This is very wasteful of the available frequency spectrum. QAM restores the balance by placing two independent double sideband suppressed carrier signals in the same spectrum as one ordinary double sideband supressed carrier signal.

2.Amplitude limiting and threshoulding : -

i)Amplitude limiting is “a process in which the amplitude of output signal is limited to a desired level or margin irrespective of the variations in the input signal”.

ii) Amplitude limiter is an electronic device which clips (removes) the amplitude of output signals to a desired margin irrespective of variations in the input signal

iii)The undesirable input amplitude is clipped by a limiter circuit and gives the desirable margin of output.

iv)Amplitude limiters are used in FM (Frequency modulation) receivers to eliminate the undesirable amplitude changes caused by noise.

v)Threshould effect is defined as the value of input signal to noise ratio below which the output signal to noise ratio decreases much rapidly than the input signal to noise ratio. It is the property of envelope detectors used for the demodulation of modulated signals.

vi) It occurs due to presence of large noise and therefore causes loss in the message signal. When the noise is very large as compared to the input at envelope detector, the message signal at the output is mixed with noise.

3.Double spotting : -

i)Double-spotting" is a term that means that the wanted station is tuned in at two spots on the dial. These spots would be just 60kHz apart if an IF of 30kHz is used.

ii) In a superheterodyne receiver, the local oscillator frequency is offset from the wanted station by the frequency of the IF amplifierFor example the wanted station is on 800kHz and the IF is 30kHz. This means that the local oscillator (which is usually higher in frequency than the tuned station) will be on 800 + 30 = 830kHz.

iii)However, if the selectivity of the RF stage is quite poor, a station on 860kHz will also give a 30kHz IF output when mixed with the local oscillator (on 830kHz).

iv) As a result, two stations - one on 800kHz and one on 860kHz - will be received at the same time. If the receiver is now tuned to 740kHz the oscillator will be on

770kHz. However, this will also give a 30kHz IF output from the 800kHz station. This means that the 800kHz station is heard at both the 800kHz and 740kHz positions on the dial.

4.Low level and high level modulation :-

A. HIGH LEVEL MODULATION:

1. Power level: modulation takes place at high modulation level.
2. Types of amplifier: highly efficient class c amplifiers are used.
3. Efficiency: very high.
4.Devices used: vacuum tubes or transistor for medium power applications.
5. Design of AF power amplifier: complex due to very high power involved.
6.Applications: high power broadcast transmitters.

B.LOW LEVEL MODULATION:

1. Power level: modulation takes place at low power level.
2. Types of amplifier: linear amplifiers are used after modulation
3. Efficiency: lower than high level modulators.
4. Devices used: transistors, JFET,OP-AMPs.
5. Design of AF amplifier: easy as it is to be done at low power.
6.Applications: sometimes used in tv transmitters(IF modulation).

5.PCM and DPCM :-

Parameters	PCM :Pulse Code Modulation	DPCM :-Differential Pulse Code Modulation
Number of bits	It uses 4, 8, or 16 bits per sample	Less than PCM
Levels, step size	Fixed step size. Cannot varied	A fixed number of levels are used.
Bit redundancy	Present	Can permanently remove
Quantization error and distortion	Depends on the number of levels used	Slope overload distortion and quantization noise are present but very less as compared to PCM
The bandwidth of the transmission channel	Higher bandwidth has been required since the number of bits is absent	Lower than PCM bandwidth
Feedback	No feedback in Tx and Rx	Feedback exists
Signal to noise ratio (SNR)	HIGH	LOW