CSE 509 17-02-20

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Course contents:

Chapter 01 → Introduction to communication and signal.

Chapter 02 → Continuous wave modulation (AM, FM, PM) + power.

Chapter 03 → Voice digitization and Transmission (Sampling, Nyquist theorem), PAM, PPM, PWM, PCM, SNR, Companding, Delta modulation, Differential PCM.

Chapter 04 → Bandwidth utilization (multiplexing) FDM, TDM, WDM

Chapter 05 → Multiple Access Techniques. FDMA, TDMA, CDMA, SDMA

Chapter 06 → Digital Modulation and information theory. Line codes. ASK, PSK, FSK, QAM, GMSK.

Reference books:

① BP Lathi → 3rd edition.
② Simon Hayking → Digital communication.

Chapter 01 :

Significance of human comunication:

① Communication is the process of exchanging information.

② main barriers are language and distance,

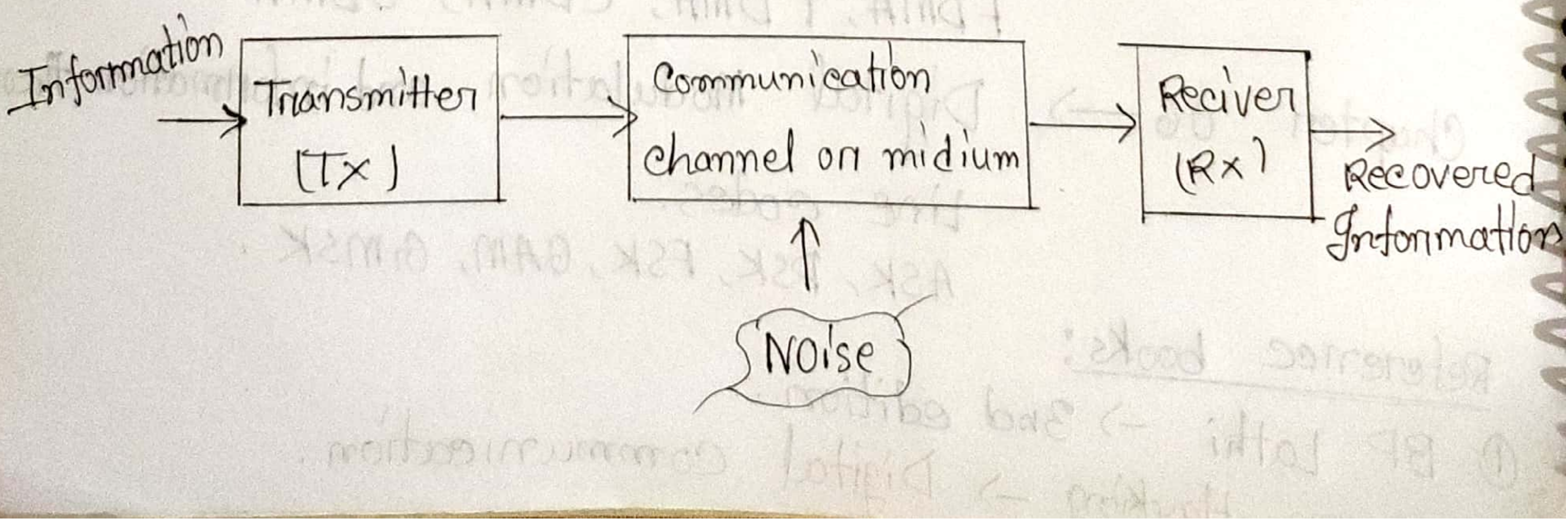
Method of Communication:

① face to face
② Signals
③ Written words (Letters)
④ Electrical inovation → Telegraph, Telephone, Radio, TV Internet.

Communication System:

Basic component : ⓐ Transmitter.
ⓑ Receiver.
ⓒ Channel or medium.

Block Diagram:



Transmitter: It is a collection of electronic component and circuits that converts electrical signals into a signal suitable for transmission over a given medium. It is made up of oscillators, amplifiers, tuned circuits, filter, modulators and other circuits.

Communication channel: It is the medium by which the electronic signals is sent from one place to another.

Types are: Electrical conductors, optical media, free space.

Reciver: It is a collection of electronic components and circuits that accepts the transmitted signal from the channel and converts it back to a form understandable by human. It contains amplifier oscillators, mixers or detector.

Transciever: Combination of Tx and Rx. It is an electronic unit that incorporates circuits that both send and receive signals.

Example: Telephone, Fax, mobile etc.

☐ Describe the comunication system with proper explanation?

Ans: Block diagram and explanation of all parts.

☐ Noise: Noise is a random, undesirable electronic energy that enters the communicating medium and interfaces with transmitted message.

Signal and Noise Ratio: The signal to noise ratio indicates the relative strengths of the signal and the noise in a communication System.

$$SNR = \frac{\text{Signal } P(\omega)}{\text{Noise } P(\omega)}$$

$$SNR_{dB} = 10 \log_{10} SNR$$

Intensity

Sources of Noise:

① External Noise: It comes from sources over which we have little or no control such as-

ⓐ Industrial sources: Motor, generators etc.

ⓑ Atmospheric: Naturally occuring electrical disturbances in the earth atmosphere.

ⓒ Space: The sun radiates a wide range of signal in a broad noise spectrum.

② Internal Noise: ⓐ Thermal noise.
ⓑ Semi conductor.
ⓒ Inter modulation distortion.

Types of electronic communication:

ⓐ Direction of communication
↳ Simplex / Duplex

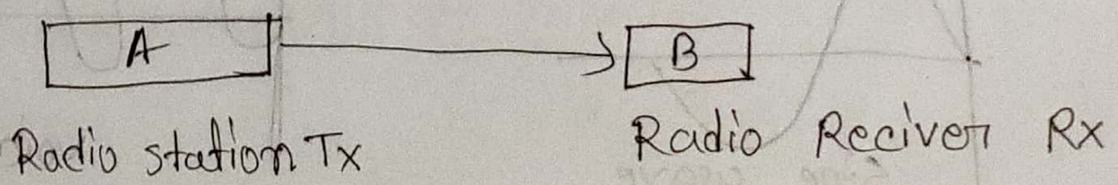
ⓑ Types of transmitted signal.
↳ Analog / Digital

ⓒ Use of modulation
↳ Base band / Broad band.

Direction of communication:

ⓐ Simplex: Simplest method of electronic communicatio[n] is referred to as simplex. This type of communi-cation is one way. (Single direction)

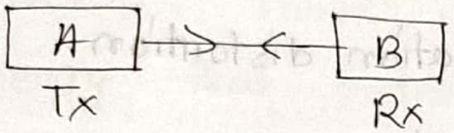
Example: Radio, TV etc



(b) Duplex: The type of communication is two way.

Duplex is 2 types.

1. Full duplex.
2. Half duplex



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(1) Full Duplex: If both transmit and receive works simultaneously. Example → Telephone.

(2) Half Duplex: The two way communication in which only one partially transmits at a time.

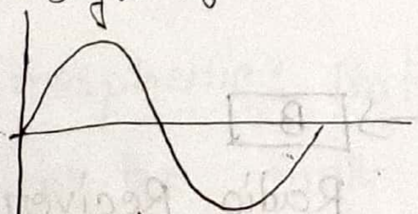
Example: Police, military, radio transmission etc.

Types of transmitted signal:

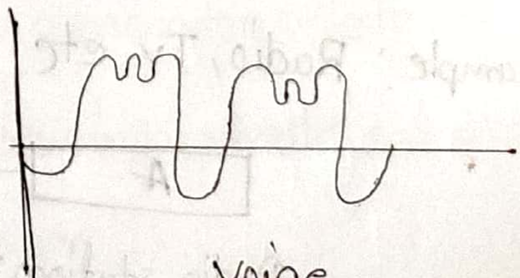
Analog signal: An analog signal is smoothly and continuously varying voltage or current.

Ex: Sine wave, voice etc.

~~Digital signal:~~

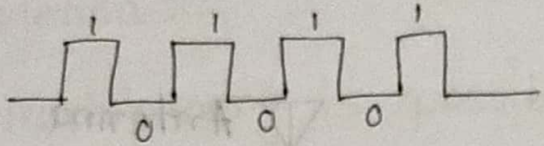


Sine wave



Voice

Digital Signal: Digital signals change in step on in discrete increments



Use of modulation:

Baseband Transmission: In this type of transmission the signal is unmodified and not modulated.

Ex: Telephone, Intercom.

Problem: For long distance not usefull.

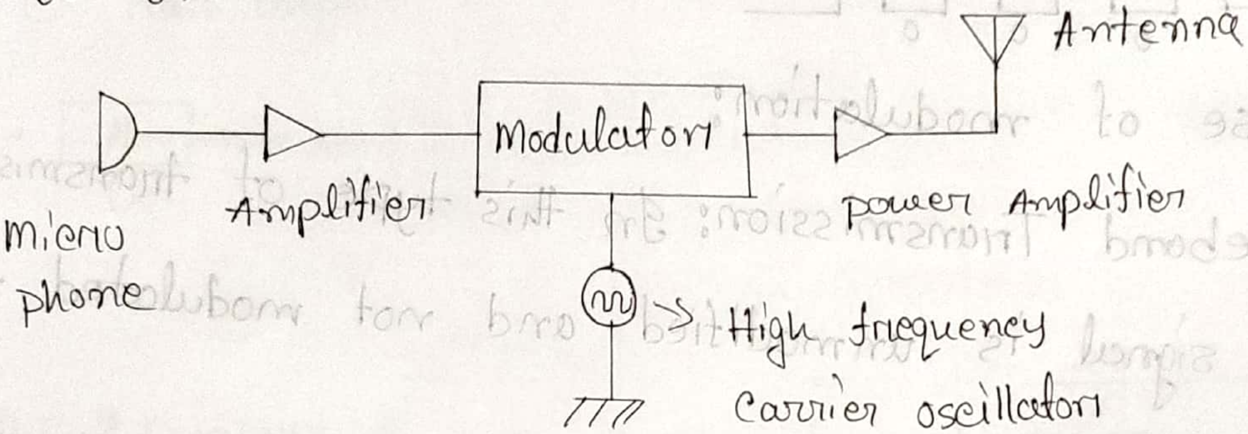
Broad band Transmission: A broadband transmission takes place when a carrier signal is modulated, amplified and sent to the antenna for transmission.

Ex: wifi, modem, Bluetooth, wimax.

Carrier signal: A high frequency signal that is modulated by audio, video or data.

Which communication system refer in long distance transmition?

田 Modulation : Modulation is an technique for transmitting information efficiently from one place to another



Modulation are 3 types:

① Amplitude Modulation (AM)

② Frequency Modulation (FM)

③ Phase Modulation (PM)

田 Why do we need modulation?

Ans:

① To design practical antenna for wirless communication: The maximum length of a good antenna must be $\lambda/4$. Since $\lambda = \frac{c}{f}$, so if frequency is around 3 KHz, $\lambda = \frac{3\times10^8}{3\times10^3} = 10^5$ m which is huge Length. So by modulation technique we can use a high frequency carrier signal and hence obtain shorter and more economical size antenna.

(ii) Multiplexing is possible only with modulation:
It is the sharing of the same media for different signal. If the frequency of the signals of different stations (radio) are same then we won't be able to separate them at the reciver side. So, modulation could be used to allow multiplexing where different carrier frequency signals could be used to differentiate each signals.

✱ Bandwidth: Bandwidth is the portion of the electromagnetic ~~signal~~ spectrum occupied by a signal.

✱ Electromagnetic Spectrum: The range of electrical signals encompassing all frequencies is referred to electromagnetic spectrum.

Voice frequency → 300 – 3000 Hz
Low frequency → 30 – 300 KHz

Medium frequency ⟶ 300 − 3000 KHz

High frequency ⟶ 3 − 30 MHz

Super High frequency ⟶ 1 − 30 GHz

26-02-20

Introduction to Signal: A signal is a set of information or data which are physical quantities or variables by means of which message or information can be transmitted.

Kind of signal:

① Continuous time signal: It will be continuous.

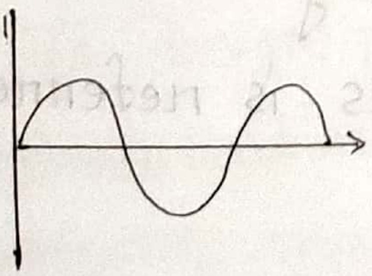
② Discreate time signal: It will be discreate.

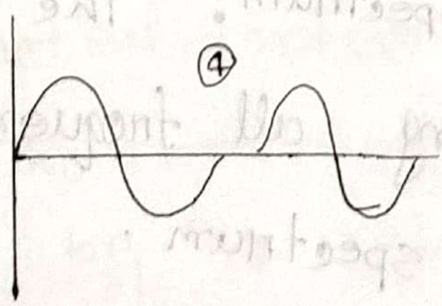
③ Periodic Signal: It will be repeat for a certain time period.

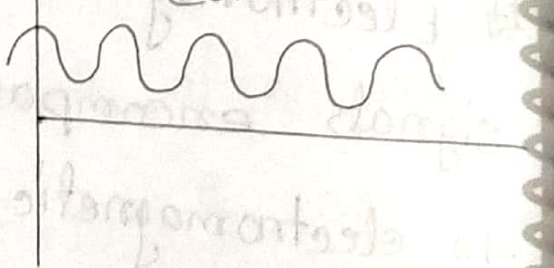
④ Aperiodic signal: It won't be repeated for a certain time period.

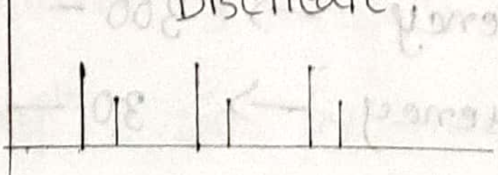
⑤ Power Signals: It has finite power.

⑥ Energy Signal: It has finite energy.

③


④


continuous


Discreate


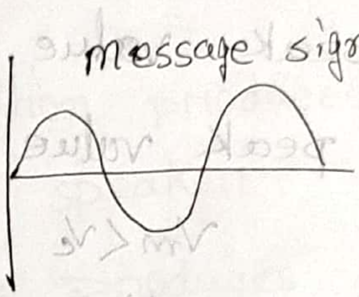
Chapter 2:

Continuous wave modulation:

Amplitude modulation: In modulation process the message signal modifies the carrier signal.

$A \sin(2\pi f t + \phi)$

$\uparrow$ AM $\uparrow$ FM $\uparrow$ PM



$m(t) \longrightarrow$ [modulator] $\longrightarrow$ Modulated wave

$\uparrow$ $c(t)$

$V_m(t) = V_m \cos 2\pi f_m t$

$V_c(t) = V_c \cos 2\pi f_c t$

⇒ In AM modulation, the amplitude of carrier signal will be vary according to the message signal. No change frequency or phase of the carrier

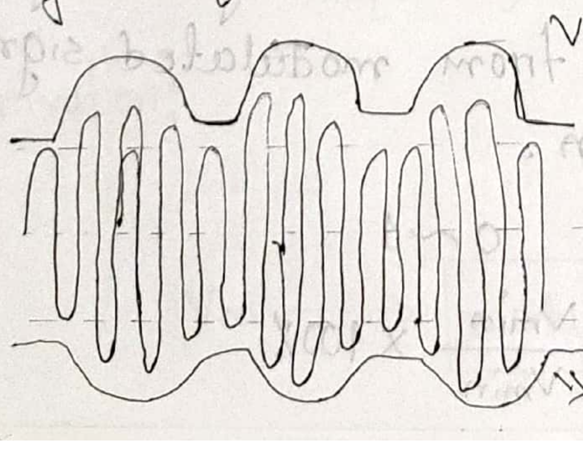


Fig.: AM

Envelope: An imaginary line called envelope connects the positive and negative peaks of the carrier waveform.

Distortion:

→ In AM modulation, it is particularly important for the peak value of the message signal be less than the peak value of the carrier.

$$V_m < V_c$$

→ If not distortion would occur and original signal will change.

Modulation index (m): The modulation index is the degree of modulation that gives a relationship between amplitude of the modulating signal and of the carrier signal.

$$m = \frac{V_m}{V_c}$$

Over modulation and Distortion: If m is >1, [$V_m > V_c$] then distortion would occur and original signal could not be attained from modulated signal. This is over modulation.

m should be between $0 \sim 1$

$$m = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100\%$$

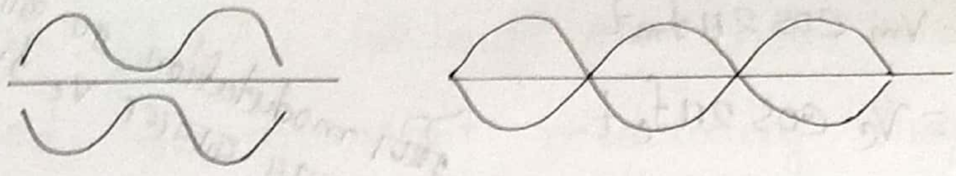


Fig.: Overmodulation

◻ Why distortion occur in amplitude modulation?

◻ Effects of distortion:

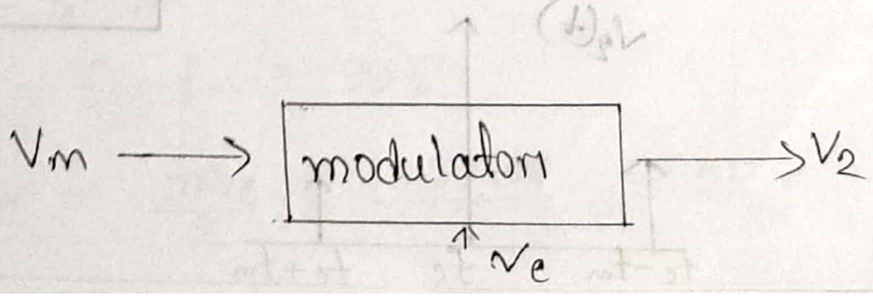
① Distortion of voice transmition produces harsh or unnetural sounds in the speaker.

② Distortion of video signals produces a scrambled a inaccurate picture in TV screen.

◻ Find out the expression of amplitude modulation wave with upper sideband and lower sideband.

⇒ Sideband or side frequency are generated as parts of the modulation process and occur in the frequency spectrum directly above and below the carrier frequency.

Derivation:



Let

$V_{(m)} = V_m \cos 2\pi f_m t$

$V_{(c)} = V_c \cos 2\pi f_c t$

→ এটা modulation এর আগে থাকে কারণে V_c fixed

Instantenous voltage $V_i = V_{(c)} + V_{(m)} = V_c + V_m \cos 2\pi f_m t$

O/P $\quad V_2 = V_i * \cos 2\pi f_c t$

$= V_c \cos 2\pi f_c t + V_m \cos 2\pi f_m t \cdot \cos 2\pi f_c t$

$= V_c \cos 2\pi f_c t + m V_c \cos 2\pi f_m t \cdot \cos 2\pi f_c t$

$$= \underbrace{V_c \cos 2\pi f_c t}_{\text{Carrier}} + \frac{m V_c}{2} \underbrace{\cos 2\pi (f_c + f_m) t}_{\text{U.S.B}} + \frac{m V_c}{2} \underbrace{\cos 2\pi (f_c - f_m) t}_{\text{L.S.B}}$$

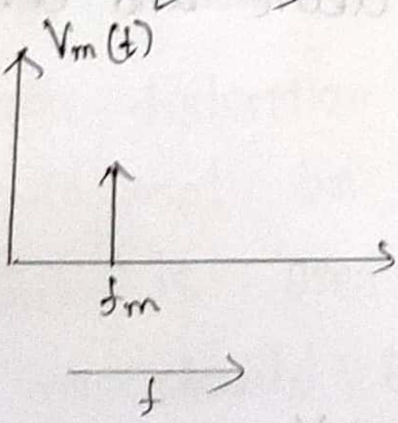
$[\because 2\cos A \cos B = \cos(A+B) + \cos(A-B)]$

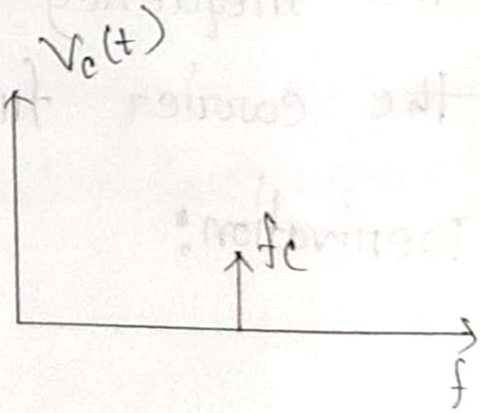
$\therefore f_{U.S.B} = f_c + f_m$

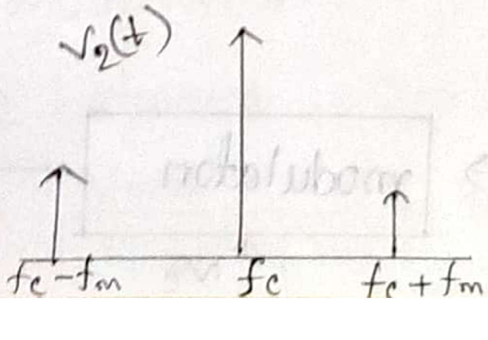
$f_{L.S.B} = f_c - f_m$

Bandwidth $= f_{U.S.B} - f_{L.S.B}$

In frequency domain:







Example 01: A standerd AM broad cast station is allowed to transmit modulating frequency up to 5 KHz if the AM station is transmitting on a frequency of 980 KHz what are the side band frequency and total bandwidth.

Ans: $f_m = 5\ KHZ$

$f_c = 980\ KHZ$

$f_{U.SB} = f_c + f_m = 5 + 980 = 985\ KHZ$

$f_{L.SB} = \cancel{98}\ f_c - f_m = 980 - 5 = 975\ KHZ$

$\text{Bandwidth} = f_{U.S.B} - f_{L.S.B} = 985 - 975 = 10$

Example 02: A sinusoidal carrier voltage of frequency 1 MHz and amplitude 100v. Is amplitude modulated by sinusoidal voltage frequency 5 KHz producing 50% modulation. Calculate frequency and amplitude of lower Sideband and uper side band.

Soln: $f_c = 1\ MHz = 10^3\ KHZ$

$f_m = 5\ KHZ$

$\therefore f_{U.S.B} = f_c + f_m = 10^3 + 5 = 1005$

$\therefore f_{L.S.B} = f_c - f_m = 995$

$\text{Amplitude} = \frac{mV_c}{2} = \frac{0.5 \times 100}{2} = 25\ V$

04-03-20

Example 03: A carrier wave of frequency 10 MHz and peak value of 10V amplitude modulated by a 5 KHz sine wave of amplitude 6V.

Determine
① Modulation factor /or index.
② Side band frequency.
③ Amplitude of sidebands
④ draw the frequency spectra.

Solⁿ: Given $f_c = 10\ MHz = 10\times10^3\ KHz$

$V_c = 10V$

$f_m = 5KHz$

$V_m = 6V$

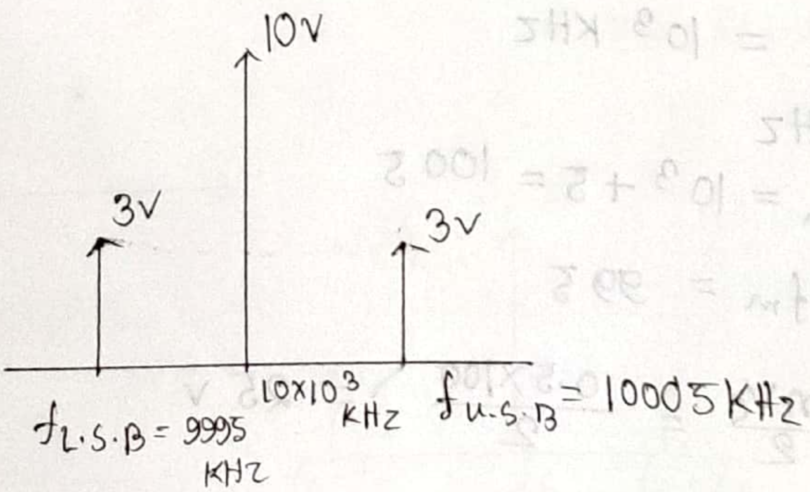
① modulation factor $m = \frac{V_m}{V_c} = \frac{6}{10} = 0.6$

(ii) $f_{U.S.B} = f_c + f_m = (10\times10^3) + 5 = 10005\ KHz$

$f_{L.S.B} = f_c - f_m = 10\times10^3 - 5 = 9995\ KHz$

(iii) Amplitude $= \frac{mV_c}{2} = \frac{0.6\times10}{2} = 3V$

(iv)



Amplitude Modulation power calculation:

→ In radio transmission, the AM signal is amplified by a power amplifier.

→ The AM signal is a composite of the carrier and sideband signal voltages.

→ Each signal produces power in the antenna.

→ Total transmitted power (P_T) is the sum of the carrier power (P_c) and power of two sidebands ($P_{U.S.B}$ & $P_{L.SB}$)

Peak envelope power $\frac{V_{rms}^2}{R}$

carrier power $P_c = \frac{V_c^2}{R} = V_c^2$ $[\because R = 1\Omega]$

sideband Power $P_{USB} = \left(\frac{m V_c}{2}\right)^2 \cdot \frac{1}{R}$

$= \frac{m^2 V_c^2}{4} = P_c \frac{m^2}{4}$

similarly $P_{LSB} = P_c \frac{m^2}{4}$

Total power content inside the signal

$P_T = P_c + P_{USB} + P_{LSB}$

$= V_c^2 + \frac{m^2 V_c^2}{4} + \frac{m^2 V_c^2}{4}$

$= V_c^2 + \frac{m^2 V_c^2}{2}$

$= V_c^2 \left(1 + \frac{m^2}{2}\right)$ $\therefore P_T = P_c \left(1 + \frac{m^2}{2}\right)$

* Calculate the amplitude modulation power or Derive the equation.

Example 04: A communication transmitter radiates a 1000 w amplitude modulated signal if carrier power is 700w. Calculate the modulation index.

Soln:

$P_c = 700\,W \qquad m = ?$

$$P_T = P_c\left(1+\frac{m^2}{2}\right)$$

$$\Rightarrow 1000 = 700\left(1+\frac{m^2}{2}\right)$$

$$\Rightarrow 1+\frac{m^2}{2} = 1.429$$

$$\Rightarrow \frac{m^2}{2} = 0.429$$

$$\Rightarrow m^2 = 0.857$$

$$\therefore m = 0.926 \quad \text{Ans:}$$

Example 5: A carrier wave of 500W is subjected to 100% amplitude modulation.

(1) Determine power in side band.

(2) Power of modulated wave.

Soln: (1) Power in sideband $= P_{USB} + P_{LSB}$

$$= \frac{1}{2} m^2 P_c$$

$$= \frac{1}{2} \times 1 \times 500 = 250\,w$$

(ii) $P_T = P_c + P_{USB} + P_{LSB}$

$$= 500 + 250 = 750\,W \quad \text{Ans:}$$

Classification of amplitude modulation:

① Double sideband suppressed carried / Double sideband (DSSC / BSB)

② single sideband (SSB)

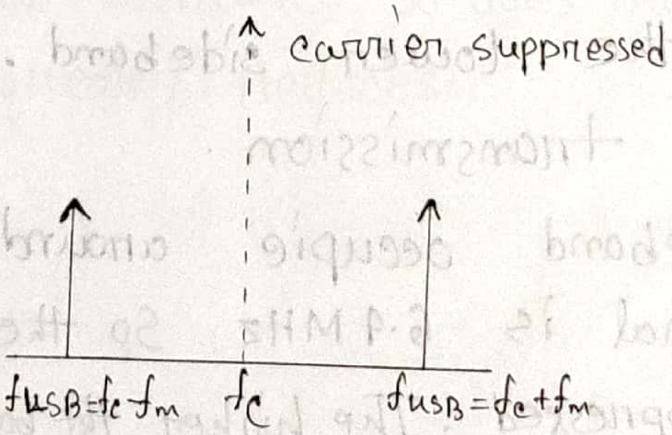
③ Vestigial sideband (VSB)

Double sideband suppressed carrier (DSSC): Since signal information is contain within the sidebands. so in this type the carrier is suppressed leveing behinds the sidebands only:-

-> No power wasted on carrier.

-> A balanced modulator is a circuit used to produce the sum and differences of frequency of DSSC signal. But to cancel or balance out the carrier.

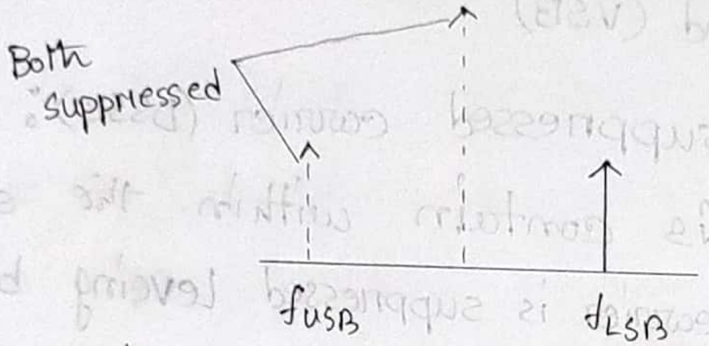
-> Double sideband is not widly used as it takes larger bandwidth and it is difficult to demodulate at receiver.



⊞ Solution of: use of pilot carrier (Low Level)

Single sideband: One sideband is all that is necessary to convey the information.

-> Here both carrier and one sideband is supressed



⊞ Why single sideband is more effective than DSSC?

Ans: Spectrum space is conserve and allowed more signal to be transmitted.

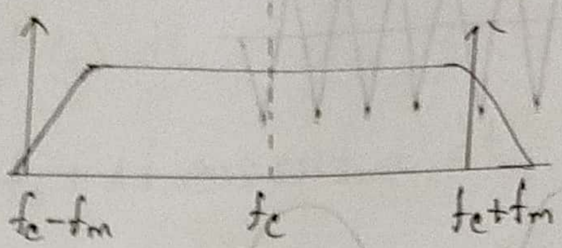
-> All power is channeled in single sideband which produces a stronger signal.

-> Occupied bandwidth space is narrower.

-> Noise in the signal is reduced.

Vestigial sideband (VSB): It is produced by partially suppressing the lower sideband. It is usually used in TV transmission.

-> Each sideband occupie around 4.2 MHz. But the allowed signal is 6.4 MHz. So the lower sideband partially suppressed. The higher the bandwidth lower the quality.



Limitation of amplitude modulation:

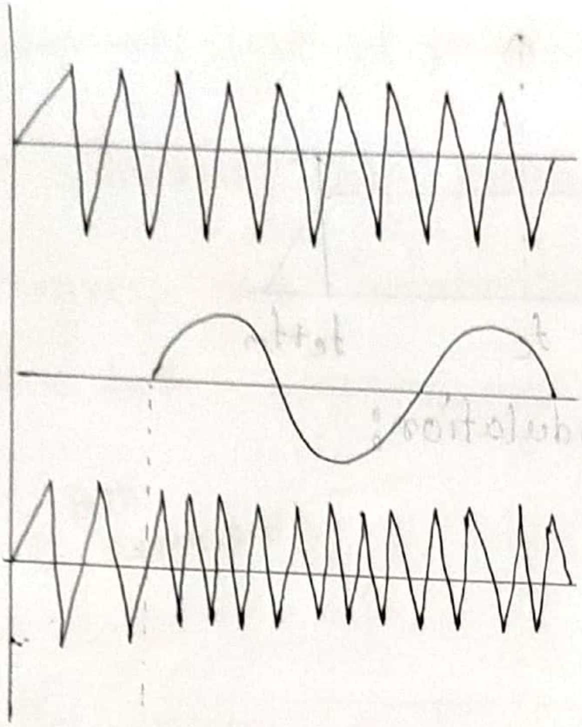
1. Noisy reception.
2. Low efficiency.
3. Small operating range.
4. Lack of audio quality.

16-03-20

Fundamental an circuits of frequency Modulation:

Basic principle of FM modulation: In FM modulation the frequency of a carrier signal is varied in accordance to the amplitude of the modulation signal.

As the amplitud increase so does frequency and as it drops frequency reduces.



Modulation index & sidebands / Modulated signal expression:

Considering a message / modulating signal

$m(t) = A_m \cos 2\pi f_m t$ --- (1)

and carrier $c(t) = A_c \cos 2\pi f_c t$ --- (2)

Since frequency modulation.

Instantaneous frequency

$f_i(t) = f_c + K_f A_m \cos 2\pi f_m t$ --- (3)

Since $A_m \cos 2\pi f_m t$ is voltage component.

$\therefore$ $K_f A_m \cos 2\pi f_m t$ = frequency.

$\therefore f_i(t) = f_c + \Delta f \cos 2\pi f_m t$ --- (4)

where $\Delta f = K_f A_m$ = frequency deviation.

The angle $\theta_i(t)$ of the FM signal is

$$\theta_i(t) = 2\pi \int_0^t f_i(t)\, dt$$

$$= 2\pi \int_0^t f_c + \Delta f \cos 2\pi f_m t \, dt$$

$$= 2\pi f_c t + \frac{2\pi \cdot \Delta f}{2\pi f_m} \sin 2\pi f_m t$$

$$= 2\pi f_c t + \frac{\Delta f}{f_m} \sin 2\pi f_m t$$

Since $\frac{\Delta f}{f_m} = \beta$ = modulation index for FM

$\therefore$ Eq (5) becomes

$$\theta_i(t) = 2\pi f_c t + \beta \sin(2\pi f_m t)$$

$\therefore$ FM signal

$$s(t) = A_c \cos \theta_i(t)$$

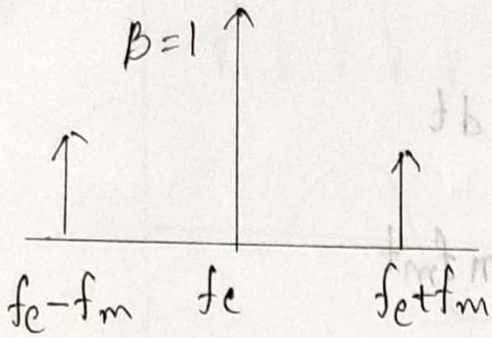
$$= A_c \cos[2\pi f_c t + \beta \sin(2\pi f_m t)]$$

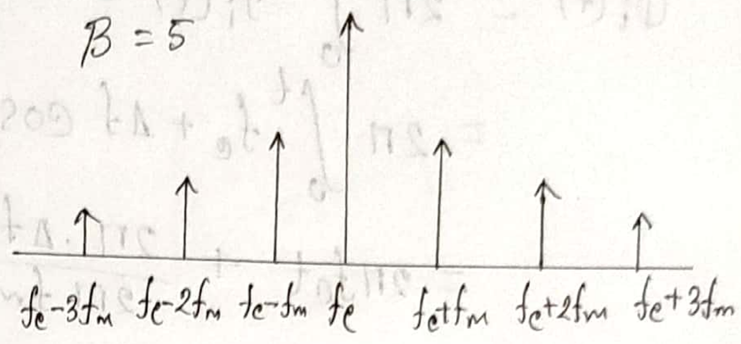
Depending on the value of β. we can classify FM into:

① Narrowband FM (NBFM) where β is small compared to 1.

② Wideband FM (WBFM) where β is much larger than 1.

State the carson's rule:





→ The higher the modulation index in FM the greater the number of significant sidebands and wider the bandwidth of the signal.

→ When spectrum conservation is necessary the bandwidth of an FM signal can be restricted by putting an upper limit on the modulation index.

There are two values of find the Bandwidth of FM signal.

① Carson's rule: The transmission bandwidth of an FM signal generated by a signal Tune modulating signal of frequency f_m is:

$$B_T = 2\Delta f + 2f_m = 2\Delta f\left(1 + \frac{1}{\beta}\right)$$

Example 06: The carrier and modulating frequency of an FM transmitter are 100 MHz and 15 kHz respectivly. if the maximum frequency deviation 75 kHz. Find the bandwidth of FM signal

$f_c = 100$ MHz
$f_m = 15$ kHz
$\Delta f = 75$ kHz

$\therefore B_T = 2 \times 75 + 2 \times 15$
$= 180$ kHz.

04-07-20

frequency deviations constant another ~~nac~~ name sensitivity

$\Delta f = K E_s$

FM vs AM

Demodulation process Demodulator importance

Carson's rules math

FM advantage disadvantage

Types of frequency demodulation is phase Lock loop

VCO এর কাজ হিসেবে একটা frequency create করে phase lock loop এ।

| শনিবার, বুধবার | 11:30

Chapter 16

04-07-20

Example 7: In a frequency modulated wave, frequency deviation constant is 75 KHz/volt and The signal amplitude is 2V. Find the maximum frequency deviation.

Soln: Frequency deviation constant $K = 75$ KHz/V

Amplitude of signal $A_m = 2V$

Max. frequency deviation $\Delta f = K A_m = 75 \times 2$
$= 150$ KHz

Note! Another name of frequency deviation constant is sensitivity.

Comparison of FM and AM:

S.NO	FM	AM
1.	The amplitude of carrier remains constant with modulation.	The amplitude of carrier changes with modulation
2.	The carrier frequency changes with modulation.	The carrier frequency remain constant with modulation.
3.	The carrier frequency changes according to the strength of the modulating signal	The carrier amplitude changes according to the strength of the modulating signal
4.	The value of modulation index (m_f) can be more than 1.	The value of modulation factor (m) cannot be more than 1 for distortionless AM signal

Advantage of FM over AM:

① It gives noiseless reception. Noise is a form of amplitude variations and a FM receiver will reject such signals.

② The operating range is quite large.

③ It gives high fidelity reception.

④ The efficiency of transmission is very high.

Disadvantage of FM:

① FM use considerably more frequency spectrum space.

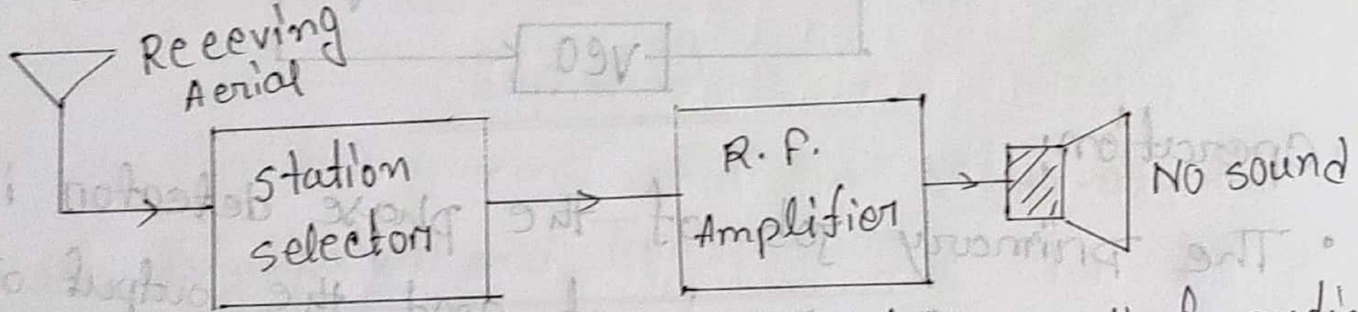
② FM uses more complex circuity for modulation and demodulation.

③ Range of communication is limited due to line of sight communication.

Demodulation: The process of recovering the audio signal from the modulated wave is known as demodulation or detection.

At the broadcasting station modulation is done to transmit the audio signal over larger distances to a reciver. When the modulated wave is picked up by the radio receiver, it is necessary to recover the audio signal from it. The process is accomplished in the radio reciver and is called demodulation.

Necessity of demodulation: It was noted previously that amplitude modulated wave consists of carrier and sideband frequencies. The audio signal is contained in the sideband frequencies which are audio frequencies. If modulated wave after amplification is directly fed to the speaker as shown in figure, no sound will be heard. It is because diaphragm of the speaker is not at all able to respond to such high frequencies. Before the diaphragm is able to move in one direction, the rapid reversal of current tends to move it in the opposite direction i.e. diaphragm will not move at all. Consequently no sound will be heard

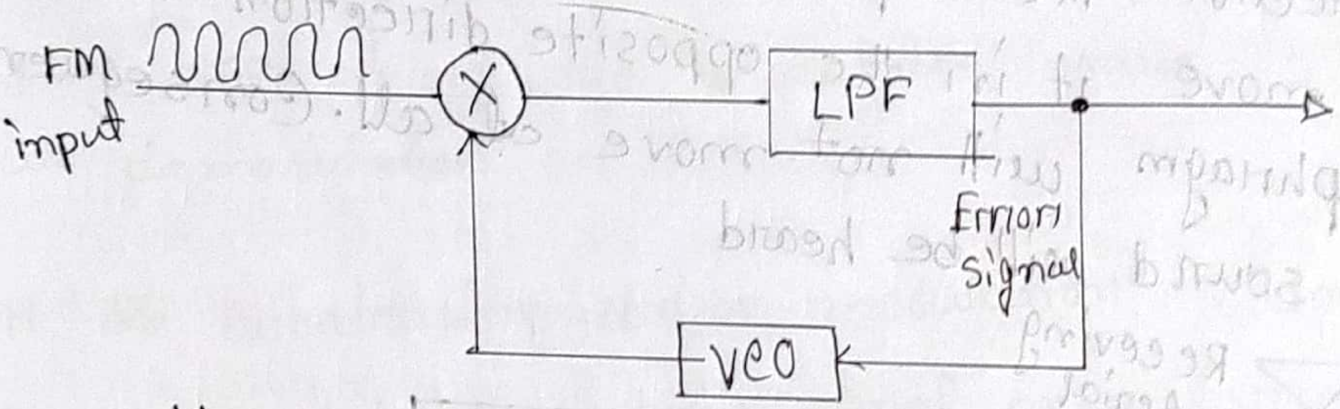


from the above discussion, it follows that audio signal must be separated from the carrier at a suitable stage in the receiver. The recovered audio signal is than amplified and fed to the speaker for conversion into sound.

Frequency demodulator:

Phase locked loops (PLL): A PLL is a frequency or phase sensitive feedback control circuit used is frequency demodulation.

It consists of
(i) Phase detector.
(ii) Low pass filter
(iii) VCO (Voltage controlled oscillator)



Operation:

- The primary job of the phase detector is to compare the FM input and the output of VCO and generate an output signal which will be filtered and controls the VCO
- If there is a phase or frequency differenc among the FM input and VCO output the phase detector output will

vary according to the differencies.

- The filtered signal is the required modulatory signal and is fed back to adjust VCO.
- When no input is given the VCO obtains no error signal message and operates at free-running mode. which is it's operating speed.

Note: VCO's main job in PLL is to create a frequency by it's own.

Chapter 3

09-07-20

Voice Digitization and Transmission

Digital Transmission of Data:

- Since the mid-1970s, digital methods of transmitting data have slowly replaced analog.
- Radio communication has remained primarily analog because the type of information to be conveyed is analog and because of high frequencies involved.

• Today, digital circuits are fast enough to handle the processing of radio signals.

• Digital processing is more cost-effective and practical.

Benefits of Digital Communication:

• Noise immunity: Digital signals, which are ~~more~~ usually binary, are more immune to noise than analog signals. Signal regeneration through the presence of regenerative repeater protects the signal from degradation.

• Error detection and correction: With digital communication, transmission errors can usually be detected and corrected.

• Compatibility with Time-Division Multiplexing: Digital data communication is adaptable to time division multiplexing schemes. Multiplexing is the process of transmitting to or more signals simultaneously on a single channel.

• Ease of signalling and accommodation of other services: Digital systems allow control information to be inserted into and extracted from a message stream independently of the nature of the transmission medium.

Disadvantage of digital communication:

- Increased bandwidth: Considerable bandwidth size is required by a digital signal.
- Complexity: Digital communication circuits are usually more complex than analog circuits.
- Need for time synchronization: A timing reference/clock is needed to control the transfer of digital data. The clock specifies when to sample the incoming signal to decide ~~to~~ which data value was transmitted.

Data Conversion: The key to digital communication is to convert data in analog form into digital form.

- Once in digital form, the data can be processed or stored.
- Data must usually be reconverted to analog form for final consumption by the user.

Basic principles of data conversion: A/D Conversion:

- An analog signal is a smooth or continuous voltage or current variation.
- Through A/D conversion these continuously variable

signals are changed into a series of binary numbers.

- A/D conversion involves two steps.

Basic principles of data conversion:

- Translating an analog signal into a digital signal is called analog to digital (A/D) conversion, digitizing a signal or encoding.
- The device used to perform this translation is known as an analog to digital converter or ADC.
- Translating a digital signal into an analog signal is called digital to analog (D/A) conversion.
- The circuit used to perform this is called a digital to analog (D/A) converter or DAC or a decoder.

A/D conversion:

- An analog signal is a smooth or continuous voltage or current variation.
- Through A/D conversion these continuously variable signals are changed into a series of binary numbers.
- A/D conversions involves two steps:
  - Sampling at regular time intervals.
  - Quantization.

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Sampling : • To retain the high frequency Information in the analog signal, a sufficient number of samples must be taken to adequatly represent the waveform.

• The minimum sampling frequency is at least twice the highest analog frequency content of the signal i.e. $f_s \geq 2H$ [sampling theorem]

• This minimum sampling frequency fs is known as the Nyquist frequency.

• In practice the sampling rate is much higher (typically 2.5 to 3 times more) than the Nyquist minimum.

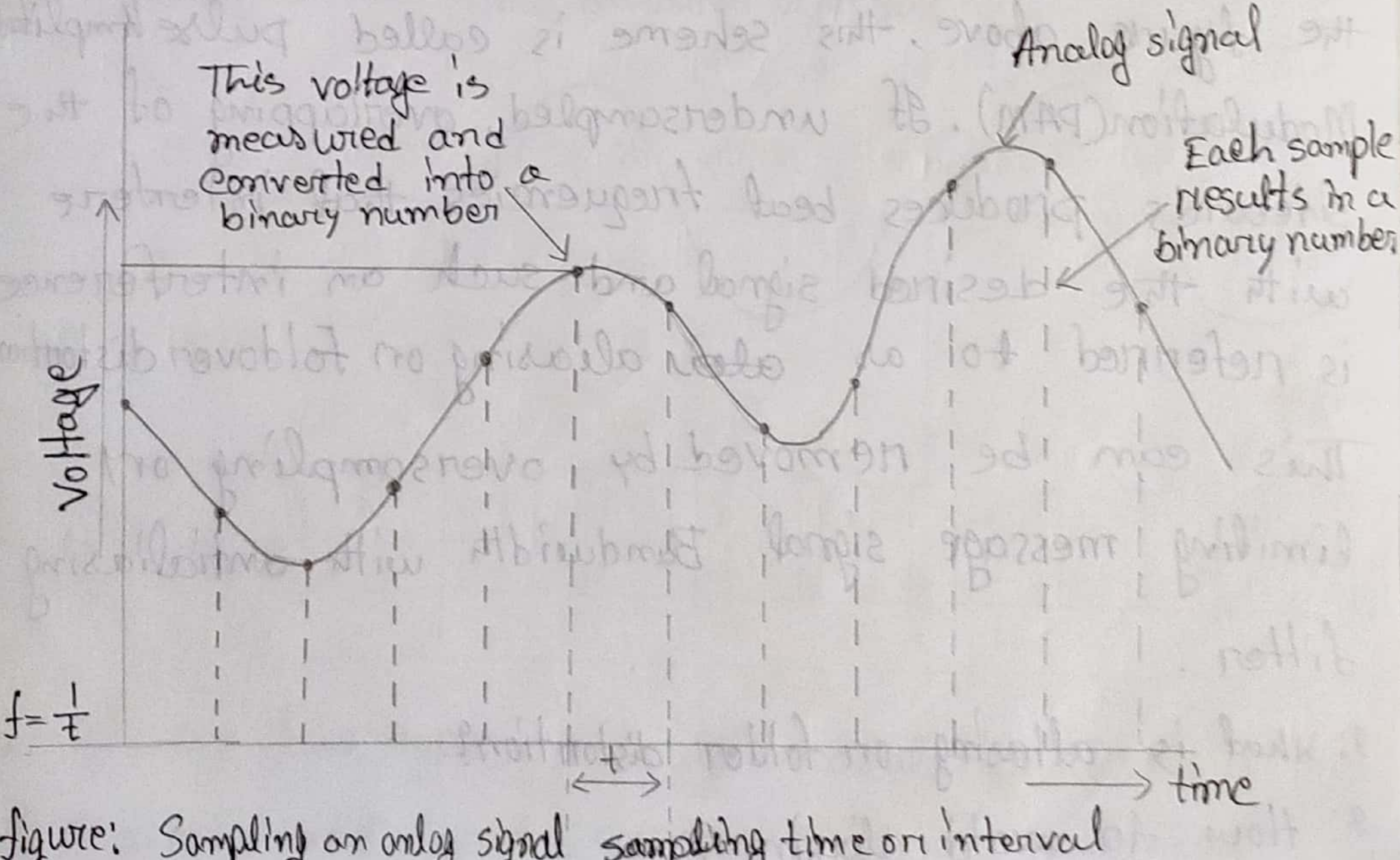
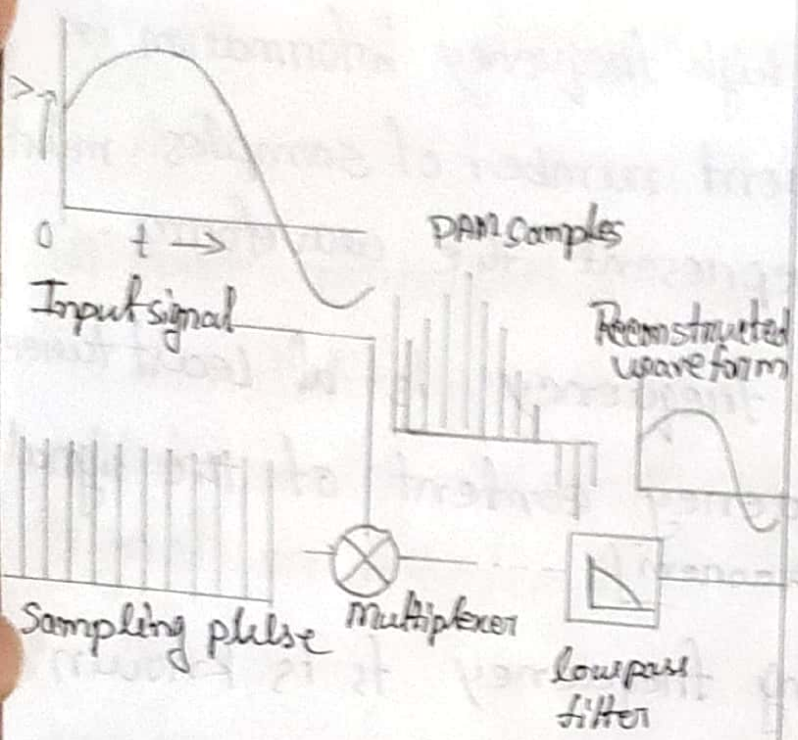
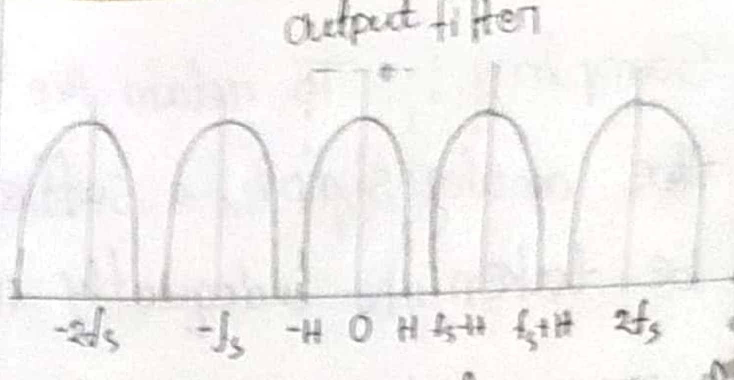


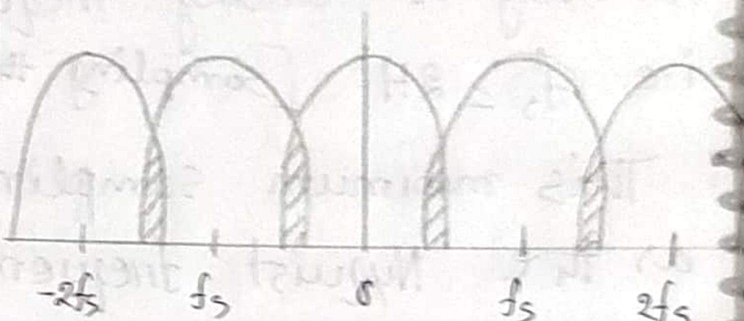
figure: Sampling an anlog signal sampling time or interval



ⓐ Sampling and recover of analog waveform



ⓑ Spectrum of PAM signal



ⓒ Spectrum overlap when $f_s < 2H$

Since the amplitude of the pulse is modulated in the figure above, this scheme is called pulse Amplitude Modulation (PAM). If undersampled, overlapping of the sidebands produces beat frequencies that interfere with the desired signal and such an interference is referred to as ~~alow~~ aliasing or foldover distortion.

This can be removed by oversampling or limiting message signal Bandwidth with antialiasing filter.

Q: What is aliasing or folder distortion?

Q: How to avoid aliasing?

• There are four basic forms of pulse modulation:

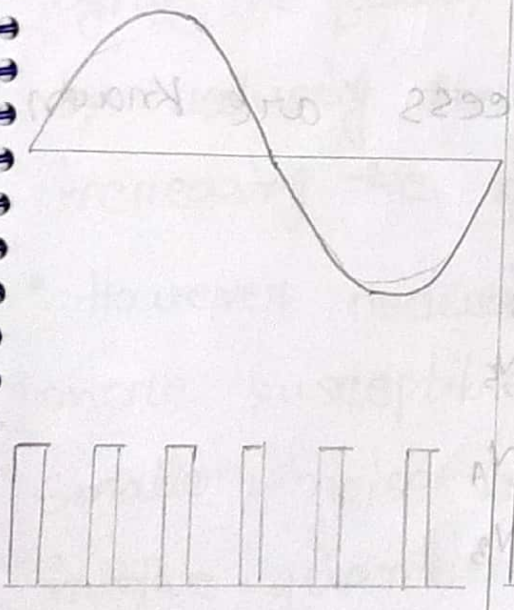
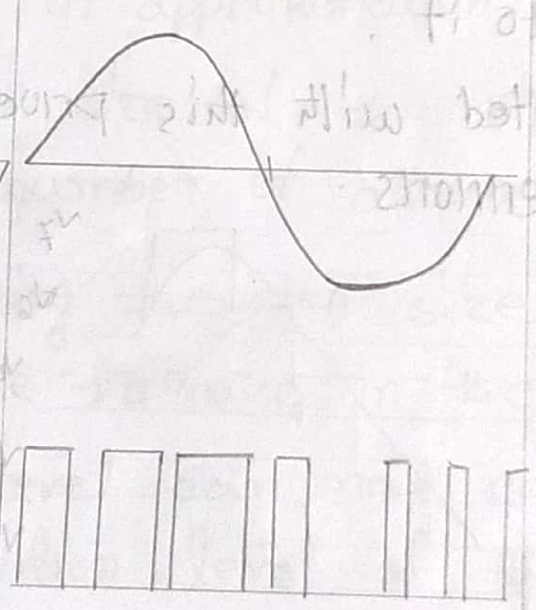
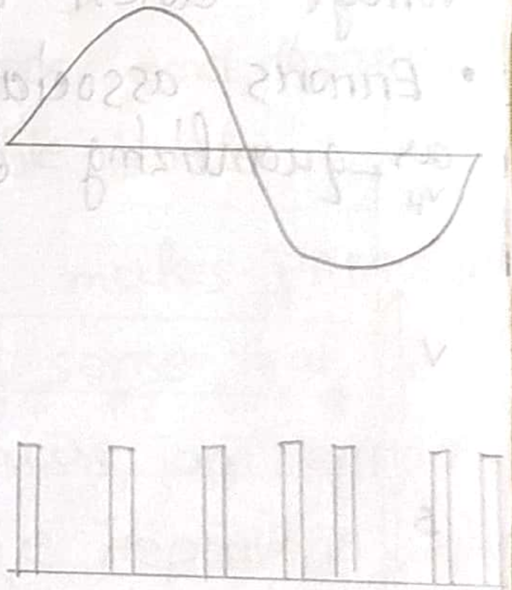
① Pulse-amplitude modulation (PAM)

② Pulse-width modulation (PWM)

③ Pulse-position modulation (PPM)

④ Pulse-code modulation (PCM)

Comparism of Pulse-Modulation Methods:

PAM	PWM	PPM
The PAM signal is a series of constant-width pulse whose amplitude in accordance withe the analog signal	The PWM signal is binary in amplitude. The information signal varies the width or time duration of the pulse	In PPM pulse change position according to the amplitude of the analog signal.
		

• Of the four types of pulse modulation, PAM is the simplest and least expensive to implement.

Quantization:

- Both analog signal and PAM signal represent an infinite number of actual voltage values.
- The A/D converter can represent only a finite number of voltage values over a specific range.
- The samples are converted to a binary number whose value is close to the actual sample value.
- An A/D converter devides a voltage range into discrete increments, each of which is represented by a binary number. This is known as Quantization.
- The analog voltage measured during the sampling process is assigned to the increment of voltage closest to it.
- Errors associated with this process are known as quantizing errors.

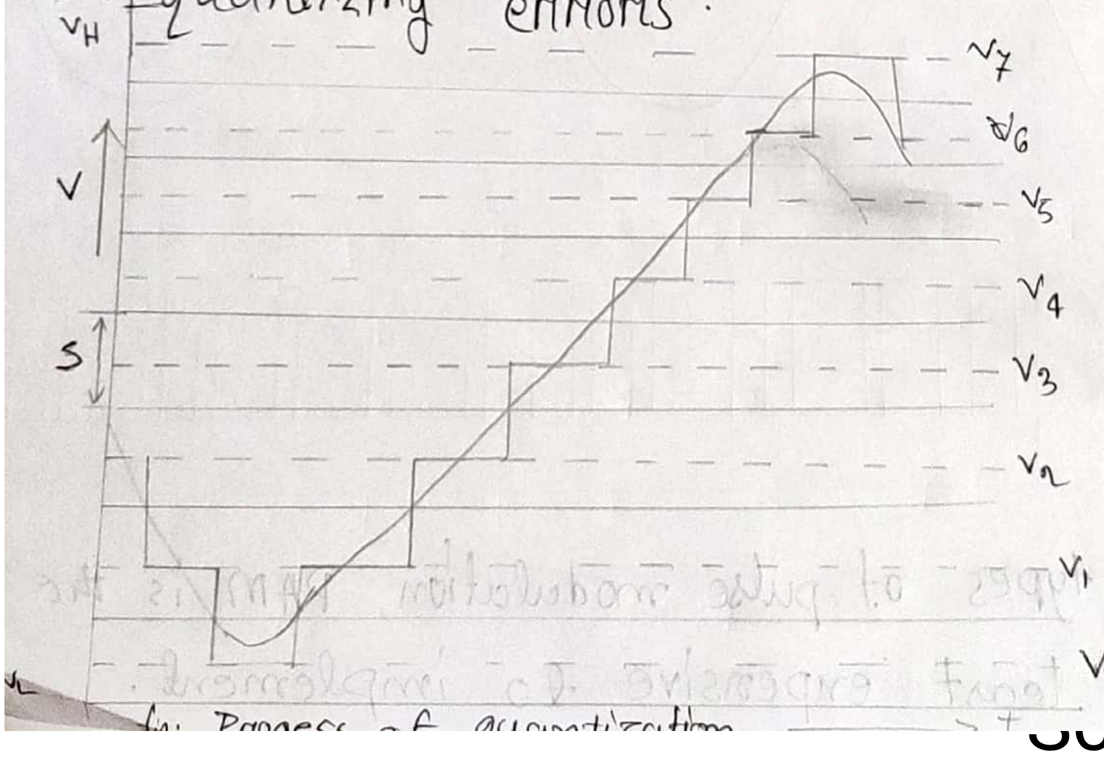


fig: Process of quantization

signal v is confined to a range from V_L to V_H and this range is divided into M equal steps. The step size s is given by $s = (V_H - V_L)/M$.

In the centre of each of these steps we locate the quantization levels $V_0, V_1, \ldots, V_{M-1}$

The quantization signal V_q takes on any one of quantized level values. A signal v is quantized to its nearest quantization level.

For example:

$$V_q = V_3 \text{ if } (V_3 - S/2) \leq V < (V_3 + S/2)$$

- The quantized signal is an approximation to the original signal. The quality of approximation may be improved by reducing the size of the steps and thereby increasing the number of allowable levels.
- However reducing the step size makes PAM signal more susceptible to noise, in the sense that a smaller noise level can now cause an error in the quantization level at the repeaters.
- The susceptibility to noise can be greatly minimized by resorting to digital coding of the PAM sample amplitudes.

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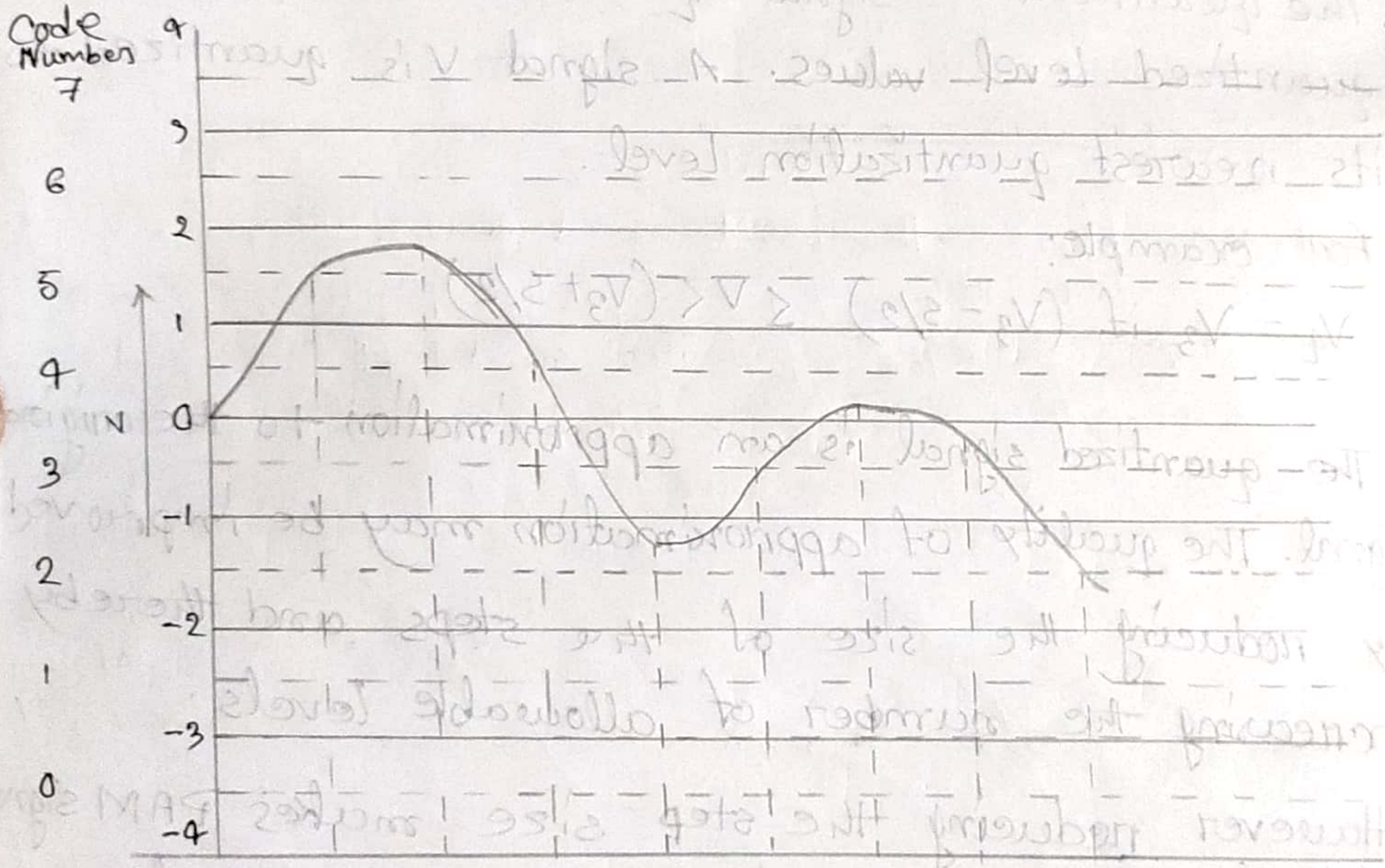
• Each quatized level is represented by a code number which is transmitted is instead of the quantized sample value itself. Code number is transmitted as a series of pulse.

• Such a system of transmission is called Pulse Code Modulation (PCM).



Sample Value	1.3	2.7	0.5	-1.1	-0.7	0.1	-0.1	-1.6
Quantized value	1.5	2.5	0.5	-1.5	-0.5	0.5	-0.5	-1.5
Code Number	5	6	4	2	3	4	3	2
Binary Code	101	110	100	010	011	100	011	010

Fig: Binary PCM

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Pulse Code Modulation:

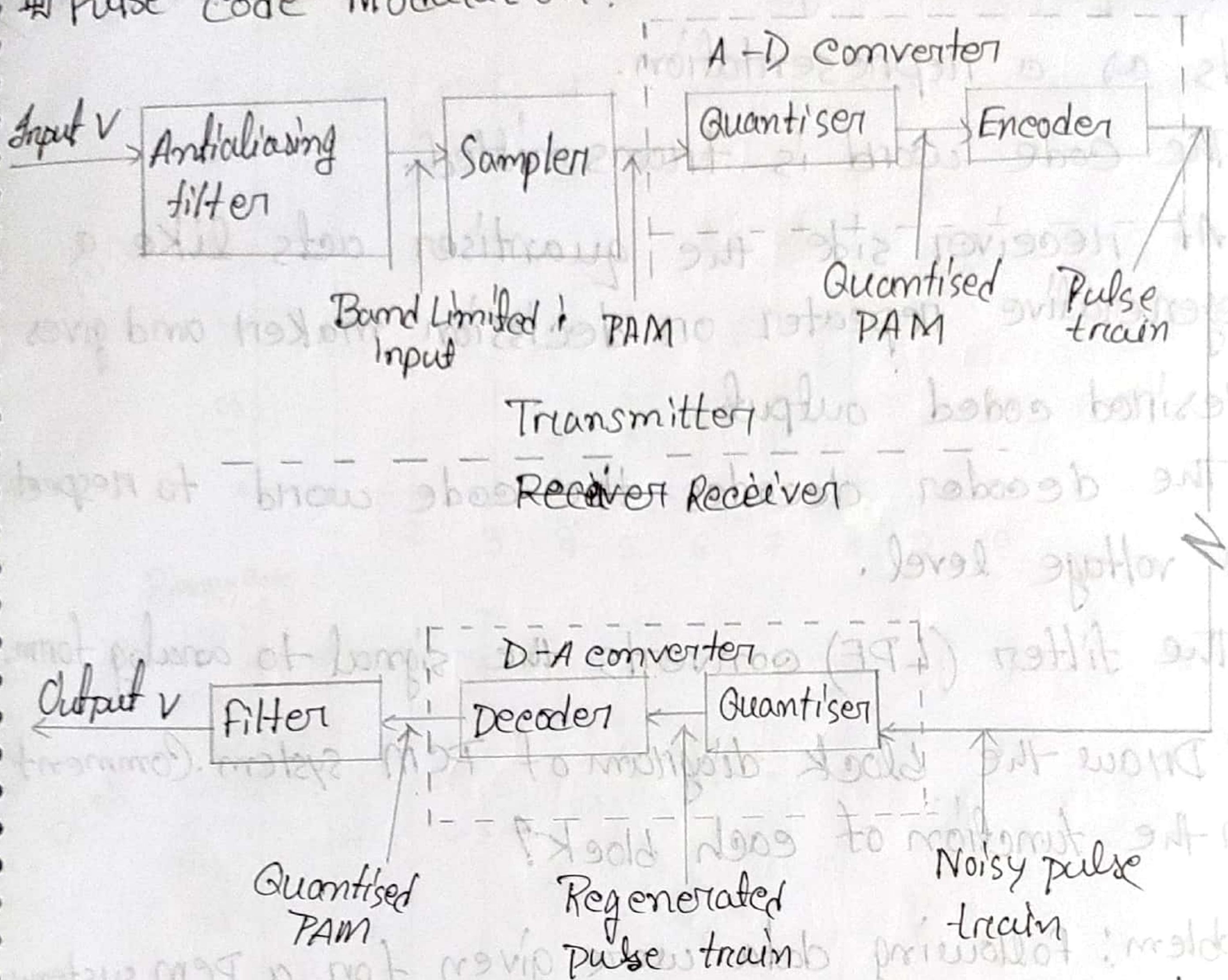


Fig: PCM system for speech communication.

PCM Process:

- The input signal is inserted to antialiasing filter to avoid foldover distortion.
- Then it goes to sampler where the signal is converted to discrete in time.
- Then to quantiser where different quantization level is set up and each sampled values takes on this respective quantized level (voltage)

- The quantized signal is then encoded with binary bits as a representation.
- The code word is transmitted.
- At receiver side the quantiser acts like a regenerative repeater or decission maker and gives desired coded output.
- The decoder decodes the code word to respective voltage level.
- The filter (LPF) converts the signal to analog form.

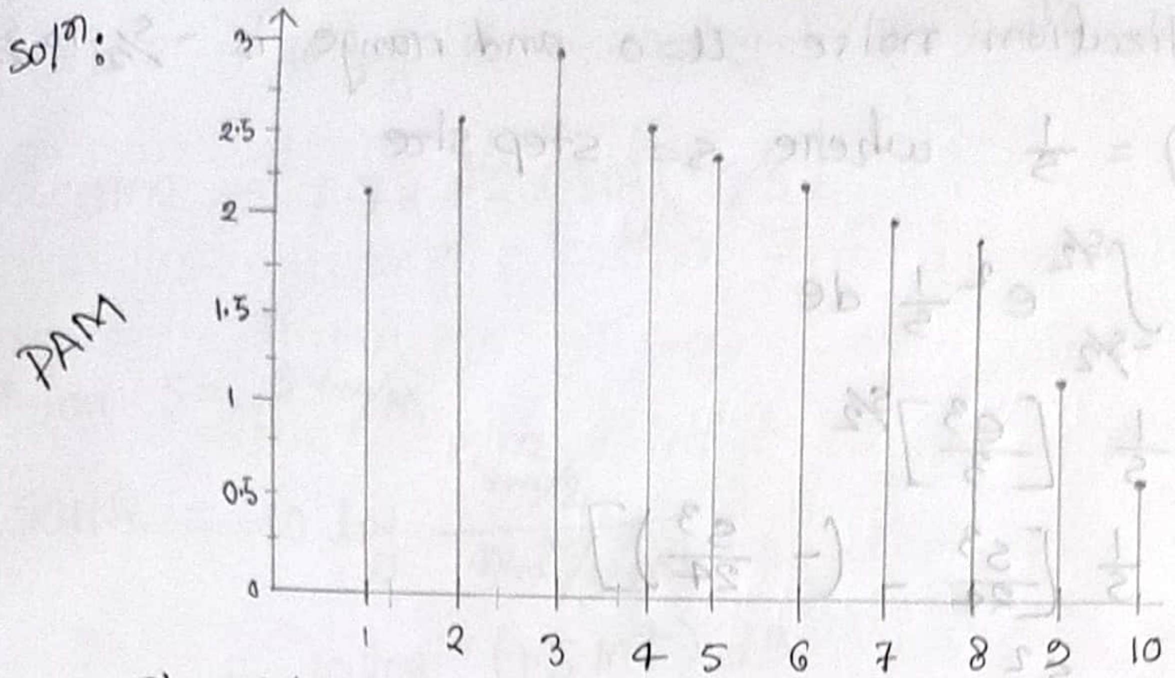
Q: Draw the block diagram of PCM system. Comment on the function of each block?

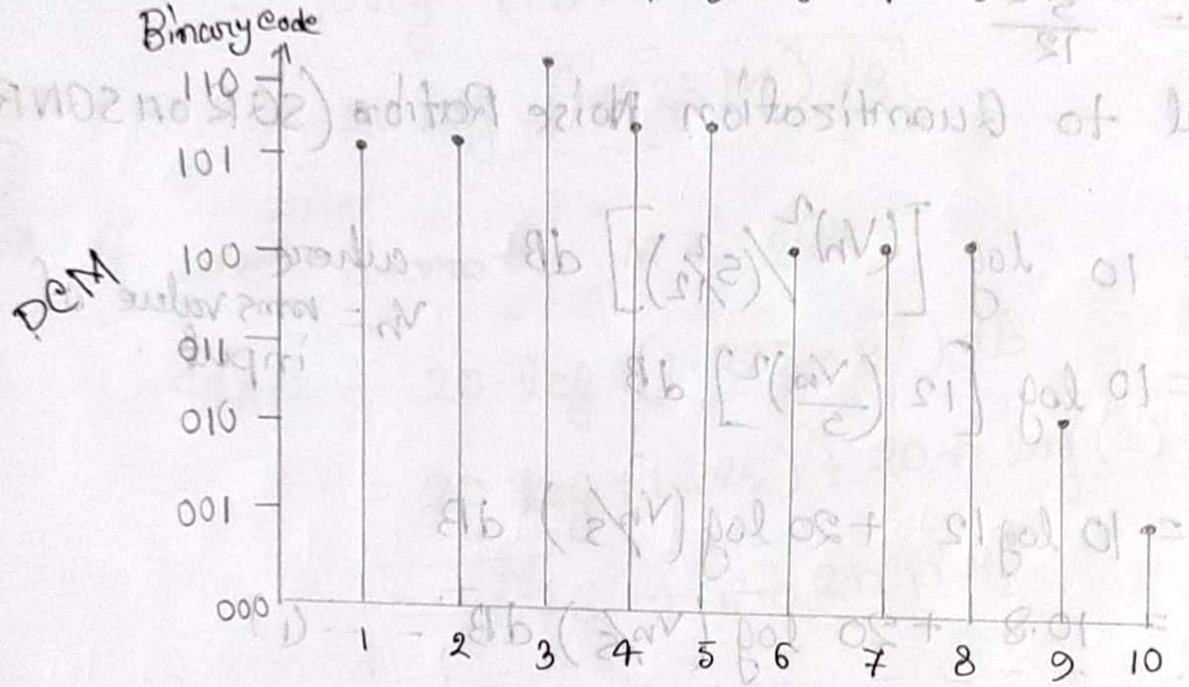
Problem: following data were given for a PCM system:

Sample instant (m sec):	1	2	3	4	5	6	7	8	9	10
Sample value:	2.3	2.7	2.9	2.6	2.4	2.2	1.9	1.8	1.2	0.6

Draw the wave shapes of corresponding PAM and PCM signal for step size 0.5v and 3 bit encodes.

Sol^n:





Quantization noise / SQNR derivation

The average quatization noise output is given by variance,

$$\sigma^2 = \int_{-\infty}^{\infty} (e - \mu)^2 \, p(e) \, de$$

where e = instantaneous error

μ = mean error

for quantization noise $\mu = 0$ and range is $-S/2$ to $S/2$

where $P(e) = \frac{1}{S}$ where $S =$ step size

$$\therefore \sigma^2 = \int_{-S/2}^{S/2} e^2 \frac{1}{S} \, de$$

$$= \frac{1}{S}\left[\frac{e^3}{3}\right]^{S/2}$$

$$= \frac{1}{S}\left[\frac{S^3}{24} - \left(-\frac{S^3}{24}\right)\right]$$

$$= \frac{S^2}{12}$$

The signal to Quantization Noise Ratio (SQR or SQNR) is

$$SQNR = 10 \log\left[(V_n)^2 / (S^2/12)\right] dB$$

where V_n = rms value of input.

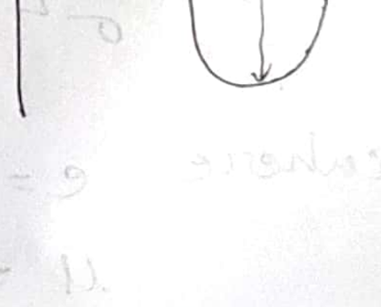
$$= 10 \log\left[12\left(\frac{V_n}{S}\right)^2\right] dB$$

$$= 10 \log 12 + 20 \log (V_n/S) \; dB$$

$$= 10.8 + 20 \log (V_n/S) \; dB \quad \text{---} (i)$$

* If the input signal is a sinusoidal wave/sine wave

$$V_n = \frac{V_m}{\sqrt{2}} \qquad S = \frac{V_H - V_L}{M} = \frac{2V_m}{M}$$



where $M =$ no of levels

$\therefore$ Eq (i) becomes

$$SQNR = 10.8 + 20 \log\left[\left(V_m/\sqrt{2}\right)/S\right] dB$$

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$= 10.8 - 20 \log \sqrt{2} + 20 \log (V_m/s)$ dB

$$SQNR = 7.78 + 20 \log (V_m/s) \text{ dB}$$

* for $S = 2V_m/M$

$$SQNR = 10 \log \frac{V_m^2/2}{4V_m^2/12M^2}$$

$$= 10 \log (1.5 M^2) \text{ dB}$$

$$= 20 \log (1.225 M) \text{ dB}$$

* If we denote $M = 2^n$

$$\therefore SQNR = 20 \log (1.225 \times 2^n) \text{ dB}$$

$$= 20 \log (1.225) + 20n \log (2)$$

$$= 1.76 + 6.02n \text{ dB}$$

Q: Derive the mathematical expression of SQNR of a PCM system.

Companding

Def^n: Companding is a system in which information is first compressed, transmitted through a band limited channel, and expanded at the receiving end.

⊠ Why do we need companding?

For highest bit rate SQR is better but there is difficulty in transmission as the transmission channel should have enough bandwidth to support it. Since the larges signals have better SQR but small ones less, so we always compress the large one rather the small one.

Process of companding: The world companding is a combination of compressing and Expanding which means that it does both. This is a non linear technique used in PCM which compress the data at the transmitter and expands the same data at the reciver. The effects of noise and crosstalk are reduced by using this technique.

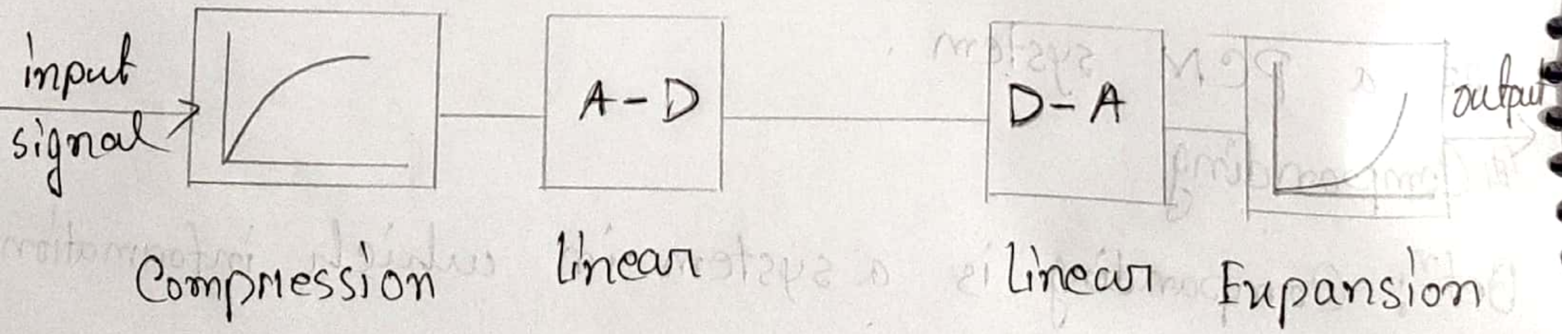


Fig: Process of companding.

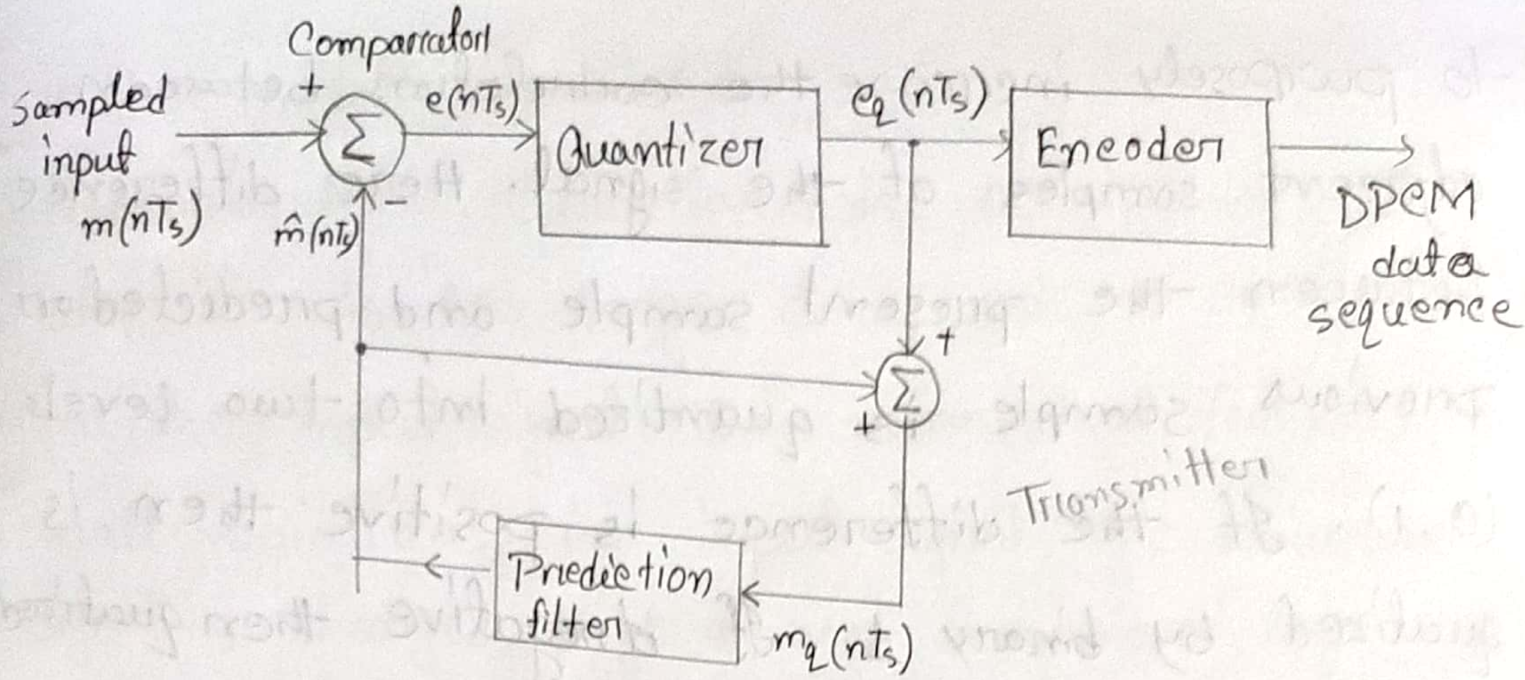
12-07-20

DPCM → Differential Pulse Code Modulation

Differential pulse code modulation (DPCM) is a procedure of converting an analog signal into a digital signal in which an analog signal is sampled and then the difference between the actual sample value and its predicted value (predicted value is based on previous sample or samples) is quantized and then encoded forming a digital value.

DPCM code words represent differences between samples unlike PCM where code words represented a sample value.

In PCM most source signals show significant correlation between successive samples so encoding use redundancy in sample values which implies lower bit rate. In such situation, it is more efficient to use DPCM where difference between sample and predicted or previous sample is transmit instead of samples value.



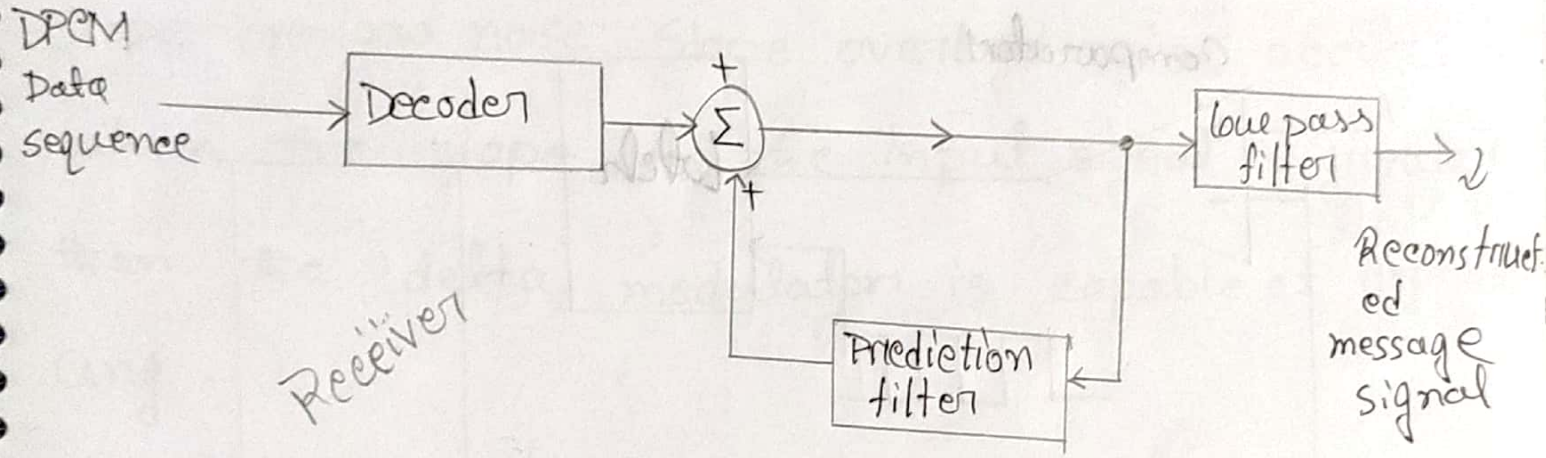
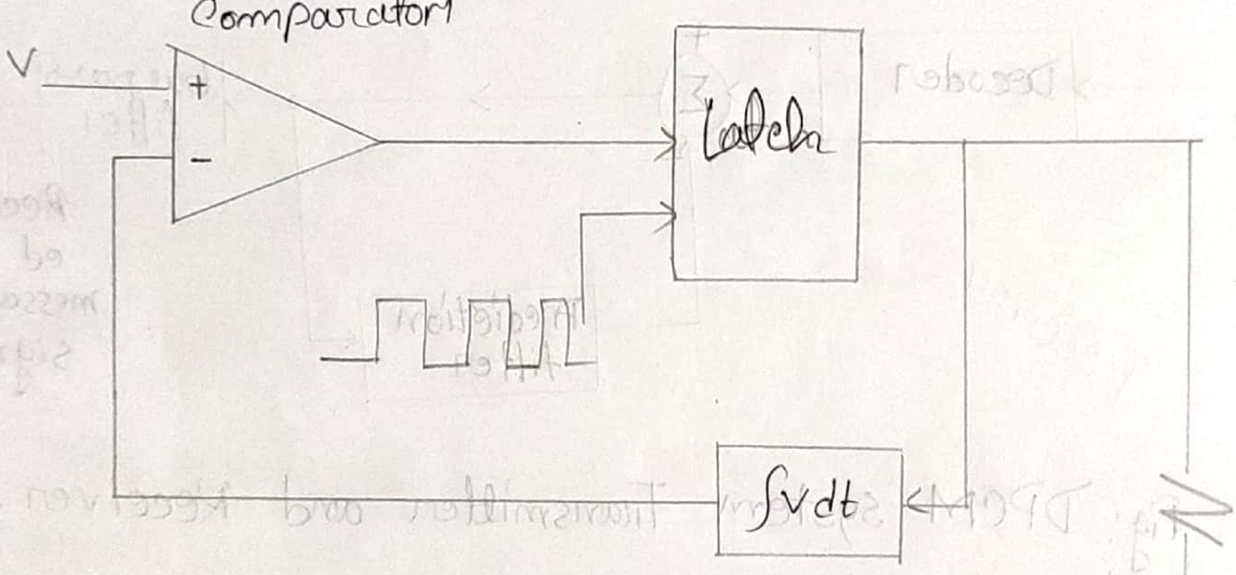


Fig: DPCM system Transmitter and Receiver.

Delta Modulation

Delta modulation (DM) is a subclass of differential pulse code modulation. It can be viewed as a simplified variant of DPCM, in which 1 bit quantizer is used with the fixed first order predictor. In this process an incoming message signal is oversampled.

to purposely increase the correlation between adjacent samples of the signal. Here difference between the present sample and predicted or previous sample is quantized into two levels (0, 1). If the difference is positive then is quatized by binary (1). If it negative then quatized by zero (0).



Transmitter

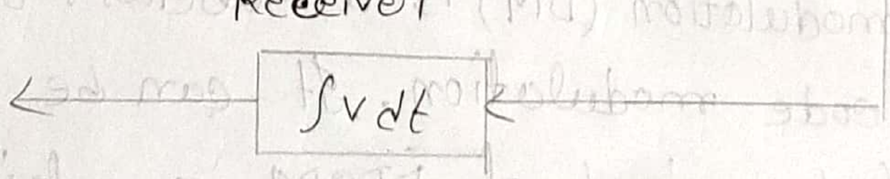


Fig: Delta Modulation

Noise in delta modulation

Granular Noise: Granular noise occurs when the step size is too large compared to small variation in the input signal. This means that for very small variations in input signal, the staircase signal is changed large amount because of large step size.

Slope overload noise: Slope overload noise occurs when the slope of the input signal is greater than the delta modulator is capable of reproducing.

Difference between DM and DPCM

S.NO	Comparison based on	DM	DPCM
1	Feedback	In DM feedback exists in transmitter.	Feedback exist in both transmitter and receiver.
2	Signal to noise ratio.	DM has poor signal to noise ratio.	DPCM has fair signal to noise ratio.
3	Transmission bandwidth	It requires lowest bandwidth	DPCM required less bandwidth than PCM
4	Levels, step size	step size is fixed	Number of levels are fixed.

5.	Efficiency	DM is less efficient than DPCM	DPCM is more efficient.
6	Number of bits	Only one bit is used per sample.	More than one but less than PCM bits are used.
7.	Quantization error and distortion.	Slop overload distortion and granular noise are present.	Slop overload distortion and quantization noise are present.
8.	Application	It is generally used in speeches and images.	It is mostly used in videos and speeches.

Maths

Equations: Quantization levels L

$$L = 2^n$$

Here n is number of bits or binary digits.

$$\therefore n = \log_2 L$$

Nyquist rate $R_N = 2B$

Number of transmitted bit $= 2nB$

Minimum channel bandwidth $= nB$ Hz

SNR or $\frac{S_o}{N_o} = (\alpha + 6n)$ dB

where $\alpha = 10 \log_{10} c$, and $c = \frac{3}{[\ln(1+\mu)]^2}$

where μ is a compression parameter. which determine the degree of compression.

Problem-1: A signal m(t) band limited to 3 KHz is sampled at a rate $33\frac{1}{3}\%$ higher than the Nyquist rate. The maximum acceptable error in the sample amplitude (the maximum quantization error) is 0.5% of the peak amplitude m_p. The quantized samples are binary coded. Find the minimum bandwidth of a channel required to transmit the encoded binary signal. If 24 such signals are time division multiplexed, determine the minimum transmission bandwidth required to transmit the multiplexed signal.

Ans: The Nyquist sampling rate is $R_N = 2\times3000 = 6000$ Hz

Actual sampling rate is $R_A = 6000\times(1\frac{1}{3}) = 8000$ Hz

Maximum quantization error $\frac{\Delta V}{2} = \frac{m_p}{L}$

$$\Rightarrow \frac{\Delta V}{2} = \frac{m_p}{L} = \frac{0.5}{100} m_p$$

$$\therefore L = 200$$

$$\therefore n = \log_2 L = \log_2 200 = 8 \text{ bit}$$

$\therefore$ Data rate $= 8\times8000 = 64000$ bit/s

$\therefore$ Minimum transmission Bandwidth $B_T = \frac{64000}{2}$

$= 32000$ Hz

$= 32$ KHz

For 24 such signal data rate is $= 24 \times 64000$

$= 1.536$ Mbits/s

$\therefore$ Minimum transmission bandwidth $= 1.536/2$

$= 0.768$ MHz

Problem-2:

A signal $m(t)$ of bandwidth $B = 4$ KHz is transmitted using a binary companded PCM with $\mu = 100$. Compare the case of $L = 64$ with the case $L = 256$ from the point of view of transmission bandwidth and the output SNR

Ans:

For $L = 64$ $n = 6$ and ~~transm~~

transmission bandwidth is $nB = 24$ KHz

$$\frac{S_o}{N_o} = (\alpha + 36) \text{ dB}$$

$$\therefore \alpha = 10 \log \frac{3}{[\ln(1+\mu)]^2}$$

$$= 10 \log \frac{3}{[\ln(101)]^2} = -8.51$$

Hence $\frac{S_o}{N_o} = 27.49 \, dB$

for $L = 256$ $n = 8$ and the transmission bandwidth

is 32 KHz

$$\therefore \frac{S_o}{N_o} = \alpha + 6n = 39.49 \, dB$$

The difference between the two SNR is 12 dB which is a ratio of 16. Thus the SNR for $L = 256$ is 16 times the SNR for $L = 64$. The former requires just about 33% more bandwidth compared to the latter.

Problem - 3: A compact disc (CD) records audio signals digitally by using PCM. Assume the audio signal bandwidth to be 15KHZ.

@ What is the Nyquist rate?

ⓑ If the Nyquist samples are quantized into $L = 65536$ levels and then binary coded, determine the number of binary digits required to encode a sample.

ⓒ Determine the number of binary digits per second (bit/s) required to encode the audio signal.

ⓓ For practical reasons discussed in the text, signals are sampled at a rate well above the nyquist rate. Practical CDs use 44100 samples per second. If $L = 65,536$, determine the number of bits per second required to encode the signal, and the minimum bandwidth required to transmit the encoded signal.

Ans:

ⓐ Nyquist rate $R_N = 2 \times 15 = 30\ KHz$

ⓑ Here $L = 65,536$

$\therefore$ Number of bits $n = \log_2 65536$

$= 16$

ⓒ Number of binary bit/s $= 2nB = 16 \times 30 = 480\ KHz$

ⓓ

Here $L = 65536$

$\therefore n = 16$

$\therefore$ Minimum Bandwidth $B_T = 16 \times 44,100 = 705.6\ KHz$

Problem-4: A television signal has a bandwidth of 4.5 MHz. This signal is sampled, quantized, and binary coded to obtain a PCM signal.

ⓐ Determine the sampling rate if the signal is to be sampled at a rate 20% above the nyquist rate.

ⓑ If the samples are quantized into 1024 Levels, determine the number of binary pulse required to encode each sample.

ⓒ Determine the binary pulse rate of the binary coded signal and the minimum bandwidth required to transmit this signal.

Ans:

ⓐ The Nyquist sampling rate $R_N = 2 \times 4.5$ MHz
$= 9$ MHz

ⓑ The actual sampling rate $R_A = 9 \times 1.2 = 10.8$ MHz

ⓑ $L = 1024$

∴ Binary pulse $n = \log_2 1024 = 10$

ⓒ Pulse rate of binary coded signal $= 10.8 \times 10^6 \times 10$
$= 1.08 \times 10^8$ bit/s

Minimum bandwidth $= \dfrac{1.08 \times 10^8}{2} = 5.4 \times 10^7$ Hz

Problem - 5:

Five telemetry signals, each of bandwidth 1 KHz, are to be transmitted simultaneously by binary PCM. The maximum tolerable error in sample amplitudes is 0.2% of the peak signal amplitude. The signals must be sampled at least 20% above Nyquist rate. Framing and synchronizing requiries an additional 0.5% extra bits. Determine the minimum possible data rate that must be transmitted, and the minimum bandwidth required to transmit this signal.

Ans: Actual sampling rate $R_A = 2 \times 1000 \times 1.2$

$= 2400\ Hz$

We know $\frac{\Delta V}{2} = \frac{m_p}{L}$

$\Rightarrow \frac{m_p}{L} = \frac{0.2}{100} m_p \quad \therefore L = 500$

And the nearest closest value of L is $L = 512$

$\therefore n = \log_2 512 = 9$

$\therefore$ Data rate of each signal is $= 9 \times 2400 = 21600\ bit/s$

$\therefore$ Data rate of 5 signals $= 5 \times 21600 = 108000\ bit/s$

$= 108\ K\ bit/s$

Including framing and synchronization bit data rate

$= 108 + \frac{108 \times 1}{100} = 108 + 1.08 = 109.08\ Kbit/s$

$\therefore$ Minimum bandwidth $= 109.08/2 = 54.54\ KHz$

Problem - 6: It is desired to set up a central station for simultaneous monitoring of electrocardiograms (ECG) of 10 hospital patients. The data from the rooms of the 10 patients are brought to a processing center over wires and are sampled, quantized, binary coded and time division multiplexed. The multiplexed data are now transmitted to monitoring station. The ECG signal bandwidth is 100 Hz. The maximum acceptable error in sample amplitudes is 0.25%. of peak signal amplitude. The sampling rate must be at least twice the nyquist rate. Determine the minimum cable bandwidth needed to transmit these data.

Ans:

$B = 100\ Hz$

Nyquist rate $= 100 \times 2 = 200\ Hz$

sampling rate $= 200 \times 2 = 400 Hz$

Here $\frac{\Delta V}{2} = \frac{mp}{L}$

$\Rightarrow \frac{mp}{L} = \frac{0.25}{100} mp \quad \therefore L = 400$

$\therefore n = \log_2 400 = 9$

Data transmission rate $= 9 \times 400 = 3600$ bit/s

Data transmission rate for 10 signal $= 3600 \times 10$ bit/s

$= 36000$ bit/s

$\therefore$ Minimum cable bandwidth needed $= 36000/2$

$= 18000\ Hz$

Chapter 4 19-07-20

Bandwidth Utilization: Multiplexing

Bandwidth utilization is the wise use of available bandwidth to achieve specific goals.
Efficiency can be achieved by multiplexing - i.e sharing of the bandwidth between multiple users.

Multiplexing:
Multiplexing is the set of techniques that allows the (simultaneous) transmission of multiple signals across a single data link.

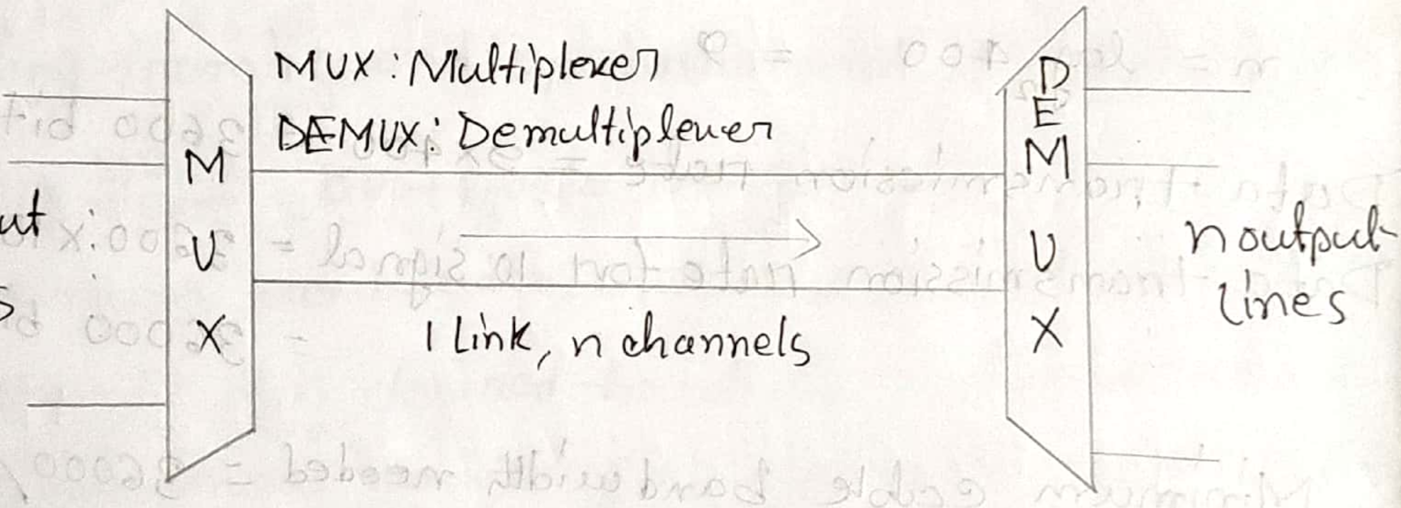
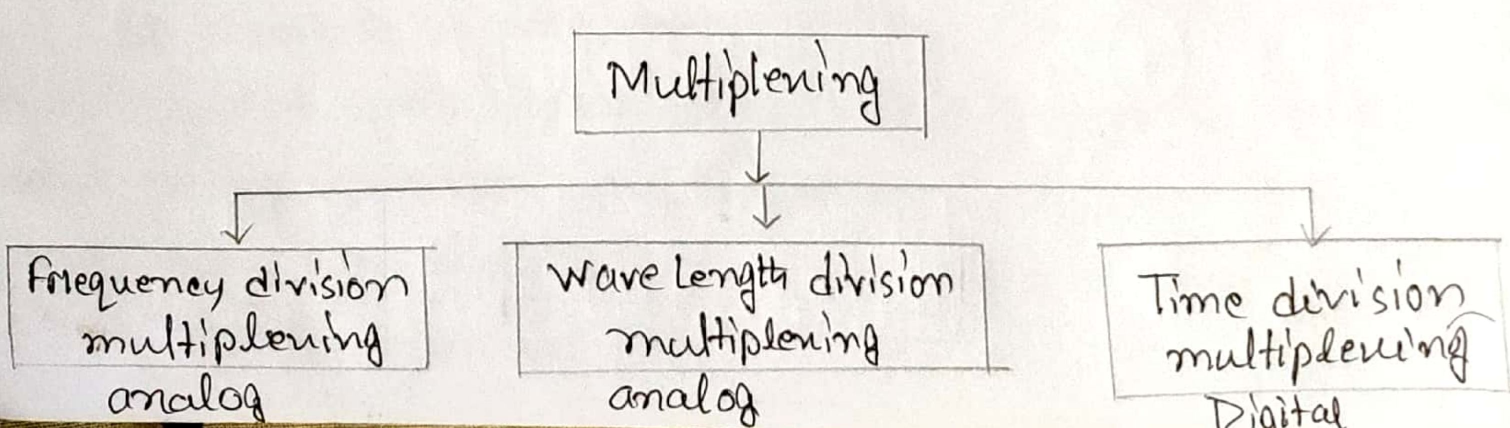
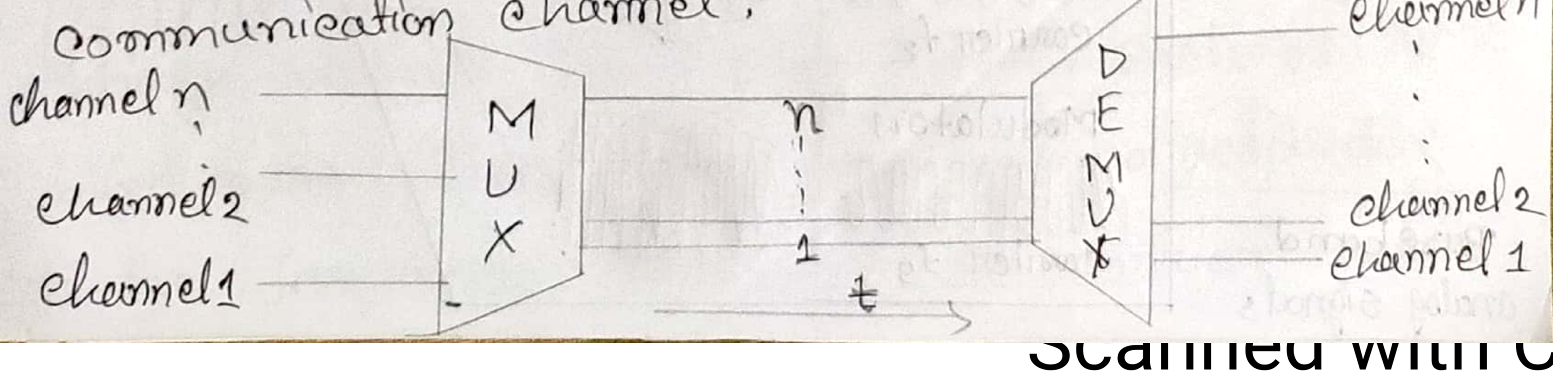


fig: Dividing a link into channel.



~~# FDM: Frequency division multiplexing (FDM) is a technique of multiplexing which means combining more than one signal over a shared medium. In FDM, signal of different frequencies are combined for concurrent transmission.~~

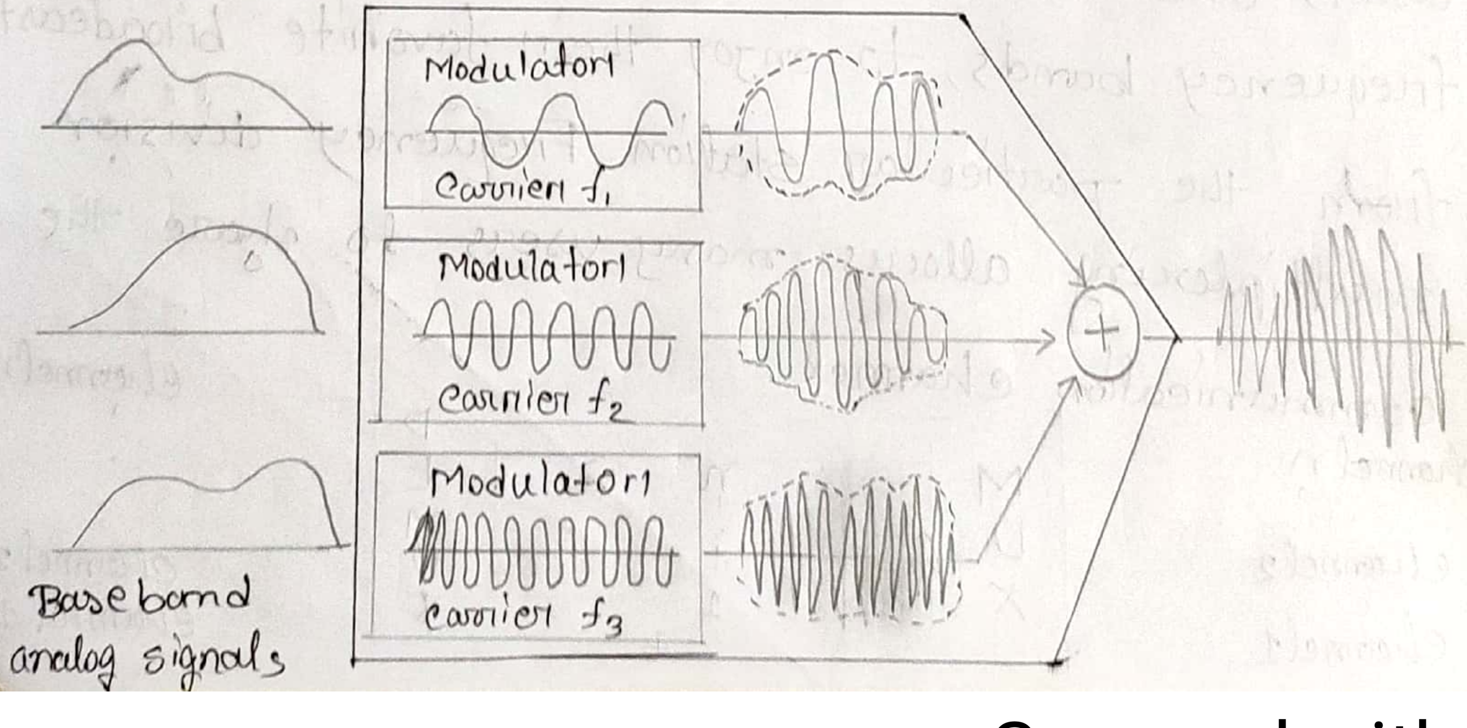
Frequency Division Multiplexing (FDM): FDM is a networking method of sharing the total available bandwidth of any communication channel by dividing then into many non-overlapping bands of frequency. Let us consider an example. The FDM can be compared to the operation of radio broadcasting where multiple frequencies are shared among users and one can tune to any of the available frequency bands, to enjoy their favorite broadcast from the particular station. Frequency division multiplexing allows many users to share the communication channel.



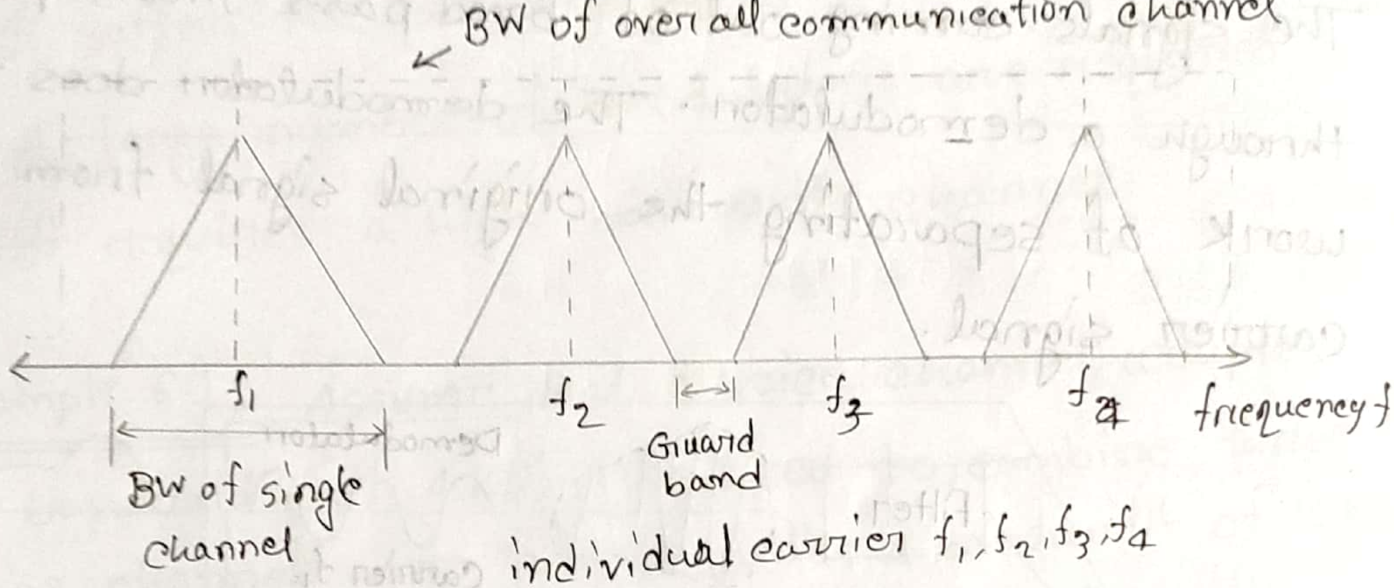
FDM Process: FDM Transmitters:

FDM is a technique in which the available bandwidth of a ~~signal~~ single transmission medium is subdivided into several frequency channels. The input signals are translated into frequency bands by using modulation techniques and they are combined by a multiplexer to form a composite signal.

The main aim of the FDM is to subdivide the available bandwidth into different frequency channels and allocate them to different ~~devices~~ modulator. The carriers which are used for modulating the signals are known as sub-carriers. They are represented as $f_1, f_2, f_3 \cdots f_n$. FDM is mainly used in radio broadcasts and TV network.



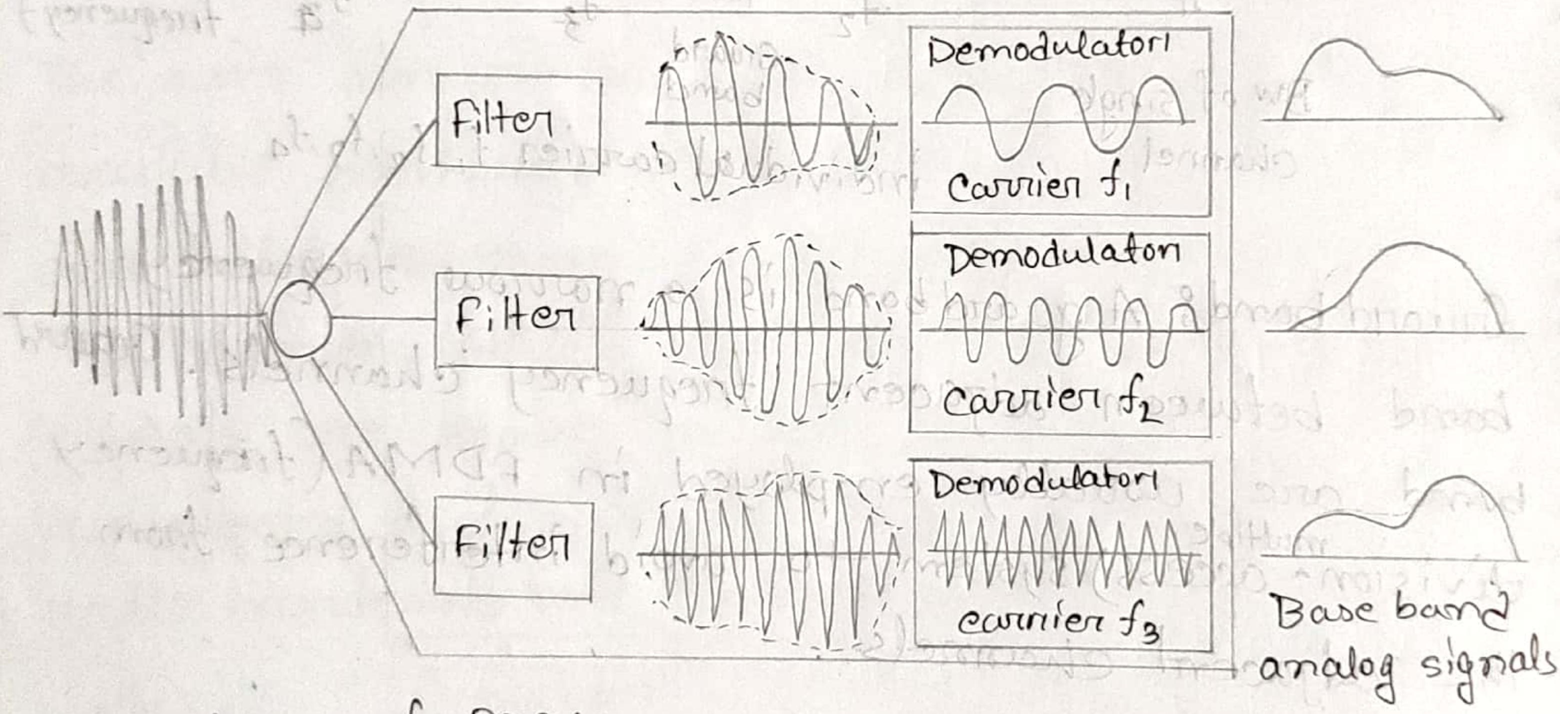
Spectrum of FDM signal: Signals are transmitted simultaneously with each occupying a different frequency slot within a common bandwidth.



Guard band: A guard band is a narrow frequency band between adjacent frequency channels. Guard band are usually employed in FDMA (frequency division multiple access) system to avoid interference from the adjacent channels.

FDM Receivers: At the receiving end, the single composite signal is received by the FDM receiver. The receiver then passes the composite signal through various band pass filters. Each of these band pass filters has a frequency corresponding to the frequencies of one of the carrier waves.

Each band pass filter will accept the signal whose frequency matchs with the frequency of the carrier signal and rejects all ~~over~~ other channels. The signals coming out of band pass filters pass through a demodulator. The demodulator does the work of separating the original signal from the carrier signal.



Advantages of FDM:

- FDM is used for analog signals.
- FDM process is very simple and easy modulation.
- A large number of signals can be sent through an FDM simultaneously.
- It does not require any synchronization between sender and receiver.

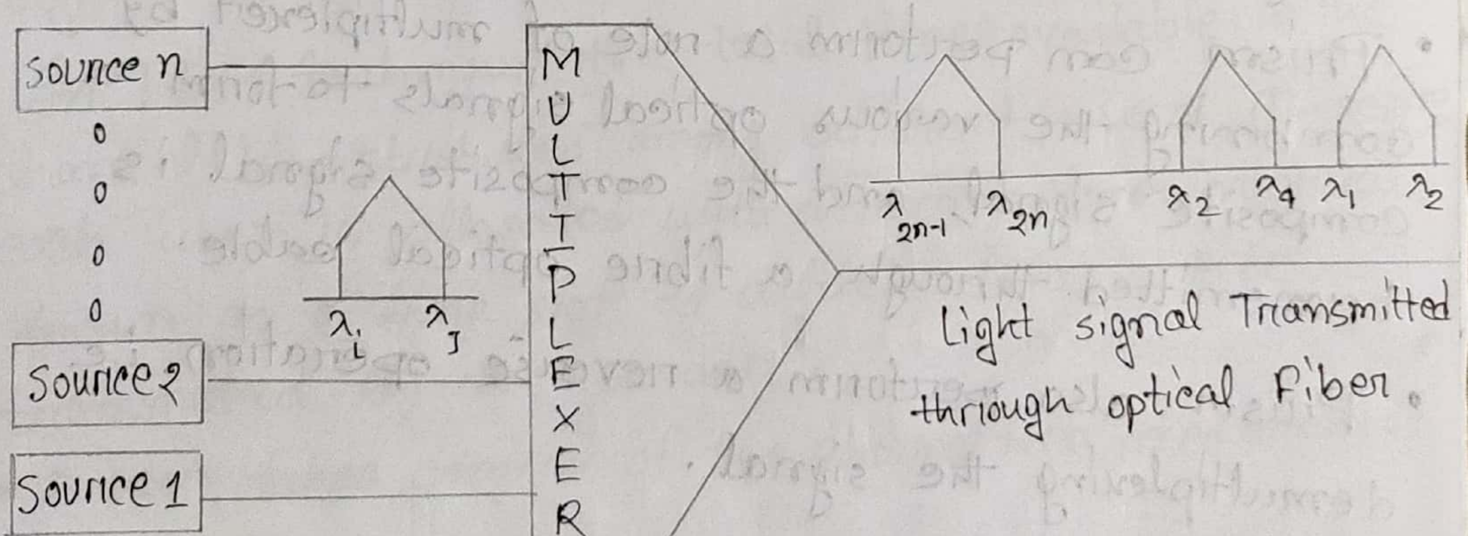
Disadvantages of FDM:

- FDM technique is used only when low-speed channels are required.
- It suffers the problem of crosstalk.
- A large number of modulators are required.
- It requires a high bandwidth channel.

Example 6.1: Assume that a voice channel occupies a bandwidth of 4 KHz. We need to combine three voice channels into a link with a bandwidth of 12 KHz, from 20 to 32 KHz. Show the configuration, using the frequency domain. Assume there are no guard band.

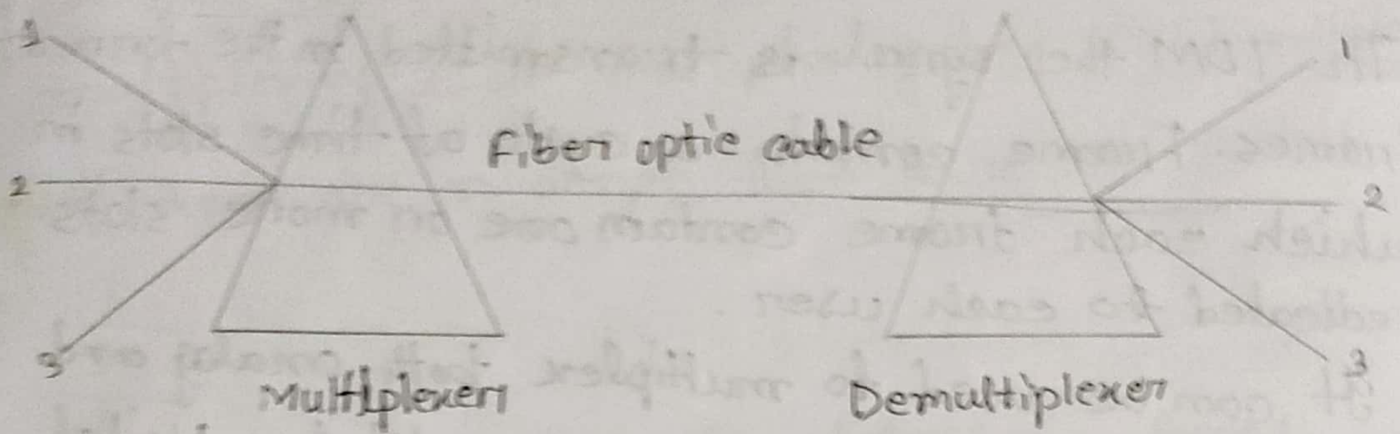
Example 6.2:

Wavelength Division Multiplexing (WDM):



WDM Transmitter

- Wavelength Division Multiplexing is same as FDM except that the optical signals are transmitted through the fibre optic cable.
- WDM is used on fibre optics to increase the capacity of a single fibre
- It is used to utilize the high data rate capability of fibre optic cable.
- It is an analog multiplexing technique.
- Optical signal from different source are combined to form a wider band of light with the help of multiplexer.
- At the receiving end, demultiplexer separates the signals to transmit them to their respective destinations.
- Multiplexing and Demultiplexing can be achieved by using a prism.
- Prism can perform a role of multiplexer by combining the various optical signals to form a composite signal, and the composite signal is transmitted through a fibre optical cable.
- Prism also perform a reverse operation, i.e. demultiplexing the signal.



Time Division Multiplexing (TDM):

- It is a digital technique.
- In frequency Division Multiplexing technique, the total time available in the channel is distributed among different users. Therefore, each user is allocated with different
- In FDM all signals operate at the same time with different frequency, but in case of ~~time~~ TDM all signals at the same frequency with different time.
- In TDM technique, the total time available in the channel is distributed among different users. Therefore each user is allocated with different time interval known as a time slot at which data is to be transmitted by the sender.
- A user takes control of the channel for a fixed amount of time.
- In TDM technique, data is not transmitted simultaneously

rather than data is transmitted one by one.

- In TDM the signal is transmitted in the form of frames. Frame contain a cycle of time slots in which each frame contain one or more slots dedicated to each user.
- It can be used to multiplex both analog and digital but mainly used to multiplex digital signals.

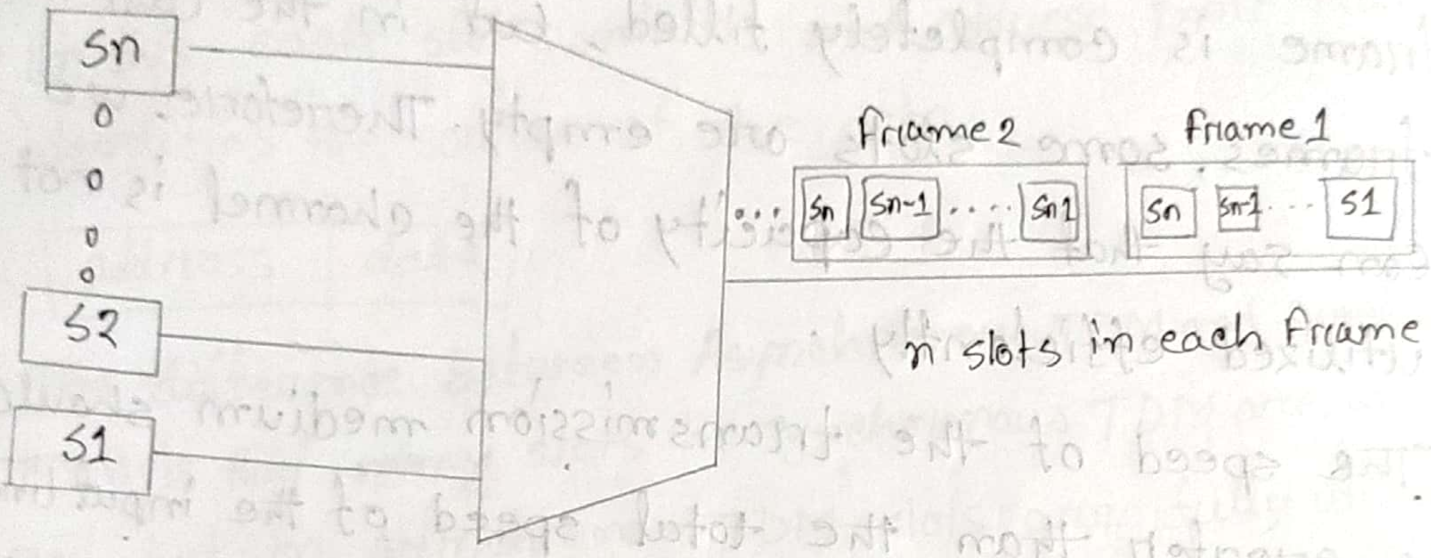
There are two types of TDM:

- Synchronous TDM.
- Asynchronous TDM.

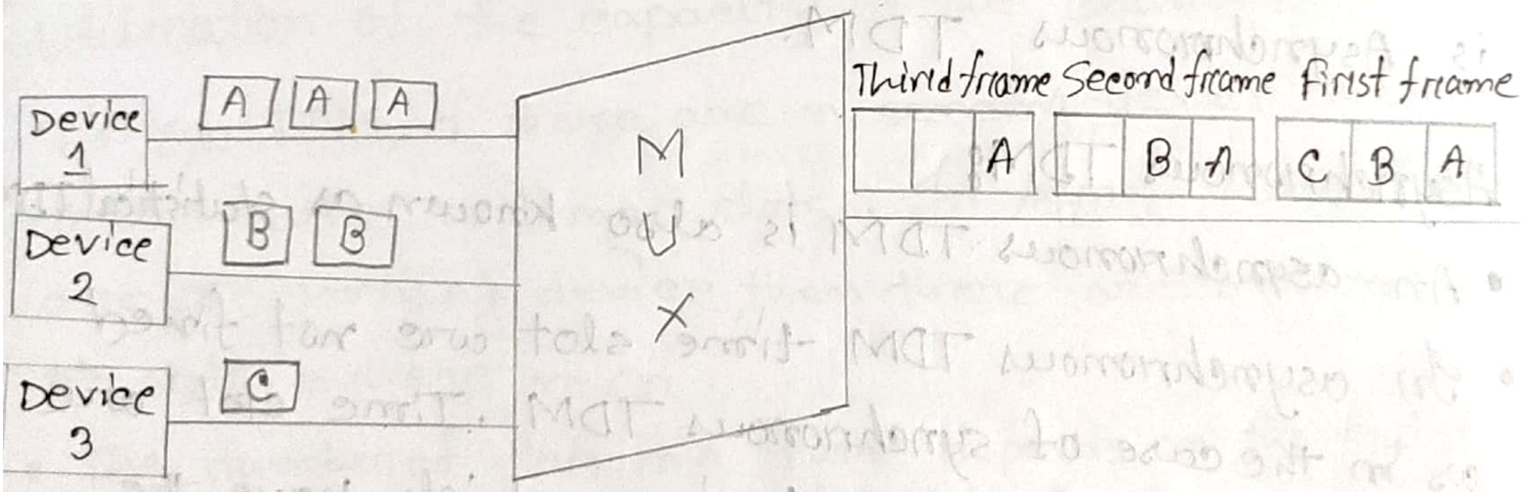
Synchronous TDM:

- A synchronous TDM is a technique in which time slot is preassigned to every device.
- In synchronous TDM, each device is given some time slot irrespective of the fact that the device contains the data or not.
- If the device does not have any data, then the slot will remain empty.
- In synchronous TDM, signals are sent in the form of frames. Time slots are organized in the form of frames. If a device does not have data for a particular time slot, then the empty slot will be transmitted.

- The most popular synchronous TDM are T-1 multiplexing, ISDN multiplexing and SONET multiplexing.
- If there are n device, then there are n slots.



concept of synchronous TDM



In the above figure, the synchronous TDM technique is implemented. Each device is allocated with some time slot. The time slots are transmitted irrespective of whether the sender has data to send or not.

Disadvantages of synchronous TDM:

- The capacity of the channel is not fully utilized as the empty slots are also transmitted which is having no data. In the above figure, the first frame is completely filled, but in the last two frames, some slots are empty. Therefore, we can say that the capacity of the channel is not utilized efficiently.
- The speed of the transmission medium should be greater than the total speed of the input lines. An alternative approach to the synchronous TDM is Asynchronous TDM.

Asynchronous TDM:

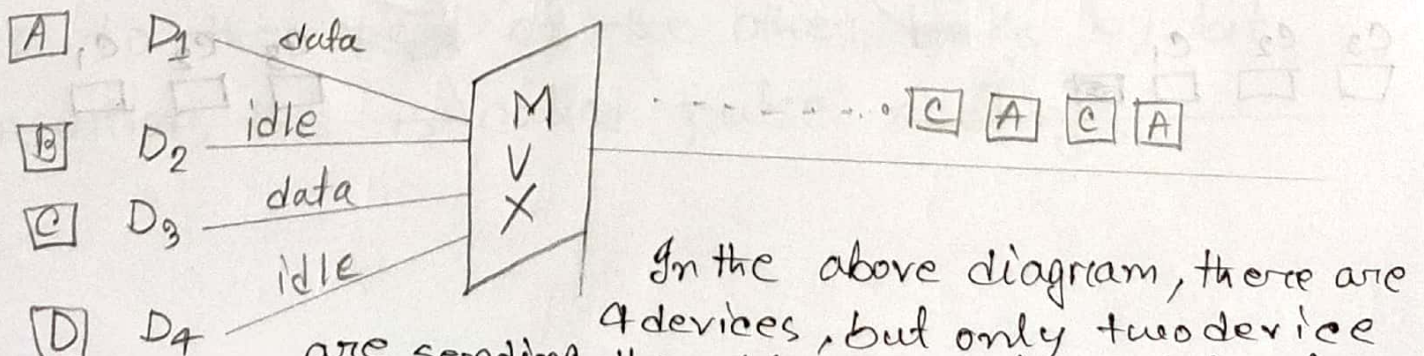
- An asynchronous TDM is also known as statistical TDM.
- In asynchronous TDM time slot are not fixed as in the case of synchronous TDM. Time slot are allocated to only those device which have the data to send. Therefore, we can say that asynchronous TDM transmits only the data from active workstation.
- It dynamically allocates the time slots to the devices.
- Total speed of the input line can be ~~greater~~ greater

than the capacity of the channel.

- ~~It ae~~ This accepts the incoming data streams and creates a frame that contains only data with no empty slots
- In this, each slot contains ~~an~~ address part that identifies the source of the data.

address	data

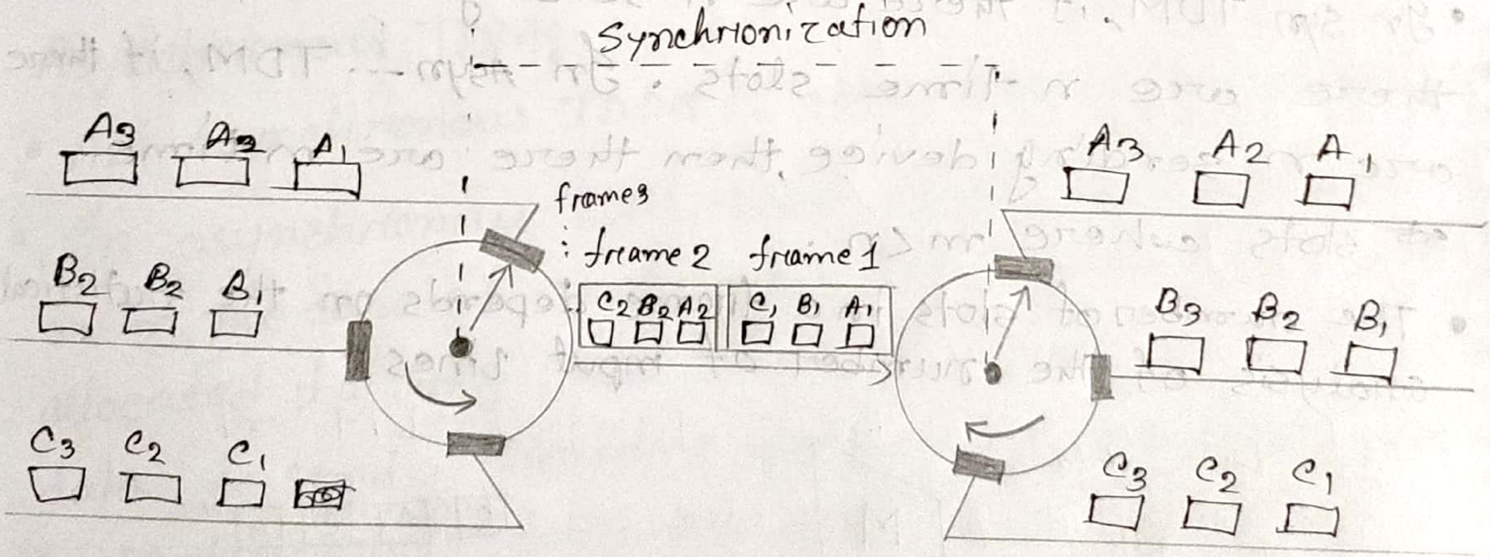
- The difference between Asynchronous TDM and Synchronous TDM is that many slots in Synchronous TDM are unutilized, but in asynchronous TDM, slots are fully utilized. This leads to smaller transmission time and efficient utilization of the capacity of the channel.
- In syn TDM, if there are n sending device, then there are n time slots. In Asyn... TDM, if there are n sending device, then there are m time ~~st~~ slots where $m < n$.
- The number of slots in a frame depends on the statistical analysis of the number of input lines.



In the above diagram, there are 4 devices, but only two device are sending the data. Therefore, data of A and C are only transmitted through the transmission line.

Interleaving: Interleaving synchronous TDM can be compared to a very fast rotating switch. As the switch open in front of a device, that device has the opportunity to send a specified amound (x bits) of data onto the path. The switch moves from device to device at a constant rate and in a fixed order. This process is called Interleaving.

Interleaving can be done by bit, by byte or any other data unit. It other words the multiplexer can take one byte from each device, then othe byte from each device and so on. In a given system the interleaving units will always be of same size.



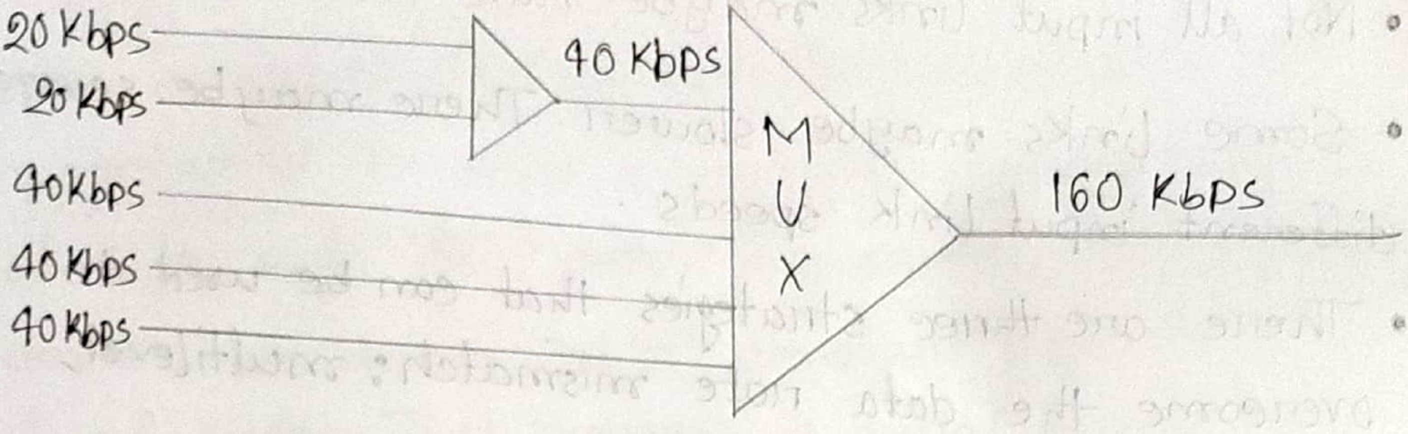
Data Rate Management:

- Not all input links maybe have the same data rate.
- Some links maybe slower. There maybe several different input link speeds.
- There are three strategies that can be used to overcome the data rate mismatch: multilevel, multislot and pulse stuffing.

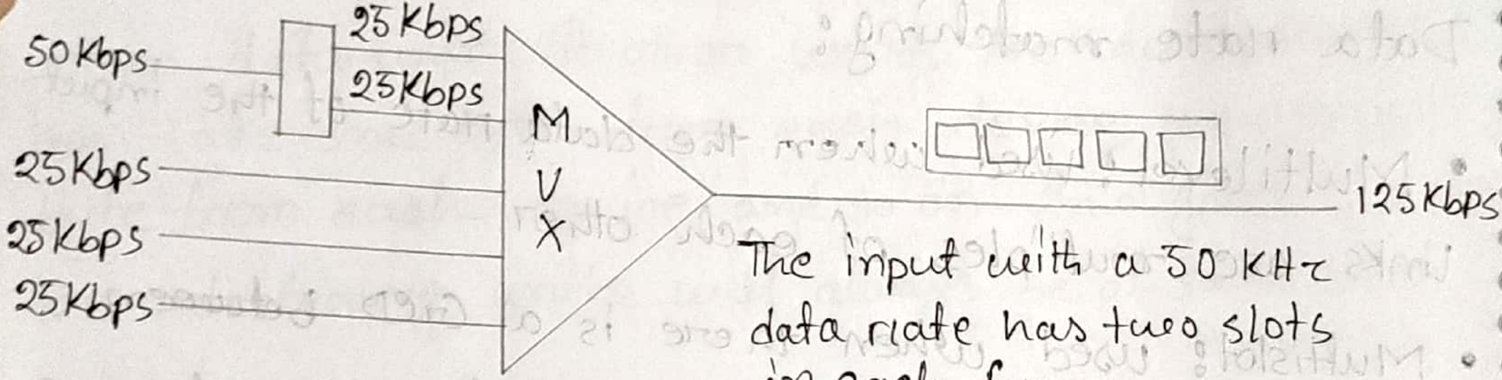
Data rate matching:

- Multilevel: used when the data rate of the input links are multiples of each other.
- Multislot: used when there is a GCD between the data rates. The higher bit rate channels are allocated more slots per frame, and the output frame rate is a multiple of each input link.
- Pulse Stuffing: used when there is no GCD between the links. The slowest speed link will be brought up to the speed of the other links by bit insertion, this is called pulse stuffing.

Multilevel multiplexing:

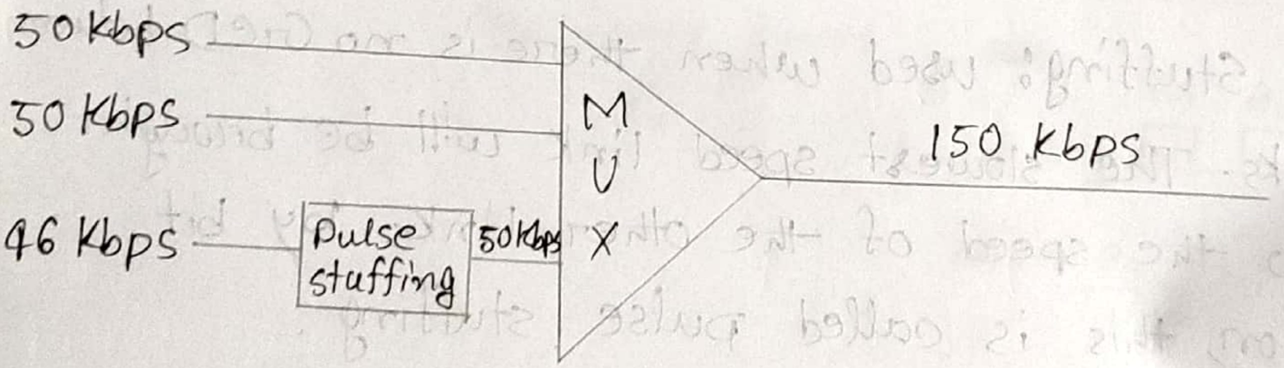


Multiple-slot multiplexing:



The input with a 50 KHz data rate has two slots in each frame.

Pulse stuffing:



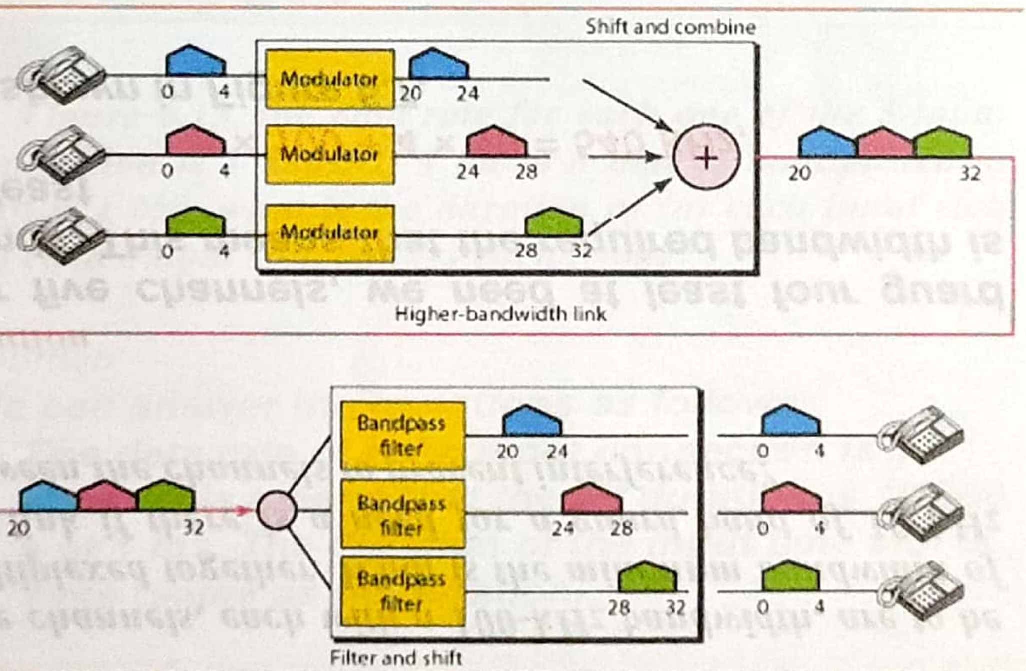
Example 6.1

Assume that a voice channel occupies a bandwidth of 4 kHz. We need to combine three voice channels into a link with a bandwidth of 12 kHz, from 20 to 32 kHz. Show the configuration, using the frequency domain. Assume there are no guard bands.

Solution

We shift (modulate) each of the three voice channels to a different bandwidth, as shown in Figure 6.6. We use the 20- to 24-kHz bandwidth for the first channel, the 24- to 28-kHz bandwidth for the second channel, and the 28- to 32-kHz bandwidth for the third one. Then we combine them as shown in Figure 6.6.

Figure 6.6 *Example 6.1*



Example 6.2

Five channels, each with a 100-kHz bandwidth, are to be multiplexed together. What is the minimum bandwidth of the link if there is a need for a guard band of 10 kHz between the channels to prevent interference?

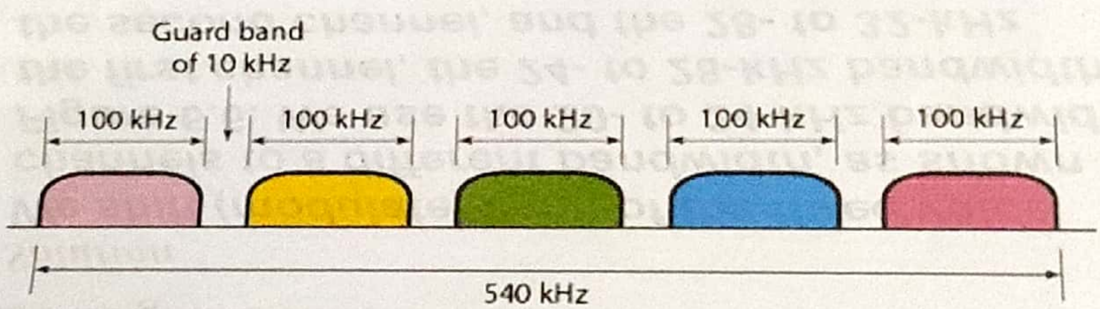
Solution

For five channels, we need at least four guard bands. This means that the required bandwidth is at least

$$5 \times 100 + 4 \times 10 = 540 \text{ kHz},$$

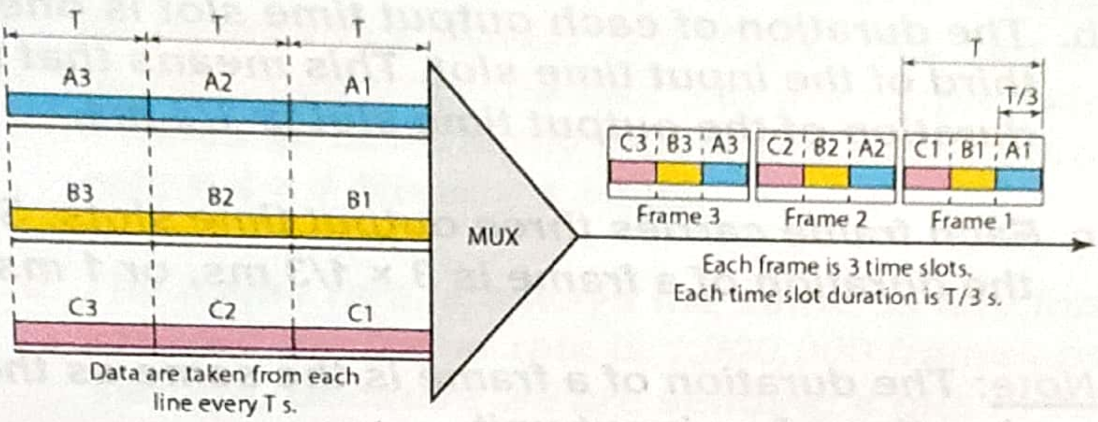
as shown in Figure 6.7.

Figure 6.7 ***Example 6.2***



F.

Figure 6.13 *Synchronous time-division multiplexing*



Example 6.5

In Figure 6.13, the data rate for each one of the 3 input connection is 1 kbps. If 1 bit at a time is multiplexed (a unit is 1 bit), what is the duration of (a) each input slot, (b) each output slot, and (c) each frame?

Solution

We can answer the questions as follows:

a. The data rate of each input connection is 1 kbps. This means that the bit duration is 1/1000 s or 1 ms. The duration of the input time slot is 1 ms (same as bit duration).

Example 6.5 (continued)

b. The duration of each output time slot is one-third of the input time slot. This means that the duration of the output time slot is 1/3 ms.

c. Each frame carries three output time slots. So the duration of a frame is 3 × 1/3 ms, or 1 ms.

Note: The duration of a frame is the same as the duration of an input unit.

Example 6.6

Figure 6.14 shows synchronous TDM with 4 1Mbps data stream inputs and one data stream for the output. The unit of data is 1 bit. Find (a) the input bit duration, (b) the output bit duration, (c) the output bit rate, and (d) the output frame rate.

Solution

We can answer the questions as follows:

a. The input bit duration is the inverse of the bit rate:

1/1 Mbps = 1 μs.

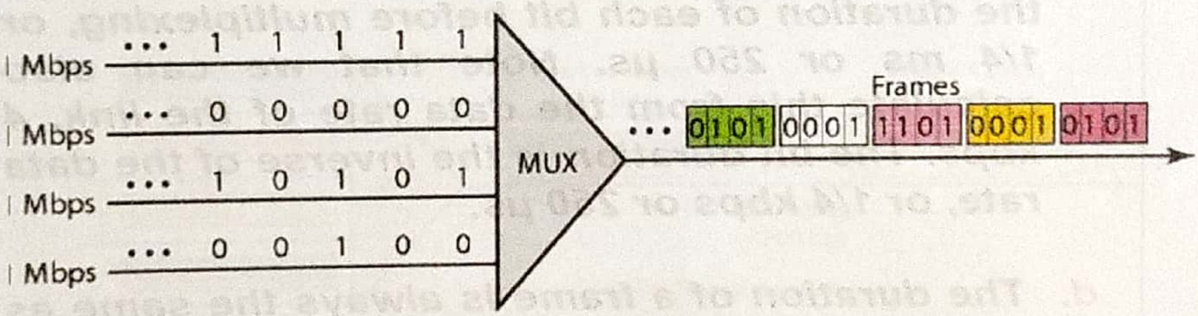
b. The output bit duration is one-fourth of the input bit duration, or ¼ μs.

Example 6.6 (continued)

c. **The output bit rate is the inverse of the output bit duration or 1/(4μs) or 4 Mbps. This can also be deduced from the fact that the output rate is 4 times as fast as any input rate; so the output rate = 4 × 1 Mbps = 4 Mbps.**

d. **The frame rate is always the same as any input rate. So the frame rate is 1,000,000 frames per second. Because we are sending 4 bits in each frame, we can verify the result of the previous question by multiplying the frame rate by the number of bits per frame.**

Figure 6.14 *Example 6.6*



Example 6.7

Four 1-kbps connections are multiplexed together. A unit is 1 bit. Find (a) the duration of 1 bit before multiplexing, (b) the transmission rate of the link, (c) the duration of a time slot, and (d) the duration of a frame.

Solution

We can answer the questions as follows:

a. **The duration of 1 bit before multiplexing is 1 / 1 kbps, or 0.001 s (1 ms).**

b. **The rate of the link is 4 times the rate of a connection, or 4 kbps.**

Example 6.7 (continued)

c. **The duration of each time slot is one-fourth of the duration of each bit before multiplexing, or 1/4 ms or 250 μs. Note that we can also calculate this from the data rate of the link, 4 kbps. The bit duration is the inverse of the data rate, or 1/4 kbps or 250 μs.**

d. **The duration of a frame is always the same as the duration of a unit before multiplexing, or 1 ms. We can also calculate this in another way. Each frame in this case has four time slots. So the duration of a frame is 4 times 250 μs, or 1**

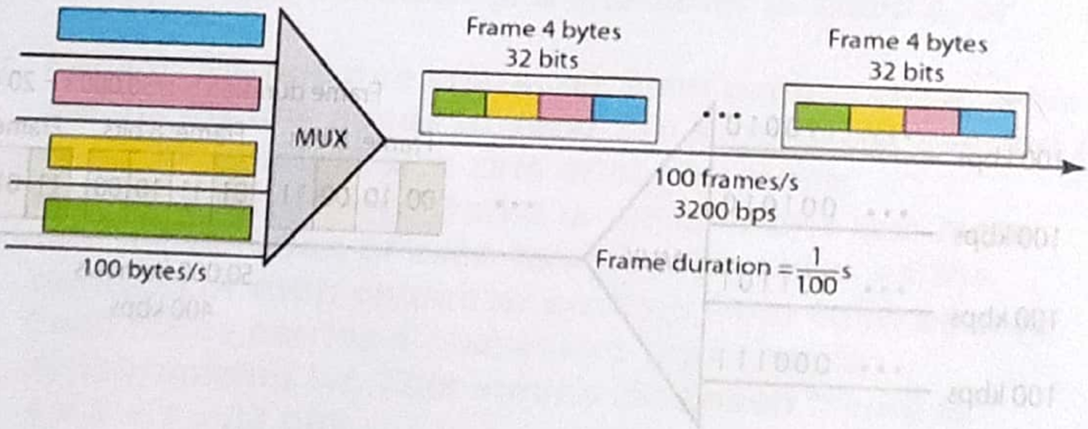
Example 6.8

Four channels are multiplexed using TDM. If each channel sends 100 bytes /s and we multiplex 1 byte per channel, show the frame traveling on the link, the size of the frame, the duration of a frame, the frame rate, and the bit rate for the link.

Solution

The multiplexer is shown in Figure 6.16. Each frame carries 1 byte from each channel; the size of each frame, therefore, is 4 bytes, or 32 bits. Because each channel is sending 100 bytes/s and a frame carries 1 byte from each channel, the frame rate must be 100 frames per second. The bit rate is 100 × 32, or 3200 bps.

Figure 6.16 *Example 6.8*



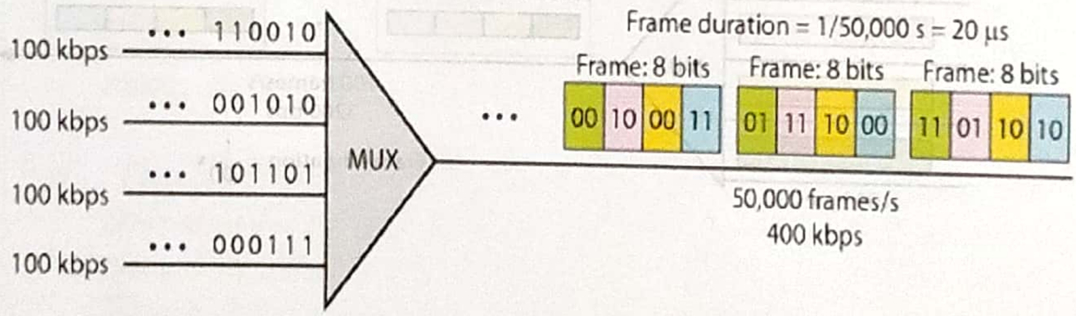
Example 6.9

A multiplexer combines four 100-kbps channels using a time slot of 2 bits. Show the output with four arbitrary inputs. What is the frame rate? What is the frame duration? What is the bit rate? What is the bit duration?

Solution

Figure 6.17 shows the output (4x100kbps) for four arbitrary inputs. The link carries 400K/(2x4)=50,000 2x4=8bit frames per second. The frame duration is therefore 1/50,000 s or 20 μs. The bit duration on the output link is 1/400,000 s, or 2.5 μs.

Figure 6.17 *Example 6.9*



framing bits

Example 6.10

We have four sources, each creating 250 8-bit characters per second. If the interleaved unit is a character and 1 synchronizing bit is added to each frame, find (a) the data rate of each source, (b) the duration of each character in each source, (c) the frame rate, (d) the duration of each frame, (e) the number of bits in each frame, and (f) the data rate of the link.

Solution

We can answer the questions as follows:

a. The data rate of each source is $250 \times 8 = 2000$ bps = 2 kbps.

Example 6.10 (continued)

b. Each source sends 250 characters per second; therefore, the duration of a character is 1/250 s, or 4 ms.

c. Each frame has one character from each source, which means the link needs to send 250 frames per second to keep the transmission rate of each source.

d. The duration of each frame is 1/250 s, or 4 ms. Note that the duration of each frame is the same as the duration of each character coming from each source.

e. Each frame carries 4 characters and 1 extra synchronizing bit. This means that each frame is $4 \times 8 + 1 = 33$ bits.

Example 6.11

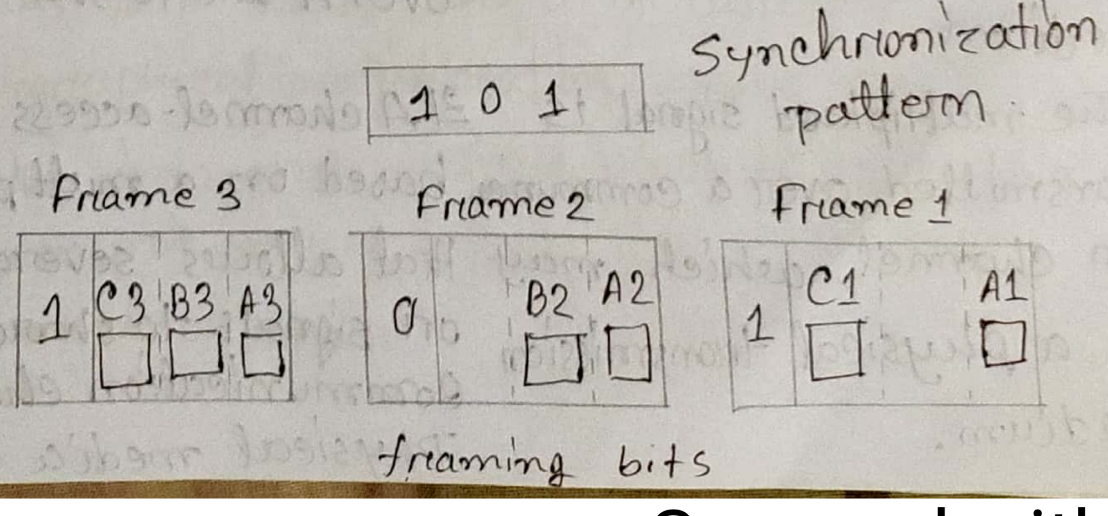
Two channels, one with a bit rate of 100 kbps and another with a bit rate of 200 kbps, are to be multiplexed. How this can be achieved? What is the frame rate? What is the frame duration? What is the bit rate of the link?

Solution

We can allocate one slot to the first channel and two slots to the second channel. Each frame carries 3 bits. The frame rate is 100,000 frames per second because it carries 1 bit from the first channel. The bit rate is 100,000 frames/s × 3 bits per frame, or 300 kbps.

Synchronization: The TDM needs synchronization between multiplexer and demultiplexer. If synchronization is not there between mux and demux, a bit going to one channel may be recived by the wrong channel. Because of this reason one or more synchronization bits are usually added to beginning of each frame. These bits are called framing bits, allows demux to synchronize with the incoming stream so that it can separate time slot accurately, in most of the cases, this synchronization information consists of 1 bit per frame, changing between 0 and 1.

- The reciver ~~f~~ Looks for the anticipated bit and starts counting bits till the end of the frame.
- Then it starts over again with reception of another known bit.



Bandwidth Utilization: Multiple Access Techniques

Multiple Access is the use of multiplexing techniques to provide communication service to multiple users over a single channel. It allows for many users at one time by sharing a finite amount of spectrum.

multiple access technique permits the communication resources of the channel to be shared by a large number of users seeking to communicate with each other.

Multiplexing	Multiple Access
1) In telecommunications and computer networks, multiplexing is a process where multiple analog message signals or digital data streams are combined into one signal over a shared medium.	1) In telecommunications and computer networks, a channel access method or multiple access method allows several terminals connected to the same multipoint physical medium to transmit over it and to share it's capacity
2) The multiplexed signal is transmitted over a communication channel, which may be a physical transmission medium.	2) A channel-access scheme is based on a multiplex method that allows several data streams or signals to share the same communication channel or Physical media.

3/ A device that perform the multiplexing is called a multiplexer (MUX) and a device that performs the reverse process is called a demultiplexer (DEMUX).	3/ A channel-access scheme is also based on a Multiple access protocol and control mechanism, also known as media access control (MAC). This protocol deals with issues such as addressing, assigning multiplex channels to different users, and avoiding collisions.
4/ It works on the physical layer.	4/ It works on the Data Link layer.

Basic types of multiple access:

1/ FDMA: frequency Division Multiple Access
2/ TDMA: Time Division Multiple Access.
3/ CDMA: Code Division Multiple Access.
4/ SDMA: Space Division Multiple Access.

Difference between Multiple Access and Multiplexing

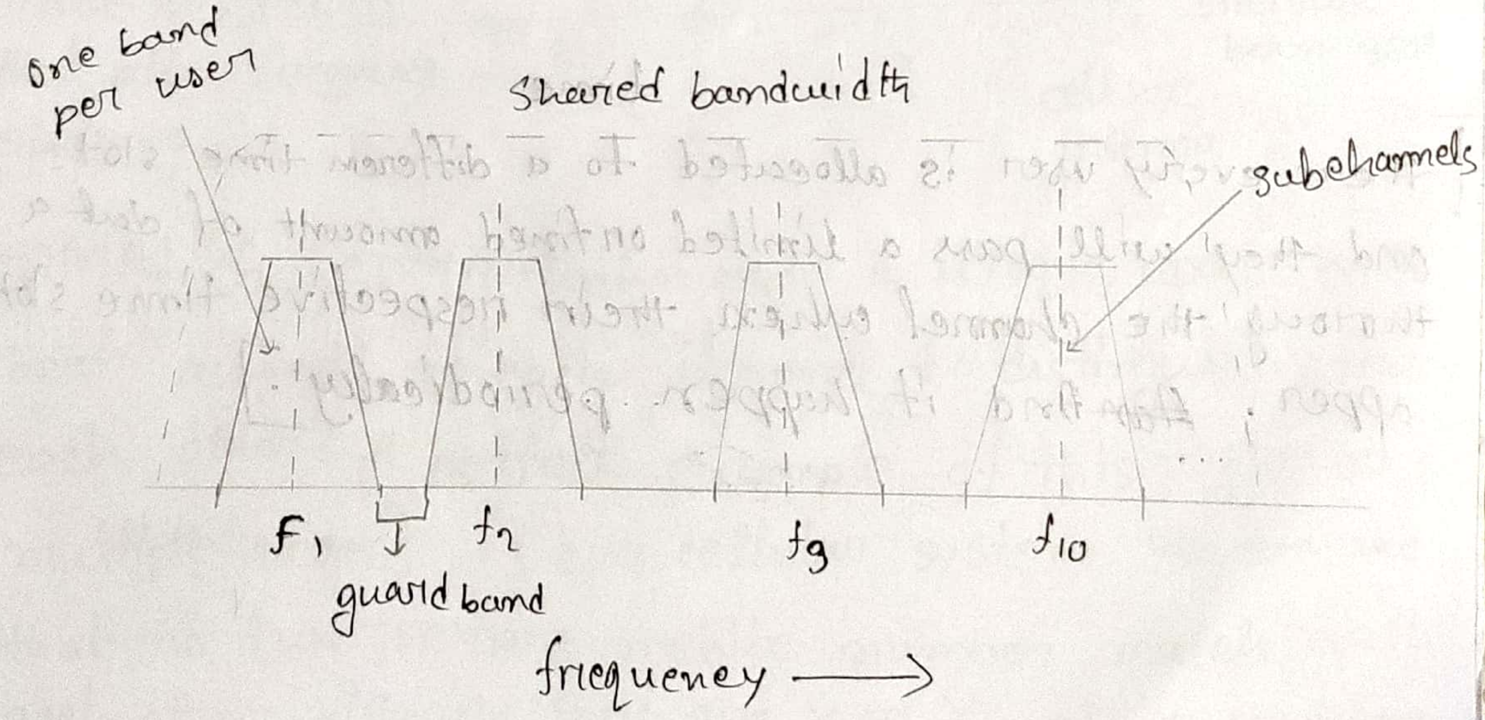
Multiplexing	Multiple Access
Multiple signals form users at same geographical location are combined into one signal to transmit over a single channel or media.	Users in different geographical locations share the use of a single channel or media to transmit over it.

Scanned with Ca

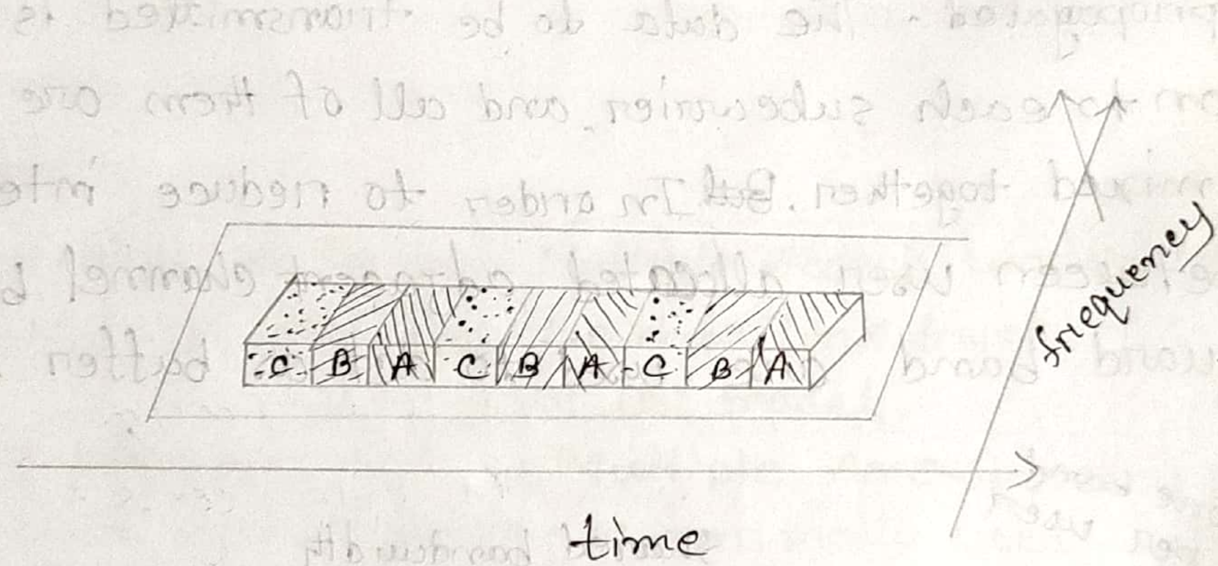
Multiplexing involves transmitting multiple signals and streams simultaneously.	Multiple access is based on schemes such as frequency division, time division and code division.
Combine several low speed signals (voice, data etc.) for transmission over a single high speed connection	Different users share the common high speed connection.
A centralized entity handles the communications from different users. These communications are organized into a frame by said entity.	The access to the resources is performed on a de-centralized basis.
Multiplexing works on the physical layer (L1) of OSI model.	Multiple Access works on the data link layer (L2) of OSI model.
Multiplexing techniques are normally used with telephone fiber optic channel and radio channels.	Multiple Access techniques are normally used with satellite channels.
There is no possibility that a transmitted channel will damage other channels because ~~other channel~~ they are transmitted from the same channel	There is always the possibility of one of the transmitted channels interfering and causing damage to other transmitted signals.

user requirements are ordinarily fixed.	User requirements can change dynamically with time.

FDMA: FDMA is the process of dividing one channel or bandwidth into multiple individual bands, each for use by a single user. Each individual band or channel is wide enough to accommodate the signal spectra of the transmission to be propagated. The data to be transmitted is modulated on to each subcarrier and all of them are linearly mixed together. ~~But~~ In order to reduce interference between user allocated adjacent channel band, guard band are used to act as buffer zones.



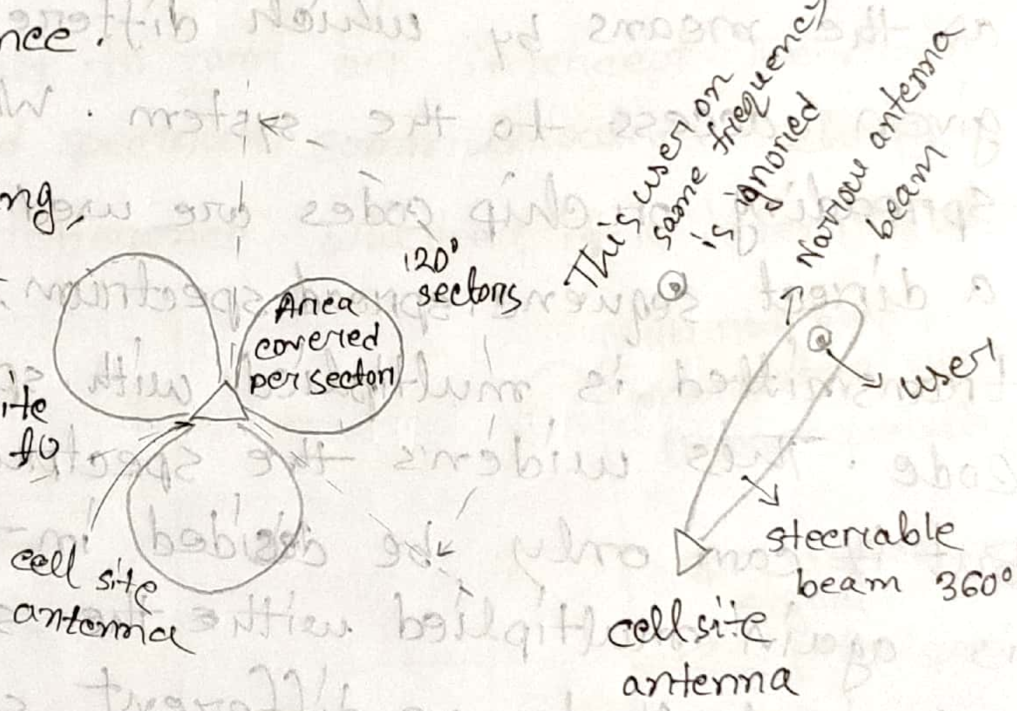
TDMA: TDMA is a digital technique that divides a single channel or band into time slots. Each time slot is used to transmit one byte or another digital segment of each signal is sequential serial data format. This technique works well with slow voice data signals, but it's also useful for compressed video and other high speed data.



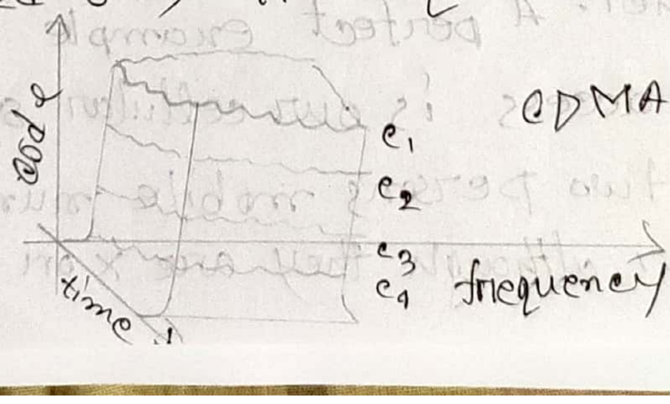
[Here every user is allocated to a differen time slot. and they will pass a limited or fixed amount of data through the channel when their respective time slot apper. ~~Aon~~ And it happen periodically.]

SDMA: SDMA uses physical separation methods that permit the sharing of wireless channels. For instance, a single channel may be used simultaneously if the users are spaced far enough from one another to avoid interface. Known as frequency reuse, the method is widely used in cellular radio system: cell sites are spaced from one another to minimize interference.

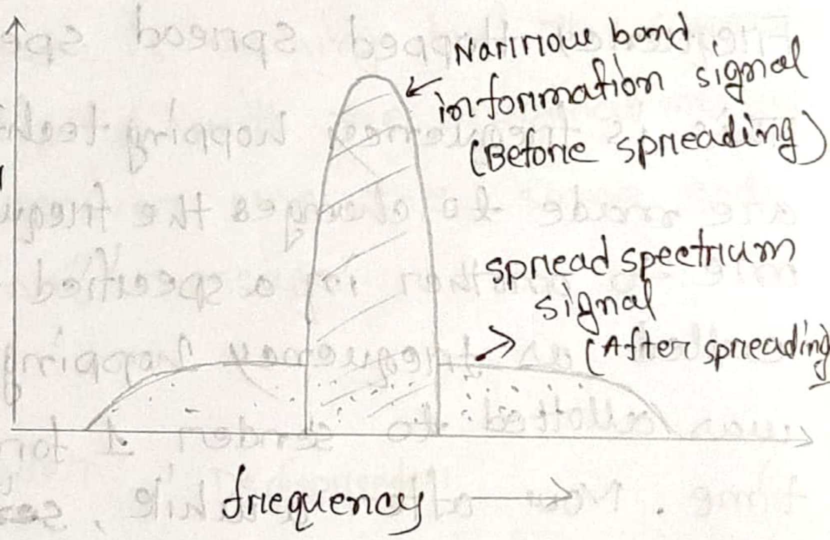
In addition to spacing, directional antennae are used to avoid interface. Most cell site use three antennas to create 120° sectors that allow frequency sharing.



CDMA: In CDMA technique, a unique code has been assigned to each channel to distinguish from each other. A perfect example of this type of multiple access is our cellular system. We can see that no two person's mobile number match with each other although they are X or Y mobile service providing companys customers using the same bandwidth. When we extracting the required data from a DSSS signal it was necessary to have correct spreading or chip code and all other data from the sources using different orthogonal chip codes would be rejected. It is therefore possible to allocate different codes, and use this as the means by which different users are given access to the system. With CDMA different spreading or chip codes are used. when generating a direct sequenc spread spectrum, the data to be transmitted is multiplied with spreading or chip code. This widen's the spectrum of the signal, But it can only be decided in the reciver if it is again multiplied with the same chip code. All signals that use different spreading codes are not seen and are discarded in the process. Thus in the presence of a variety of signals it is possible to recived only the required one.



Spread spectrum System: Spread spectrum refers to system that provide secure communication by spreading the signal over a large frequency band. The idea behind spread spectrum is to use more bandwidth than the original message while maintaining the same signal power. A spread signal does not have a clearly distinguishable peak in the spectrum. This makes the signal more dificult to distinguish from noise and therefore difficult to jam or intercept. The main advantage of spread spectrum communication technique is to prevent "interference" whether it is intentional or unintentional.



Narrow band signal:

① Band of signal accupy a narrow range of frequency

② Power density is high.

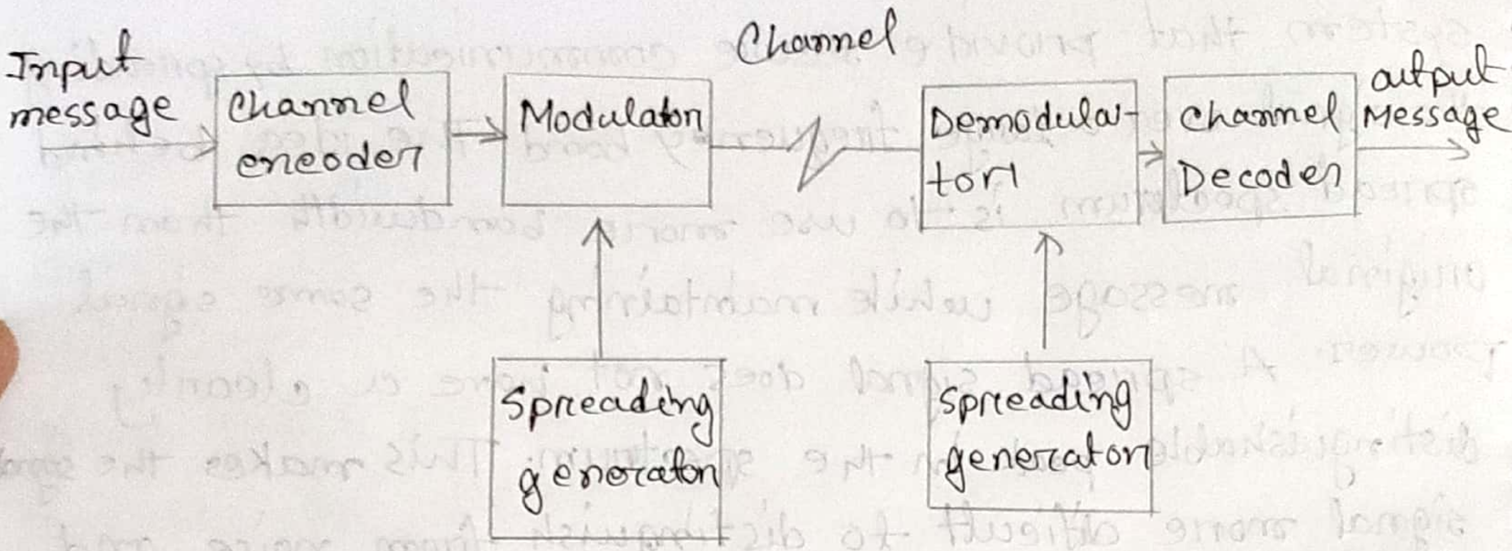
③ spread of energy is low and concentrated.

Spread spectrum Signal:

① Band of signal accupy a wide range of frequency

② Power density is very low.

③ Energy is wide spread.



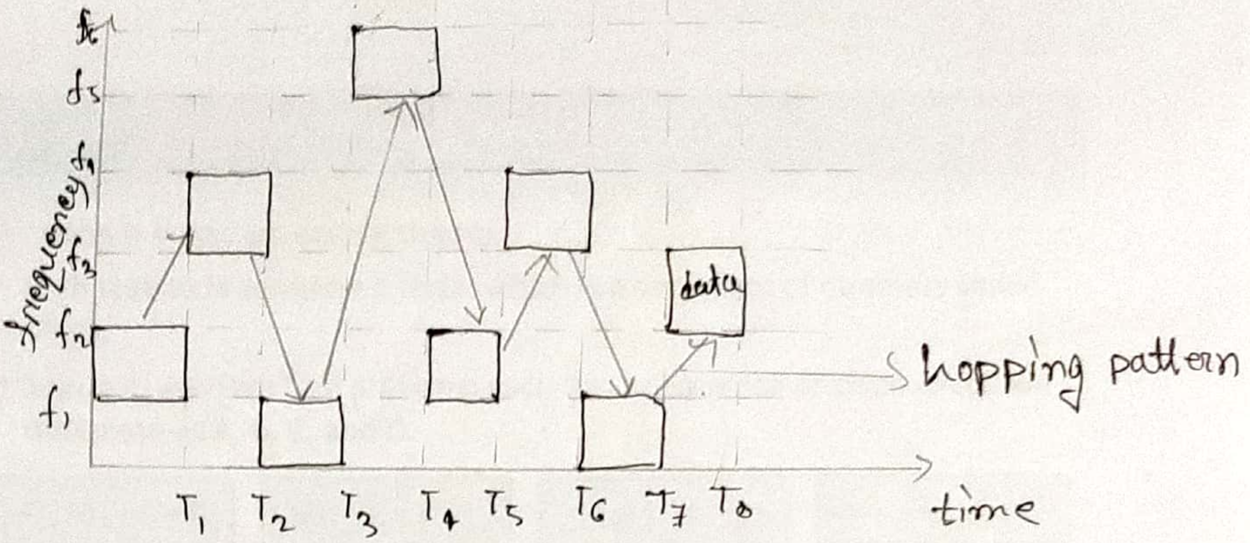
There are two important type of spread spectrum system;

① Frequency - hopping.

② Direct - sequence.

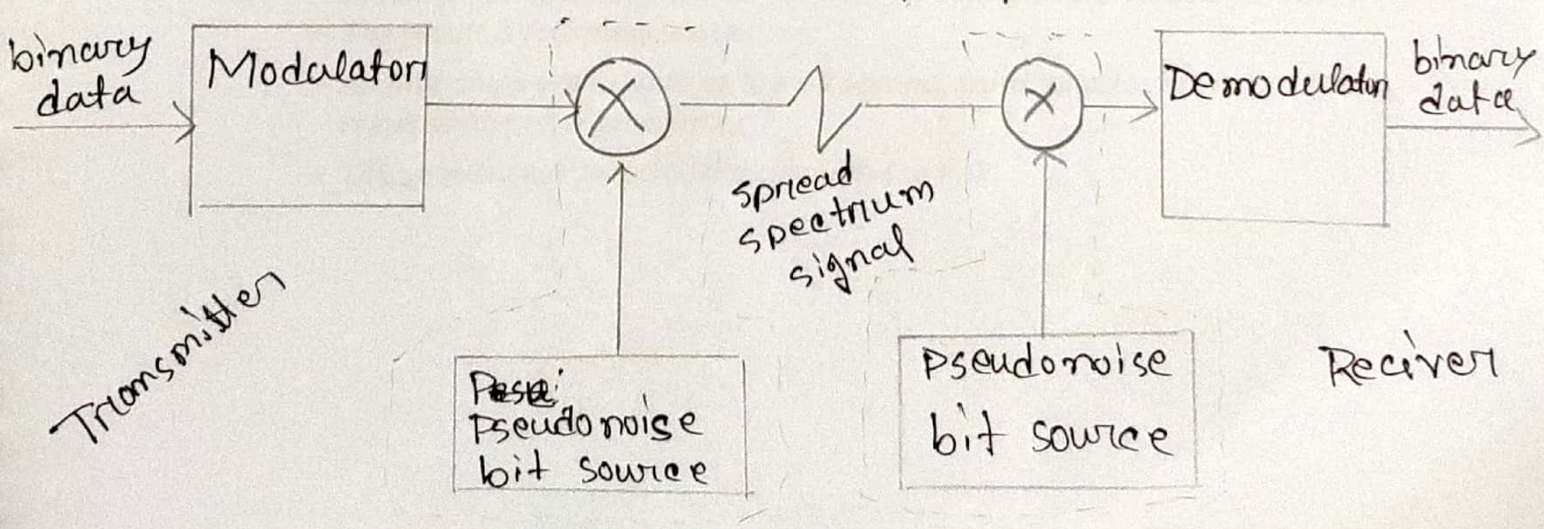
Frequency Hopped spread spectrum (FHSS):

This is frequency hopping technique, where the users are made to changes the frequencies of usage from one to another in a specified time interval hence called as frequency hopping. for example, a frequency was allotted to sender 1 for a particular period of time. Now after a while, ~~sender~~ user 1 hops to the other frequency and sender 2 uses the first frequency which was previously used by sender 1. This is call frequency reused. The frequency of the data are hopped from one to another in order to provide a secure transmission. The amount of time spent on each frequency hop is called as Duell time.



Direct sequence Spread spectrum (DSSS):

when ever user want to send data using this DSSS technique, each and every bit of the user data is multiplied by a secret code, called as chipping code. This chipping code is nothing but the spreading code which is multiplied with the original message and transmitted. The reciver use the same code to retrieve the original message.



Chips and Encoding

- CDMA is based on coding theory.
- Each station is assigned a code, which is a sequence of numbers called *Chips*.
- Suppose we have four stations; each has a sequence of chips which we designate as A, B, C, and D.

A	B	C	D
+1, +1, +1, +1	+1, −1, +1, −1	+1, +1, −1, −1	+1, −1, −1, +1

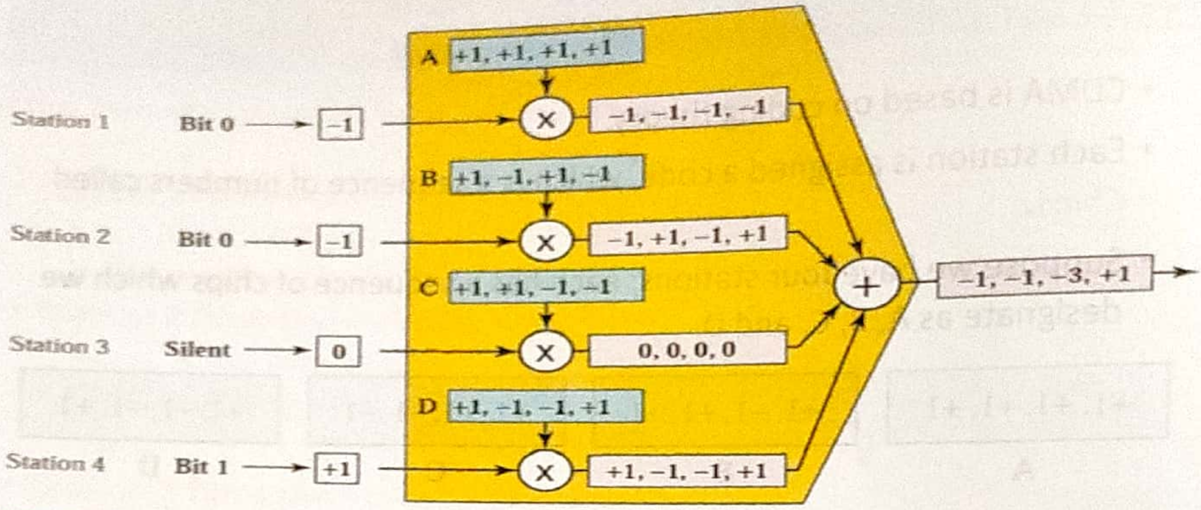
We follow these rules for encoding:

- *If a station needs to send a 0 bit, it sends a -1;*
- *If it needs to send a 1 bit, it sends a +1.*
- *When a station is idle, it sends no signal, which is represented by a 0.*

Data bit 0 ⟶ **−1** **Data bit 1** ⟶ **+1** **Silence** ⟶ **0**

Multiplexer

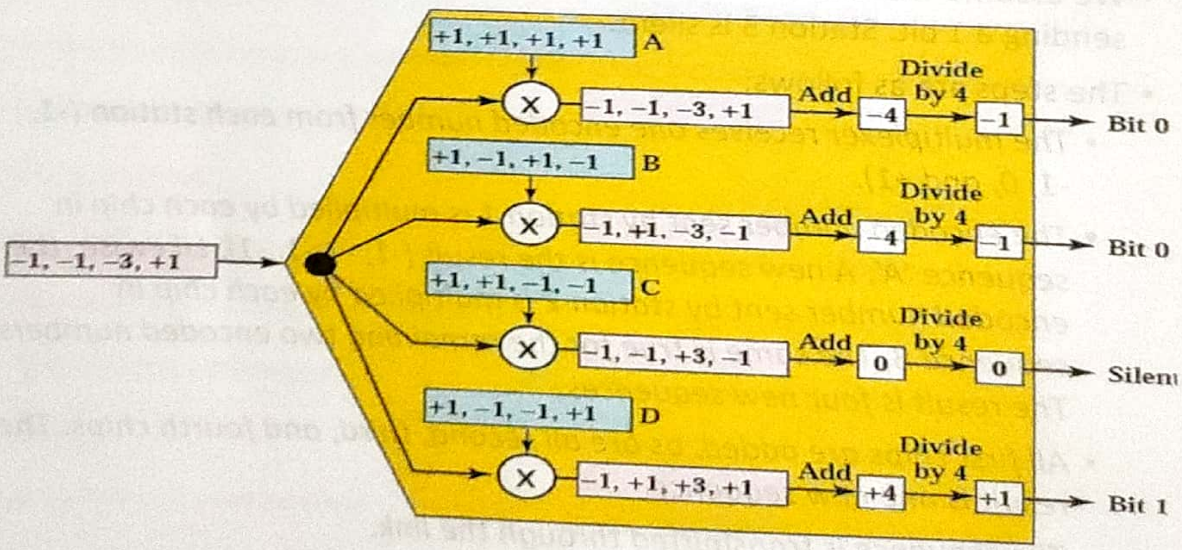
- As a simple example, we show how four stations share the link during 1-bit interval. The procedure can easily be repeated for additional intervals.
- We assume that stations 1 and 2 are sending a 0 bit and channel 4 is sending a 1 bit. Station 3 is silent.
- The steps are as follows:
    - *The multiplexer receives one encoded number from each station (-1, -1, 0, and +1).*
    - *The encoded number sent by station 1 is multiplied by each chip in sequence 'A'. A new sequence is the result (-1, -1,-1, -1). Likewise, the encoded number sent by station 2 is multiplied by each chip in sequence B. The same is true for the remaining two encoded numbers. The result is four new sequences.*
    - *All first chips are added, as are all second, third, and fourth chips. The result is one new sequence.*
    - *The sequence is transmitted through the link.*



Demultiplexer

The steps at the demultiplexer are as follows:

- The demultiplexer receives the sequence sent across the link.
- It multiplies the sequence by the code for each receiver. The multiplication is done chip by chip.
- The chips in each sequence are added. The result is always +4, -4, or 0.
- The result of step 3 is divided by 4 to get -1, +1, or 0.
- The number in step 4 is decoded to 0, 1, or silence by the receiver.



Orthogonal Sequences

- Orthogonal Sequences let us return to the chip sequences.
- The sequences are not chosen randomly; they were carefully selected.
- The sequences are called orthogonal sequences.

Sequence Generation

- To generate sequences, we use a Walsh table, a two-dimensional table with an equal number of rows and columns.
- Each row is a sequence of chips. The Walsh table W_1 for a one-chip sequence has one row and one column.
- We can choose -1 or +1 for the chip.
- According to Walsh, if we know the table for N sequences W_N we can create the table for $2N$ sequences W_{2N}.

Walsh Table

The W_N with the overhead bar stands for the complement of W_N where each +1 is changed to -1 and vice versa.

$$W_1 = \begin{bmatrix} +1 \end{bmatrix} \qquad W_{2N} = \begin{bmatrix} W_N & W_N \\ W_N & \overline{W_N} \end{bmatrix}$$

Let us see how we can create W_2 and W_4 from W_1.

$$W_1 = \begin{bmatrix} +1 \end{bmatrix}$$

$$W_2 = \begin{bmatrix} +1 & +1 \\ +1 & -1 \end{bmatrix}$$

$$W_4 = \begin{bmatrix} +1 & +1 & +1 & +1 \\ +1 & -1 & +1 & -1 \\ +1 & +1 & -1 & -1 \\ +1 & -1 & -1 & +1 \end{bmatrix}$$

- *After we select W_1, W_2 can be made from four W_1's, with the last one the complement of W_1.*
- *After W_2 is generated, W_4 can be made of four W_2's, with the last one the complement of W_2*
- *Of course, W_8 is composed of four W_4's, and so on. According to the tables, the sequences for two stations accessing one link are +1, +1 and +1, -1.*

Properties of Orthogonal Sequences

Orthogonal sequences have properties that are suitable for CDMA. They are as follows:

- If we multiply a sequence by -1. every element in the sequence is complemented (+1 becomes -1 and -1 becomes +1). We can see that when a station is sending -1 (bit 0), it is sending its complement.
- If we multiply two sequences, element by element, and add the results, we get a number called the *inner product*. If the two sequences are the same, we get N, where *N* is the number of sequences; if they are different, we get 0. The inner product uses a dot as the operator. So *A.A* is N, but *A.B* is 0.
- The inner product of a sequence by its complement is -N.
  So *A .(-A)* is *-N*.

Example

Check to see if the second property about orthogonal codes holds for our previous CDMA example.

Solution:

The inner product of each code by itself is N. This is shown for code C; you can prove for yourself that it holds true for the other codes.

$$\begin{aligned} C\,.\,C &= [+1, +1, -1, -1]\,.\,[+1, +1, -1, -1] \\ &= 1 + 1 + 1 + 1 \\ &= 4 \end{aligned}$$

If two sequences are different, the inner product is 0.

$$\begin{aligned} B\,.\,C &= [+1, -1, +1, -1]\,.\,[+1, +1, -1, -1] \\ &= 1 - 1 - 1 + 1 \\ &= 0 \end{aligned}$$

Example

Check to see if the third property about orthogonal codes holds for our previous CDMA example.

Solution:

The inner product of each code by itself is N. This is shown for code C; you can prove for yourself that it holds true for the other codes.

$$C \cdot (-C) = [+1,\ +1,\ -1,\ -1] \cdot [-1,\ -1,\ +1,\ +1]$$
$$= -1 - 1 - 1 - 1$$
$$= -4$$

If two sequences are different, the inner product is 0.

$$B \cdot (-C) = [+1,\ -1,\ +1,\ -1] \cdot [-1,\ -1,\ +1,\ +1]$$
$$= -1 + 1 + 1 - 1$$
$$= 0$$

Orthogonal Codes in CDMA

In the Multiplexer each station is sending the appropriate sequence:

- Station 1 is sending -*A* (-1 was multiplied by *A*)
- Station 2 is sending -*B*
- Station 3 is sending an empty sequence (all zeros)
- Station 4 is sending *D*.
- The sequence that comes out of the multiplexer is the sum of all sequences.

$$S = -A - B + D$$

Orthogonal Codes in CDMA

In the Demultiplexer all stations receive *S:*

- Station 1 finds the inner product of S and *A*.

  $S.A = (-A-B+D).A = -A.A - B.A + D.A = -4+0+0 = -4$

  The result is then divided by 4, which is -1. This is interpreted as a 0 bit.
- Station 2 finds the inner product of *S* and *B*.

  $S.B = (-A-B+D).B = -A.B - B.B + D.B = 0-4+0 = -4$

  The result is then divided by 4, which is -1. This is interpreted as a 0 bit.
- Station 3 finds the inner product of *S* and *C*.

  $S.C = (-A-B+D)C = -A.C - B.C + D.C = 0+0+0 = 0$

  This is interpreted as a silent station.
- Station 4 finds the inner product of *S* and *D*.

  $S.D = (-A-B+D)D = -A.D - B.D + D.D = 0+0+4 = 4$

  The result is then divided by 4, which is +1. This is interpreted as an 1 bit.

Comparison between FHSS and DSSS/CDMA

Both the spread spectrum techniques are popular for their characteristics. To have a clear understanding, let us take a look at their comparisons.

FHSS	DSSS / CDMA
Multiple frequencies are used	Single frequency is used
Hard to find the user's frequency at any instant of time	User frequency, once allotted is always the same
Frequency reuse is allowed	Frequency reuse is not allowed
Sender need not wait	Sender has to wait if the spectrum is busy
Power strength of the signal is high	Power strength of the signal is low
Stronger and penetrates through the obstacles	It is weaker compared to FHSS
It is never affected by interference	It can be affected by interference
It is cheaper	It is expensive
This is the commonly used technique	This technique is not frequently used