Modulation Techniques

Calculate the number of levels, baud rate, and bandwidth for sending data 10 bits at a time at a bit rate of 10 Mbps using FSK with a carrier frequency of 10 MHz. Show the allocation of frequencies and bandwidth.

Given:

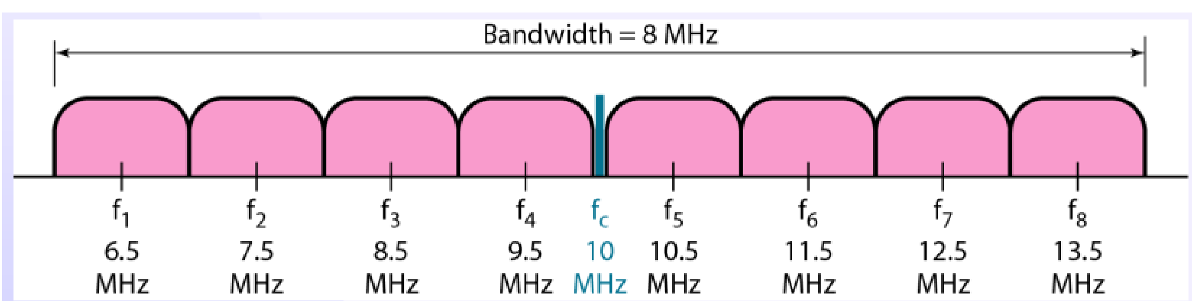
- Data = 10 bits at a time
- Bit rate = 10 Mbps
- Modulation = FSK
- Carrier frequency = 5×10^3 MHz = 5000 MHz = 5 GHz

1. Number of Levels:
Number of levels = 2^{10} = **1024 levels**

2. Baud Rate (Symbol Rate):
Baud rate = Bit rate ÷ Bits per symbol
Baud rate = 10 Mbps ÷ 10 bits = **1 Mbaud**

3. Bandwidth Calculation:
For M-ary FSK (M = number of levels): Bandwith = Signal Levels × Δf
Where: B = Bandwidth M = 1024 Δf= frequency separation between tones = Baud rate = 1 MHz

Bandwidth = 1024 × 1 MHz = **1024 MHz**



4. Frequency Allocation:
Carrier frequency (fc) = 5000 MHz
Lowest frequency = fc − (B / 2) = 10 − 512 = **-502 MHz**
Highest frequency = fc + (B / 2) = 10 + 512 = **522 MHz**

A negative frequency (**−502 MHz**) is **not possible in practice**. This indicates that **FSK with 1024 levels and 1 MHz spacing centered at 10 MHz is not feasible** — it would require frequencies below zero. If we must use 10 MHz carrier, the frequencies would span: **Frequency span: 0 MHz to 522 MHz** (clamping the lower bound to 0)

How many FM radio channels can be operated at a time in an area, given FM radio operates in the 88 MHz to 108 MHz frequency band?

Given:
FM radio operates in the frequency band of 88 MHz to 108 MHz.

Total available bandwidth:
108 MHz − 88 MHz = 20 MHz

Bandwidth required per FM channel:
1 channel = 200 kHz = 0.2 MHz

Number of FM channels that can be operated:
20 MHz ÷ 0.2 MHz = 100 channels

So, theoretically, the 88–108 MHz band can support 100 channels. However, in practice, not all 100 channels can be used simultaneously in a single area due to interference.

For a 200 KHz bandwidth spanning from 300 KHz to 500 KHz, what are the carrier frequency and bit rate if modulated using Binary ASK with d=1?

Given:

- Bandwidth = 200 kHz (from 300 kHz to 500 kHz)
- Modulation: Binary ASK
- Modulation index (d) = 1

Carrier frequency = (Lower frequency + Upper frequency) ÷ 2

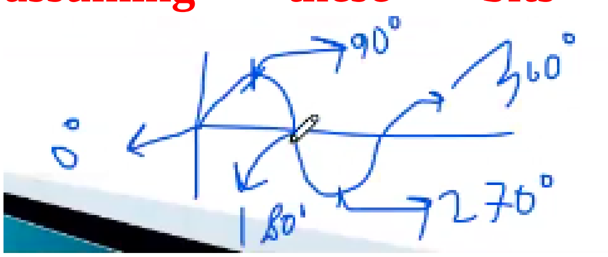
Carrier Frequency (fc): fc = (300 kHz + 500 kHz) / 2 = **400 kHz**

Bit Rate (Rb): Bandwidth = (1 + d) × S where (S= baud rate)
[d=modulation index | only applicable to fsk, value = 1]
200 kHz = 2 × S
S = 200 kHz / 2 = **100 baud** [S = N * r]

N = S/r = 100/1(ask = 1 bit) = 100 kbps

Using data "10111001," draw the waveform for BFSK and BPSK techniques, assuming these bits travel in 1 second.

Illustrate Binary BFSK and BPSK for the data 111110, assuming these bits travel in 1 second.



ASK:
1 hole modulated signal = 1 (line thk ek ghor upore)
0 hole modulated signal = 0 (line er sathe)
career frequency = 3 ta
1 hole modulated signal = 3 ta
0 hole modulated signal = 0 (kichu hbe na, line er sathe)
FSK:
1 hole modulated signal = 1 (line thk ek ghor upore)
0 hole modulated signal = 0 (line er sathe)
career frequency = 2 ta

1 hole modulated signal = 4 ta
0 hole modulated signal = 2 ta
PSK:
1 hole modulated signal = 1 (line thk ek ghor upore)
0 hole modulated signal = -1 (line thk ek ghor niche)
career frequency = 3 ta
1 hole modulated signal = 3 ta
0 hole modulated signal = prothome ekta 180° thk shuru hoye 360° e sesh hbe, erpore 2 ta full, erpore ekta 0° thk shuru hoye 180° e sesh hbe

What do you mean by signal rate and data rate? Write the relationship between signal rate and data rate. An analog signal has a bit rate of 16000 bps and a baud rate of 1000 baud. How many data elements are carried by each signal element? How many signal elements do we need?

Data Rate (Bit Rate): The number of bits transmitted per second (bps).

Signal Rate (Baud Rate): The number of signal elements (or symbols) transmitted per second.

The relationship between data rate (bit rate) and the signal rate (baud rate) is:

$$S = N * (1/r) \Rightarrow N = S * r$$

Data Rate/Bit Rate (bps) = Signal Rate (baud) × Bits per Signal Element

Bit Rate = 16000 bps, Baud Rate = 1000 baud

1. Bits per Signal Element: = Bit Rate / Baud Rate = 16000 / 1000 = 16 bits

2. Number of Signal Levels L = 2^r (since r=log2L so, L=2^r) = 2^(Bits per Symbol) = 2^16 = 65,536

(Normally baud rate = number of signal elements. jehetu baud rate deya ache question e ei, so ekhane bujhe nite hbe number of signal levels bolche, Jodi number of signal elements er kotha bole thake)

If the baud rate is 1000 baud per second and each signal element carries 4 bits, what is the bit rate? How many signal elements are required?

Given:
Baud rate = 1000 baud
Bits per signal element = 4
Bit rate = Baud rate × Bits per signal element
= 1000 × 4 = 4000 bps
Signal elements required = Bit rate ÷ Bits per signal element
= 4000 ÷ 4 = 1000 signal elements

(OR, jehetu baud rate 1000 baud & baud rate = num. of signal elements, so signal elements needed = 1000)

Calculate the baud rate for the given bit rate and type of modulation:

- **2000 bps, FSK**
- **4000 bps, ASK**
- **6000 bps, QPSK**
- **36,000 bps, 64-QAM**

BPSK, FSK, ASK: 1 bit per symbol.
QPSK: 2 bits per symbol.
16-QAM: 4 bits per symbol.
64-QAM: 6 bits per symbol.

Baud Rate = Bit Rate ÷ Bits per Signal Element

1. 2000 bps, FSK (1 bit per symbol):
Baud Rate = 2000 ÷ 1 = **2000 baud**

2. 4000 bps, ASK (1 bit per symbol):
Baud Rate = 4000 ÷ 1 = **4000 baud**

3. 6000 bps, QPSK (2 bits per symbol):
Baud Rate = 6000 ÷ 2 = **3000 baud**

4. 36,000 bps, 64-QAM (6 bits per symbol, since 64 = 2^6):
Baud Rate = 36000 ÷ 6 = **6000 baud**

Write differences between Amplitude Modulation and Frequency Modulation.

AM (Amplitude Modulation)	FM (Frequency Modulation)
1. Changes the (amplitude) of the wave	1. Changes the (frequency) of the wave
2. Gets easily disturbed by noise	2. Stays clearer even with some noise
3. Sound isn't very clear	3. Sound is clearer and better
4. Needs less bandwidth	4. Needs more bandwidth
5. Equipment is simple and cheaper	5. Equipment is more advanced and pricey

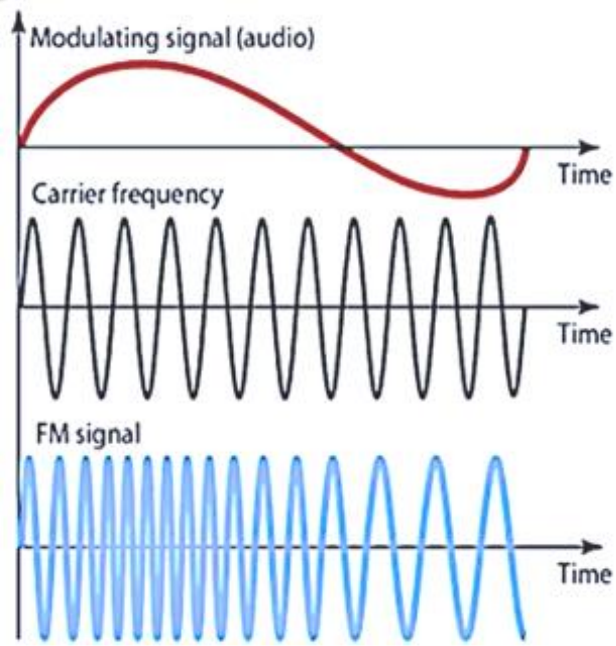
Explain Frequency Modulation with a figure. Why are guard bands required between FM radio stations?

Frequency Modulation (FM) is a way to send information by changing the frequency of a carrier wave. In FM, the amplitude of the wave stays the same, but the frequency moves up and down based on the data signal.

- If the signal is stronger (high amplitude), the frequency of the carrier wave increases.

- ➢ If the signal is weaker (low amplitude), the frequency of the carrier wave decreases.

☞ The first picture shows the message signal, which is the information we want to send. The second picture is the carrier wave, that will carry the message. Its frequency changes based on how strong (amplitude) the message signal is.



In the third picture, you see the frequency modulated signal. When the message signal is small, the carrier wave's frequency stays close to normal. When the message signal gets bigger, the carrier's frequency goes up. When the message signal gets smaller again, the carrier frequency goes down.

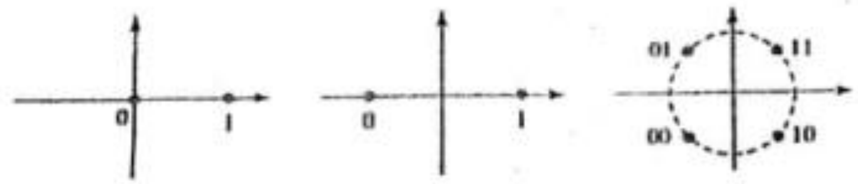
So, in frequency modulation, the carrier wave's frequency changes directly with the message signal's strength. This makes FM better at avoiding noise and interference than Amplitude Modulation (AM), where noise can mess up the signal because it changes the wave's strength.

Frequency Modulation is used in FM radio broadcasting, radar, two-way radio systems, and video-transmission systems.

☞ Guard bands are small frequency gaps kept between two adjacent FM stations. Guard Bands Required Between FM Radio Stations to

➢ Stop signals from mixing with each other
➢ Cut down noise and interference
➢ Help every station sound clear and clean

Explain the number of signal elements and data elements in each of the following figures:



◈ Figure 1 (Leftmost):

- Two distinct points on the horizontal axis: labeled `0` and `1`
- ◈ **Signal elements:** 2 (each point represents a different signal element)
- ◈ **Data elements:** 1 bit per signal (because 2 symbols = 2^1 = 1 bit)

✓ This represents a **Binary Amplitude Shift Keying (ASK)** or **BPSK**-like scheme.

◈ Figure 2 (Middle):

- Two points on the horizontal axis, one labeled `0` and the other `1`
- Similar to the first but possibly representing **different phases**
- ◈ **Signal elements:** 2
- ◈ **Data elements:** 1 bit per signal

✓ This could represent **Binary Phase Shift Keying (BPSK)** — using 2 phases to carry 1 bit.

◈ Figure 3 (Rightmost):

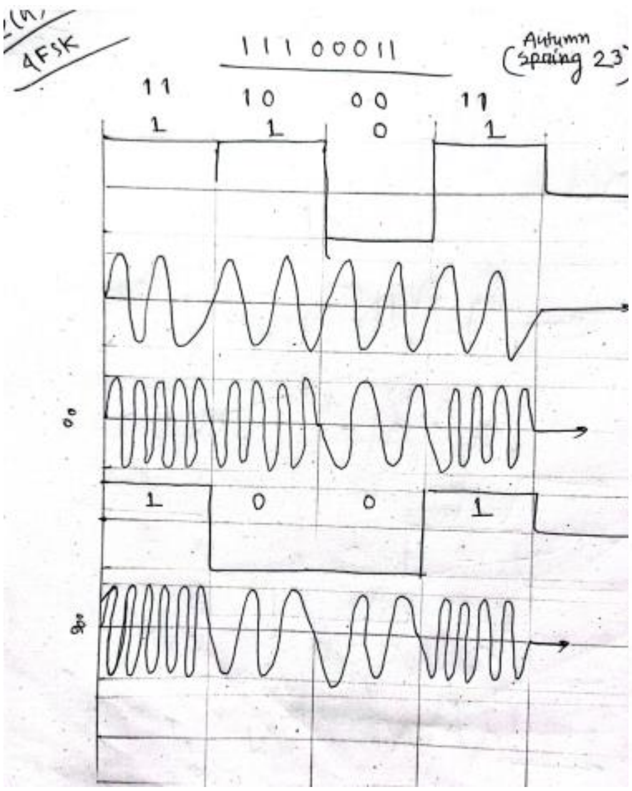
- Four points arranged in a circle labeled with bit pairs: 00, 01, 10, 11
- This is a **constellation diagram of 4-PSK (QPSK)**
- ◈ **Signal elements:** 4
- ◈ **Data elements:** 2 bits per signal (because 4 symbols = 2^2 = 2 bits)

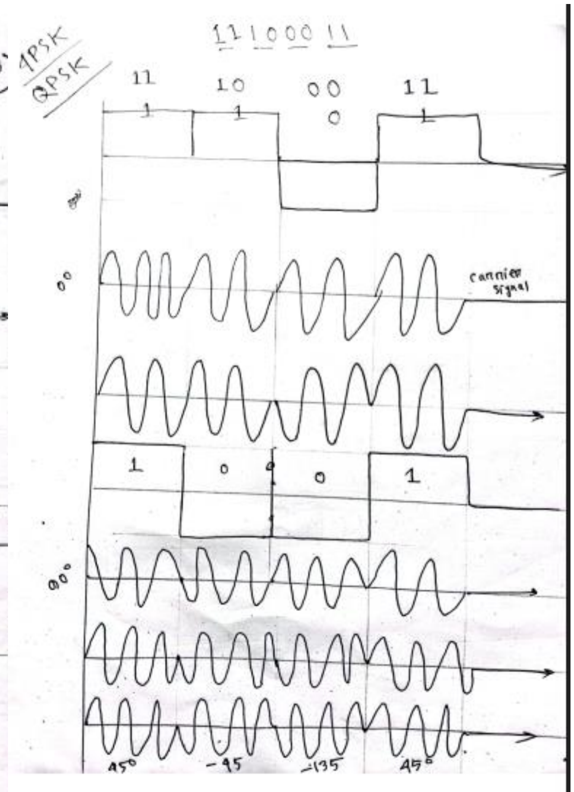
✅ Each phase point represents a unique 2-bit combination.

Wireless systems usually use analog transmission for -transmitting the digital data. In this case environment, distance, bitrate etc. are important issues to be considered. Using data "11100011" draw the waveform for 4FSK, 4PSK techniques. Write your comments on each of these modulation techniques. 4PSK=QPSK (SAME) == উপরেরটার মতো

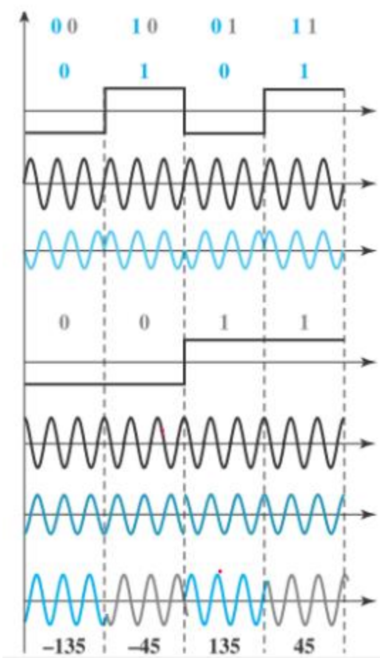
4FSK (4-Level Frequency Shift Keying) uses four separate frequencies to send data, where each one stands for a 2-bit combination. It needs more bandwidth than phase-based methods, but it's better at handling noise. That's why it works well in noisy wireless environments. Since each frequency is clearly different, it's easier for the receiver to detect and decode the signal, even if the signal isn't very strong.

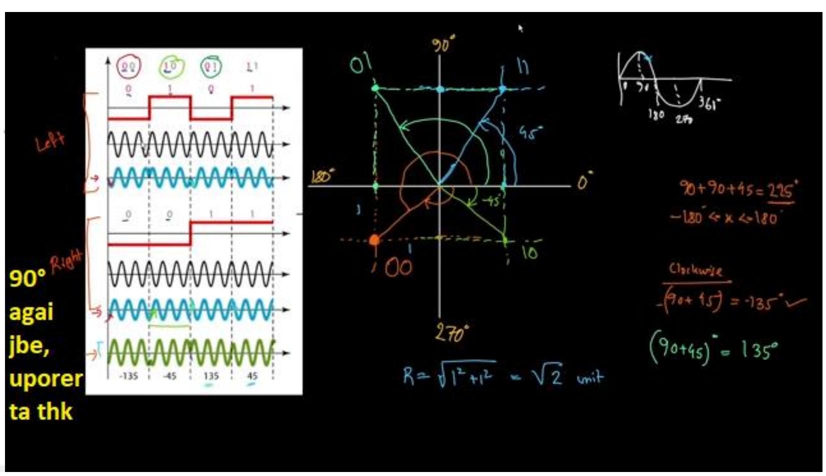
4PSK (4-Level Phase Shift Keying) uses the same frequency but changes the phase to send different data symbols. This saves bandwidth, which is helpful when the available spectrum is limited. However, it's more affected by phase noise and needs very accurate timing to work well. That makes the receiver more complex, especially in poor signal conditions.





Illustrate QPSK when the data bits are 11 10 01 00.

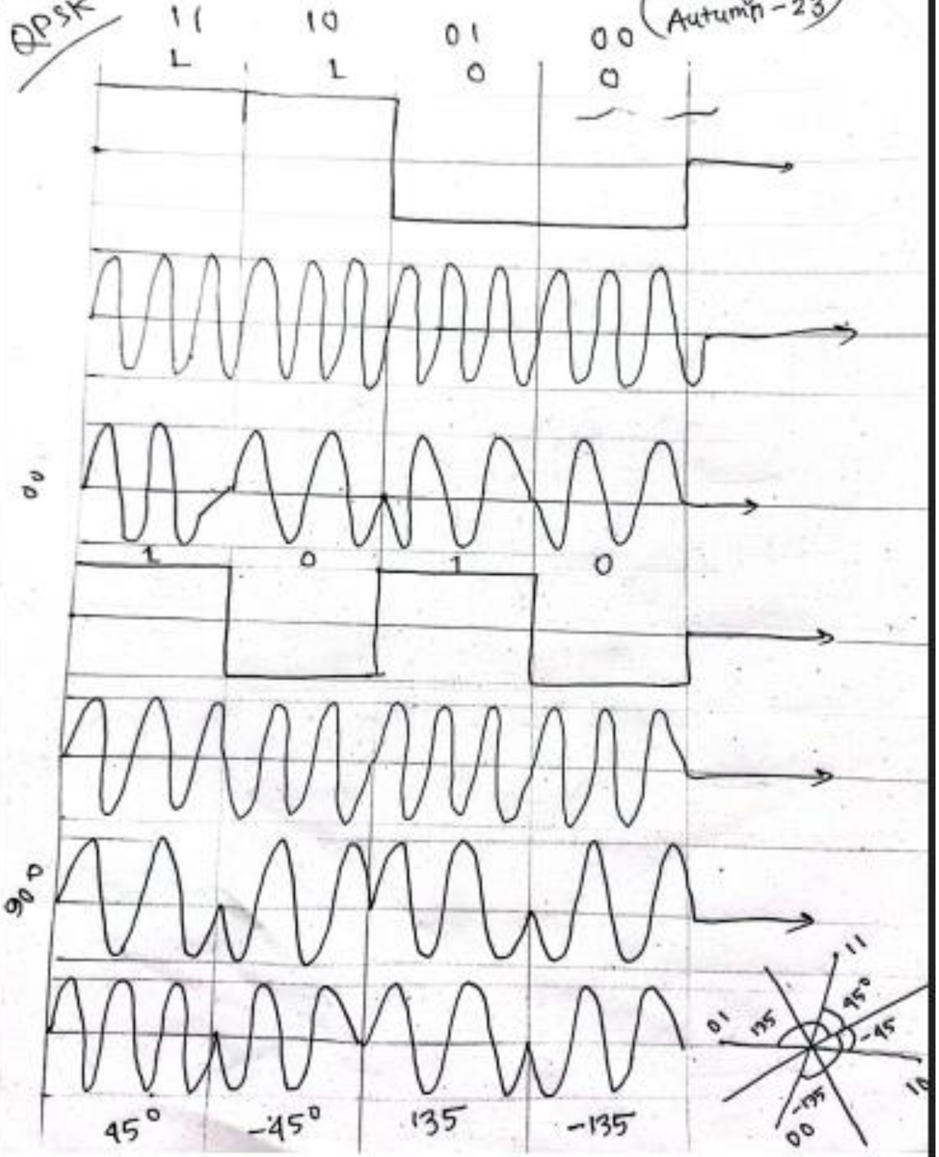




9) QPSK

11100100

(Spring-24
Autumn-23)



$$\text{Data rate} = \text{Frame rate} * \text{Frame size}$$

$$\text{Frame size} = \underbrace{N}_{\text{source}} * \underbrace{\text{bit rate}}_{\text{source}}$$

$$\text{Frame rate} = \frac{\text{Input bit rate}}{\text{time slot}}$$

$$\text{Frame duration} = \frac{1}{\text{Frame rate}}$$

$$\text{Input bit duration} = \frac{1}{\text{bit rate}}$$

$$\text{Output bit duration} = \frac{\text{Input bit duration}}{\text{Channel}}$$

$$\text{Output bit rate} = \frac{1}{\text{Output bit duration}}$$

$$1\,\mu s = \frac{1}{M}$$

$$1\,ms = \frac{1}{\text{Kbps}}$$

$$1\,\text{Mbps} = 10^3\,\text{Kbps}$$

$$1\,ms = 10^3\,\mu s$$

HIMEL

A corporation possesses a satellite channel with a bandwidth of 10 MHz and intends to establish 40 distinct and independent channels, each capable of transmitting a minimum of 10 Mbps. Design a suitable modulation technique for this communication requirement. (V.V.I)

To send 40 separate and independent channels, each needing at least 10 Mbps, over a 10 MHz satellite bandwidth, we can use a method called **Orthogonal Frequency Division Multiplexing (OFDM)** along with **256-QAM** modulation. OFDM works by breaking the available bandwidth into many smaller parts called subcarriers. These subcarriers are designed to be "orthogonal," meaning they don't interfere with each other. This allows us to send many data streams at once efficiently, which is perfect for fitting multiple channels into a limited frequency band like this.

We divide the 10 MHz total bandwidth equally among the 40 channels. So, each channel gets
10 MHz ÷ 40 = 0.25 MHz = 250 kHz.

To reach a 10 Mbps data rate in 250 kHz of bandwidth, each channel needs a **spectral efficiency** of
10 Mbps ÷ 250 kHz = 40 bps/Hz.

To achieve this, we use **256-QAM**, which carries **8 bits per symbol**. If we send symbols at a rate of **1.25 million symbols per second (1.25 Msps)**, then
1.25 Msps × 8 bits/symbol = 10 Mbps.

So, this setup satisfies the requirement.

In short, this design uses OFDM to divide the spectrum into 40 non-overlapping parts. Each channel uses **256-QAM** at **1.25 Msps** and takes up **250 kHz**, adding up to the full 10 MHz. This method fits all 40 channels neatly and makes the system strong against interference, which is important in satellite communication.

1(a) —

(Spring 24) Autumn 2022
Spring 2023
Autumn – 2023

A corporation process a satellite channel with a bandwidth of 10 MHz. The corporation intends to establish 40 distinct and independent channels, each capable of transmiting a minimum of 10 Mbps. Design a suitable modulation technique for this communication requirement.

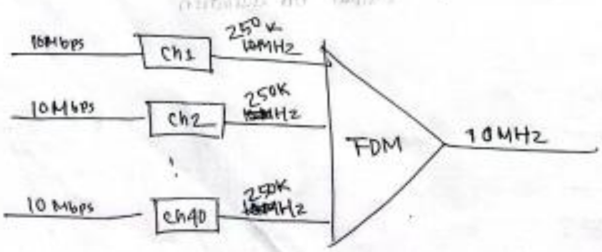
$\therefore$ Bandwidth $(B) = (1+d) \times S$ signal rate $(S) = N \times \frac{1}{r}$

$= (1+d) \times N \times \frac{1}{r}$

From given, Here,

A channel with 10MHz bandwidth to establish 40 distinct $N = 10$ Mbps

ind. channel then the $B = \frac{10 \text{ MHz}}{40} = \frac{10000}{40} = 250$ KHz



1(a) – Spring 2022

A corporation has a medium of satalite channel with a 1 MHz Frequency bandwidth. The corporation needs to create 4 separate independent channels each capable of sending at least 1 Mbps you are design an appropiate modulation technique. (let $d=0$)

The Bandwidth $B=(1+d)\times S$ Here,

$d=$ Depends on the modulation and filtering process $=0$

Signal rate $(S)=N\times\frac{1}{n}$

The data rate in bps $(N)=1$ Mbps

$n=$ the number of data elements carried in one signal element $=\log_2 L$ Here,

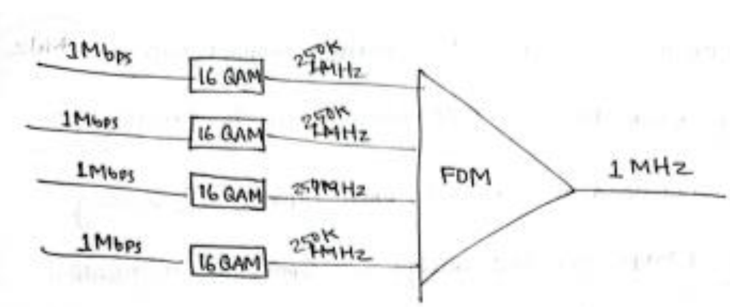
L is the type of signal element

From given, a midium with a 1 MHz bandwidth to create 4 seperate ind channel then the $B=\frac{1\,\text{MHz}}{4}=\frac{1000\,\text{kHz}}{4}=250$ kH

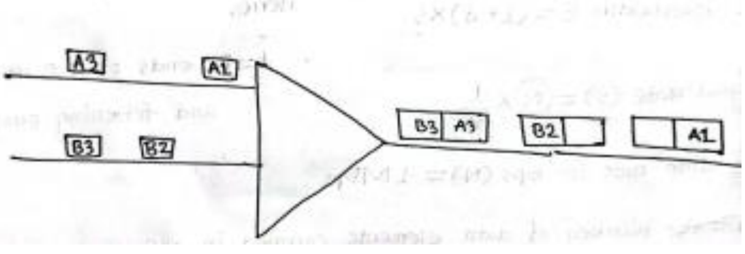
Now, S value substitude in formula then

$$B=(1+d)\times N\times\frac{1}{n}$$

$$n=(1+d)\times N\times\frac{1}{B}$$



Empty Slots :



Multiplexing, Spreading and Switching

1. Time Division Multiplexing (TDM):

Four sources, each with a bit rate of 800 kbps, need to be combined using TDM with a byte interval and synchronizing bits.

1. **What is the size of a frame in bits?**
2. **What is the frame rate?**
3. **What is the duration of a frame?**
4. **What is the data rate?**

Given:

- Number of sources = 4
- Bit rate per source = 800 kbps
- TDM (Time Division Multiplexing) with a byte interval (8 bits per source per frame)
- Synchronizing bits = 1 bit per frame (assumed, as it's common in TDM)

1. What is the size of a frame in bits?

- Each source contributes 1 byte (8 bits) per frame.
- Total data bits = 4 sources × 8 bits = 32 bits.
- Adding 1 synchronizing bit.
- Frame size = 32 bits + 1 bit = **33 bits**.

2. What is the frame rate?

- Frame rate is determined by the bit rate of a single source.
- Bit rate per source = 800 kbps = 800,000 bits per second.
- Since each frame contains 8 bits from each source, the frame rate = bit rate / bits per source per frame = 800,000 bits/s ÷ 8 bits = **100,000 frames per second**.

3. What is the duration of a frame?

- Duration of a frame = 1 / frame rate.
- Duration = 1 / 100,000 s = **10 μs**.

4. What is the data rate?

- Data rate = frame rate × frame size = 100,000 frames/s × 33 bits/frame = 3,300,000 bps = **3.3 Mbps**

Four 1 kbps connections are multiplexed together, with a unit of 1 bit. Find:

1. **The duration of 1 bit before multiplexing**
2. **The transmission rate of the link**
3. **The duration of a frame**
4. **The duration of a time slot**
5. **Frame Rate**

Given:

- Number of connections = 4
- Bit rate per connection = 1 kbps = 1000 bits per second
- Multiplexing unit = 1 bit per connection

1. The duration of 1 bit before multiplexing

- Duration of 1 bit = 1 / bit rate per connection
- Duration = 1 / 1000 s = **1 ms**

2. The transmission rate of the link

- Transmission rate = total bit rate of all connections combined
- Rate = 4 connections × 1 kbps = **4 kbps**

3. The duration of a frame

- Frame size = 1 bit per connection × 4 connections = 4 bits
- Frame duration = 1 / frame rate
- Frame rate = total bit rate / bits per frame = 4000 bits/s ÷ 4 bits = 1000 frames per second
- Duration = 1 / 1000 s = **1 ms**

4. The duration of a time slot

- Time slot duration = frame duration / number of connections
- Duration = 1 ms / 4 = **0.25 ms**

4. Frame Rate

- Frame size = 1 bit per connection × 4 connections = 4 bits
- Frame rate = total bit rate / bits per frame = 4000 bits/s ÷ 4 bits = 1000 frames per second

You need to use synchronous TDM and combine 20 digital sources, each of 100 Kbps. Each output slot carries 1 bit from each digital source, but one extra bit is added to each frame for synchronization.

1. **What is the size of an output frame in bits?**
2. **What is the output frame rate?**
3. **What is the duration of an output frame?**
4. **What is the output data rate?**

Given:

- Number of digital sources = 20
- Bit rate per source = 100 kbps = 100,000 bits per second
- Each output slot carries 1 bit from each source
- 1 extra bit added per frame for synchronization

1. What is the size of an output frame in bits?

- Each frame contains 1 bit from each of the 20 sources.
- Total data bits = 20 bits.

- Adding 1 synchronization bit.
- Frame size = 20 bits + 1 bit = **21 bits**.

2. Output frame rate:

Since each source contributes 1 bit per frame at 100 Kbps:

- Bits per second from one source = 100,000 bps
- Bits per frame from one source = 1 bit
- Frame rate = 100,000 ÷ 1 = **100,000 frames per second**

3. Duration of an output frame:

Duration of one frame = 1 ÷ Frame rate

- Duration = 1 ÷ 100,000 = **10 microseconds (10 µs)**

4. Output data rate:

Total output data rate = Frame rate × Frame size

- Data rate = 100,000 frames/s × 21 bits/frame
- **Data rate = 2,100,000 bps = 2.1 Mbps**

Distinguish between Synchronous TDM and Statistical TDM

Synchronous TDM	Statistical TDM
1. Each source gets a fixed time slot	1. Slots given only when data is ready
2. Wastes bandwidth if some sources are quiet	2. Uses bandwidth better, only when needed
3. Every frame has space for all sources	3. Frames include only active sources
4. Easy to set up and understand	4. More tricky because slots change dynamically
5. Delay is steady and easy to predict	5. Delay can change depending on traffic

2. Frequency Division Multiplexing (FDM):

Assume seven channels, each with a 100-kHz bandwidth, are to be multiplexed together using FDM. What is the minimum bandwidth of the link if there is a need for a guard band of 20 kHz between the channels? Why is a guard band needed here? Explain with an appropriate figure.

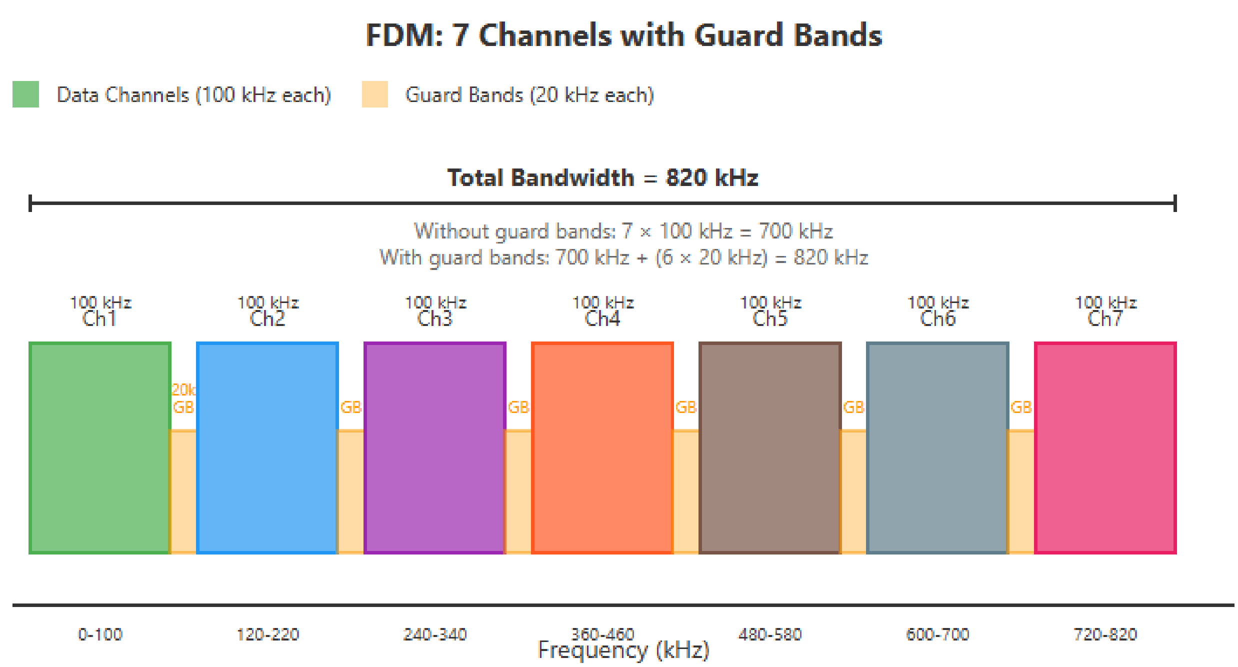
Given information:

- 7 channels, each with 100 kHz bandwidth
- Guard band of 20 kHz between adjacent channels

The multiplexed signal contains:

- 7 channels: 7 × 100 kHz = 700 kHz
- Guard bands: 6 guard bands × 20 kHz = 120 kHz (Note: Between 7 channels, there are 6 gaps requiring guard bands)

Total minimum bandwidth = 700 kHz + 120 kHz = 820 kHz



A guard band is a small gap between channels that helps:

- Stop the channels from overlapping
- Cut down interference between signals
- Make sure the receiver can clearly separate each channel

This extra space keeps the data communication clean and clear.

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 40 kHz between the channels to prevent interference?

Given information:

- 5 channels, each with 100 kHz bandwidth
- Guard band of 40 kHz between adjacent channels

The minimum bandwidth: The multiplexed signal will contain:

- 5 channels: 5 × 100 kHz = 500 kHz
- Guard bands: 4 guard bands × 40 kHz = 160 kHz (Note: We need 4 guard bands between 5 channels, not 5)

Total minimum bandwidth = 500 kHz + 160 kHz = 660 kHz

If there are 5 baseband analog signals, how many carriers are required when you are doing Frequency Division Multiplexing?

In Frequency Division Multiplexing (FDM), each baseband analog signal is modulated onto a separate carrier frequency to avoid interference. Therefore, the number of carriers required is equal to the number of baseband analog signals.

For 5 baseband analog signals, 5 carriers are required.

Write differences between Time Division Multiplexing and Frequency Division Multiplexing.

TDM (Time Division Multiplexing)	FDM (Frequency Division Multiplexing)
1. Shares time between users	1. Shares frequency between users
2. Users send data one after another	2. Users send data at the same time
3. Only one user sends data at a time	3. All users send at once on different frequencies
4. Needs accurate timing	4. Needs space (guard bands) between signals
5. Works well for digital signals	5. Works well for analog signals
6. Uses less bandwidth	6. Needs more bandwidth
7. Simple and cheaper devices	7. Devices are more complex and costly

3. Switching:

What is virtual circuit switching? Describe the total circuit setup, data transfer, and teardown phases with illustrations. Discuss the delay of a virtual circuit network with illustrations.

Virtual Circuit Network: A virtual circuit network is a cross between a circuit-switched network & a ~~diagram~~ datagram network. It has some characteristics too,

→ Addressing
→ Three phases
→ Efficiency
→ Delay
→ Circuit switched Technology in WANs

- In this network, there are setup & teardown phases in adition to the data transfer phase.
- Resources can be allocated during the setup phase.
- All packets follow the same path.
- It is normally implemented in the data link layer.

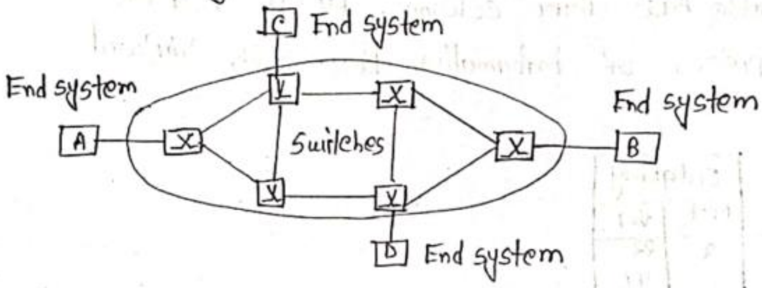


Fig: Virtual circuit Network

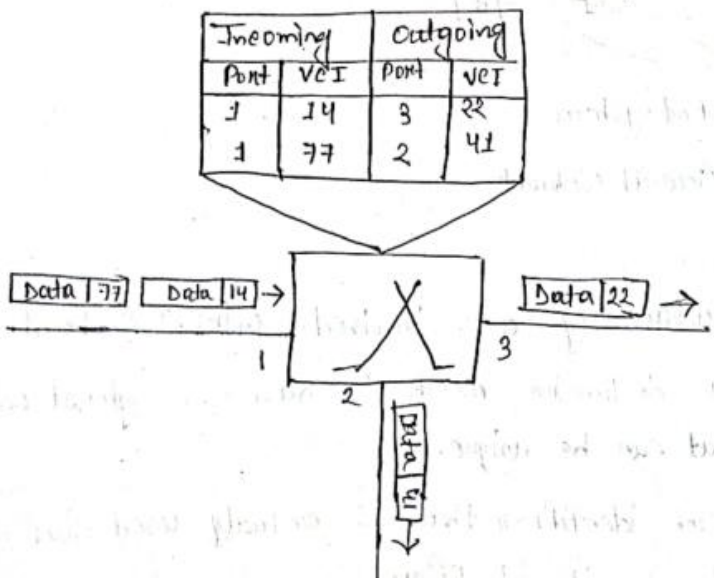
Three Phases: In this network, a source & a destination to go through three phases.

1. Setup
2. Data transfer
3. Teardown

1. Setup: A switches creates an entry for a virtual circuit.

   Example- Suppose source A needs to create a virtual circuit to B. Two steps are required: The setup request & the acknowledgement.

② **Data transfer:** To transfer a frame from a source to its destination, all wishes need to have a table entry for this Virtual circuit. ~~In its~~ This table has four columns. Means that the Suites holds four pieces of information for each Virtual circuit.



③ **Teardown phase:** In this phase, Source A after sending all frames to B. Sends a special frame called a teardown request.

④ **Efficiency:** 1st case, the delay for each packet is the same.
2nd case, Each case may encounter different delays.

⑤ **Delays in Virtual circuit Network:** There is a one time delay for setup & a one time delay for teardown. If resources are allocated during the setup Phase, there is no wait time for individual Packets.

Total delay = $3T + 3\tau$ + setup delay + Teardown delay

where,

$3T$ – Three transmission time

3τ – Three Propagation time

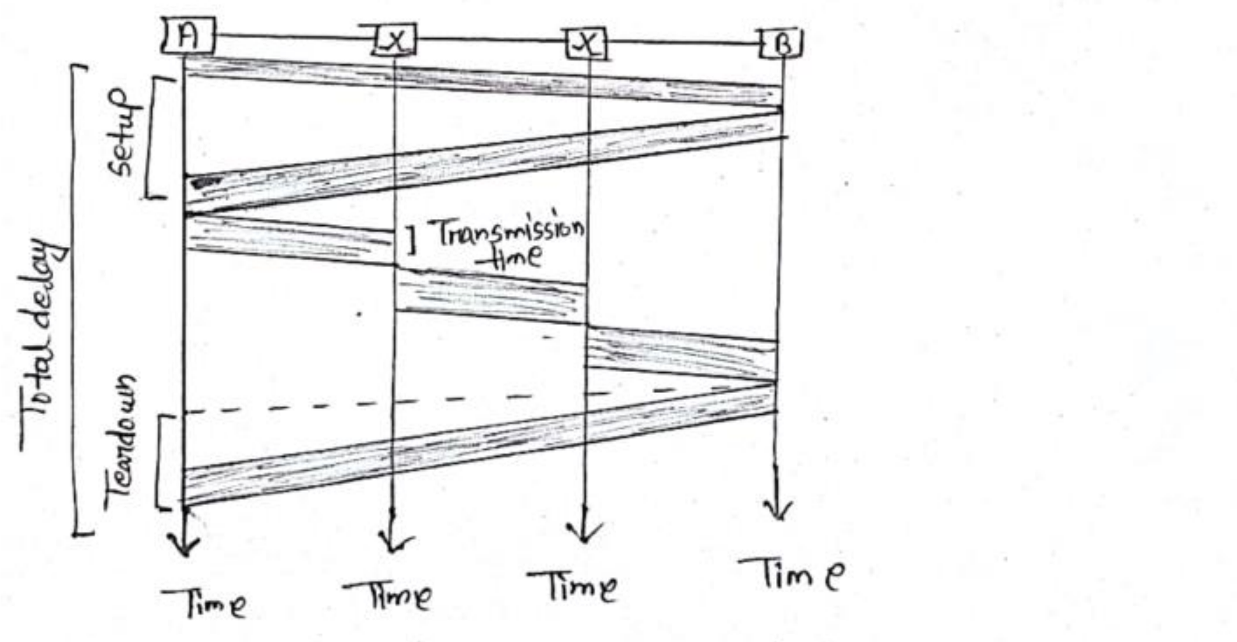


Fig: Delay In a Virtual ~~net~~ circuit network.

Compare the delay in Circuit Switched and Packet Switched Networks with appropriate examples.

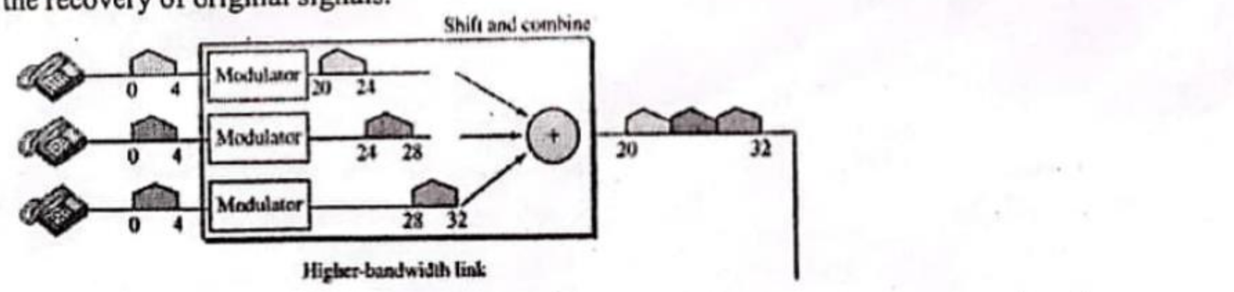
1. Circuit Switched Network: In a circuit switched network, before we start sending data, the network creates a dedicated path just for us. This means a fixed route is reserved between the sender and the receiver for the entire time we communicate. Because of this, there's a short wait at the start while the path is set up. After that, data flows smoothly and quickly since no one else uses that path.

Example: A good example is a traditional landline telephone call. When we call someone, the phone system sets up a special line only for our call. We might wait a few seconds for it to connect, but once it does, the conversation happens without interruptions and with steady delay.

2. Packet Switched Network: In a packet switched network, data is split into small pieces called packets. These packets travel separately and can take different paths to reach the receiver. There's no need to set up a fixed path first, so we start sending data right away. But because many users share the network, packets can get delayed, arrive out of order, or sometimes even get lost and need to be sent again.

Example: The internet is a typical packet switched network. When we send an email or visit a website, our data breaks into packets. Each packet might travel a different route depending on how busy the network is. This can cause some delays or lag, especially when many people are online.

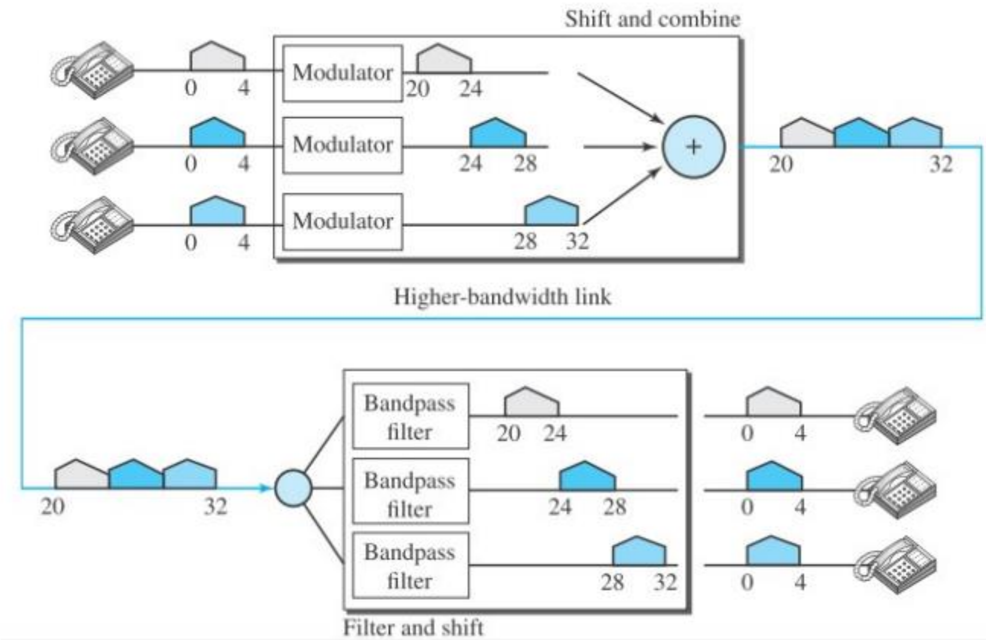
What type of multiplexing is given in the figure? Draw the diagram at the receiver side showing the recovery of original signals.



The type of multiplexing shown in the figure is Frequency Division Multiplexing (FDM).

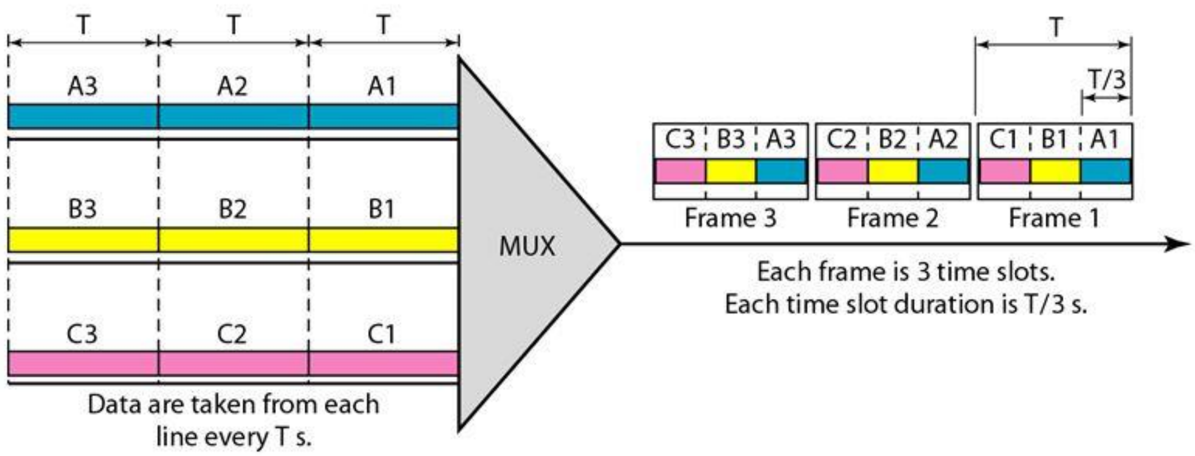
How it works:

1. Each input signal is first modulated using a different carrier frequency (like 20–24, 24–28, and 28–32 MHz).
2. These modulated signals are then mixed together and sent over a single high-bandwidth channel.
3. At the receiver end, the combined signal is separated using band-pass filters to pick out each frequency range. Then, each signal is demodulated to get back the original data.



Explain synchronous TDM with a figure. What are its advantages?

In Synchronous Time Division Multiplexing (STDM), every device is given the same time slot to send data, whether it has data to send or not. When a device's turn comes, it can send its data. But if it doesn't have any data at that moment, its time slot stays empty. These time slots are grouped into frames, and each frame has fixed time slots for each device. This method keeps the timing organized, even if some slots are unused.



Advantages of Synchronous TDM:

1. **Simple Setup:** Each device gets a fixed time to send data, which makes the system easy to design and use.
2. **Less Work:** Since time slots are already set, there's no need for extra processing or decisions.
3. **No Overlap:** Devices send data in their own time slots, so there's no chance of data crashing into each other.
4. **Stays in Sync:** The regular pattern helps both sender and receiver stay in step.
5. **Smooth Flow:** Every channel gets a steady turn to send data, which is great for voice and video.

How does Statistical TDM differ from Traditional Time Division Multiplexing (TDM)?

Statistical TDM is different from Traditional TDM because it gives time slots only when a device has data to send. In Traditional TDM, each device always gets a fixed time slot, even if it has nothing to send, which wastes bandwidth. But Statistical TDM uses time slots more wisely by sharing them based on who actually needs to send data, so there's less idle time and better use of the channel.

What is Spread Spectrum? Explain the differences between Frequency Hopping Spread Spectrum (FHSS) and Direct Sequence Spread Spectrum (DSSS).

Spread Spectrum is a way of sending signals where the data is spread out over a much wider range of frequencies than usual. This helps the signal avoid interference, noise, and makes it harder for others to listen in. Because of this, it makes wireless communication more reliable and secure.

FHSS (Frequency Hopping Spread Spectrum)	DSSS (Direct Sequence Spread Spectrum)

1. Quickly changes the carrier frequency over time	1. Mixes the signal with a special code to spread it out
2. Uses small chunks of frequency one after another	2. Uses a wide range of frequencies all at once
3. Works well against narrow interference on certain frequencies	3. Handles both narrow and wide interference better
4. Needs both sides to follow the same hopping pattern	4. Needs both sides to match the special spreading code
5. Offers some security by changing frequencies often	5. More secure because of the unique code it uses
6. Bluetooth and some military radios	6. GPS, Wi-Fi (like 802.11b), and some military systems

Describe multiplexing with a necessary example and figure.

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

→ In multiplexed system, n lines share the bandwidth of one link.

→ One link can have many channel.

→ This figure shows the basic format of a Multiplexed system.

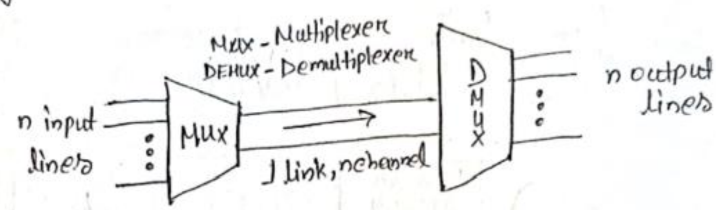


Fig: Dividing a link into channels

→ The word link refers to the Physical path.

→ channel refers to the ~~posito~~ Portion of a link that carries a transmission between a given pair of lines.

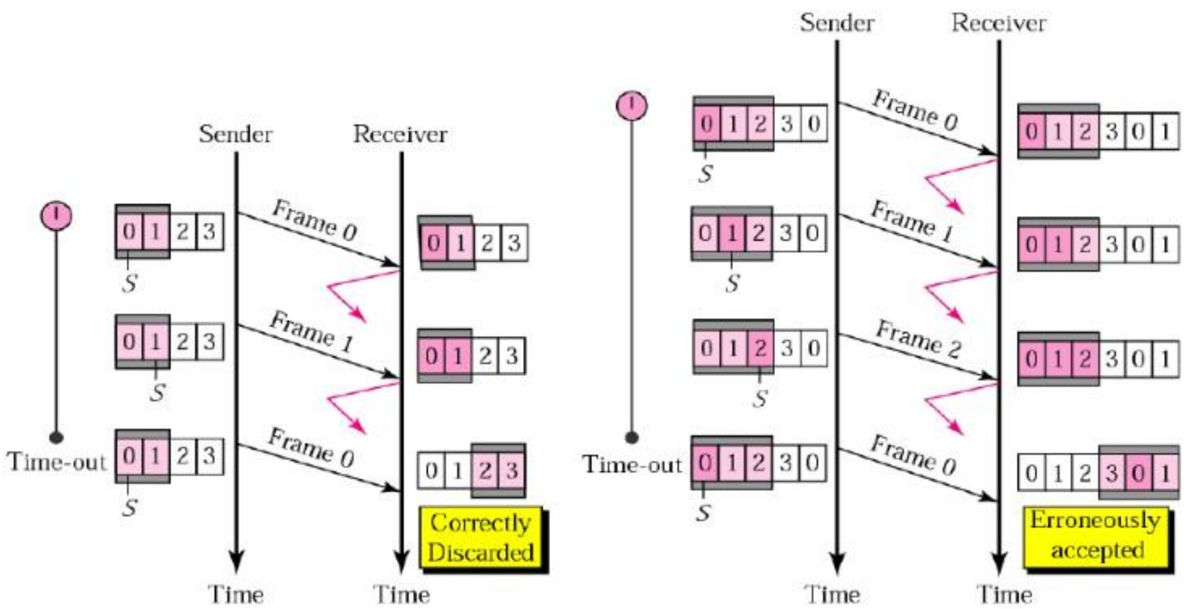
→ One link can have many (n) channels.

Example: Suppose three people want to send data over the same channel. Instead of giving each a separate line, we can use Time

Division Multiplexing (TDM). Each sender is assigned a specific time slot in which to send data. The receiver then separates the signals based on these time slots.

ARQ Protocols:

Explain, with appropriate diagrams, the consequences of not using an adequate window size for the Selective Repeat ARQ flow control protocol.



In the Selective Repeat ARQ protocol, the sender and receiver use something called a "window" to control how many frames can be sent and received without waiting for an acknowledgment (ACK). A very important rule here is that the window size must be at most half of 2 to the power of *m* (where *m* is the number of bits used for the sequence numbers). This helps keep the sequence numbers from overlapping and makes sure frames are correctly identified.

What Happens if the Window Size Is Too Big?

Let's say *m* = 2. That means the sequence numbers go from 0 to 3 (because $2^2 = 4$). According to the rule, the window size should be 2. But imagine if we set the window size to 3 by mistake.

- The sender sends frames numbered 0, 1, and 2.

- All the ACKs get lost on the way back.
- The sender times out and resends frame 0, thinking it was never received.
- The receiver, however, already got frame 0 earlier and has moved its window forward.
- Because sequence numbers repeat after 3, the receiver is now expecting a new frame 0.
- So the receiver accepts the resent frame 0 again, thinking it's a new frame.

Why Is This a Problem?

This means the receiver accepts duplicate data — it thinks the same frame is new. This causes errors in the data, which is bad for communication.

Why Keeping Window Size ≤ Half of 2^m Fixes This

If the window size is limited to 2 (half of 4), then the sequence numbers don't repeat too quickly. So a resent frame with the same sequence number won't be confused with a new frame, and duplicates won't be accepted by mistake.

Compare and contrast the Stop-and-Wait ARQ with the Go Back N ARQ flow control protocol.

Stop-and-Wait ARQ:

1. The sender sends **one frame at a time** and waits for an **ACK (acknowledgment)** before sending the next one.
2. It uses just a **1-bit sequence number** because only one frame is in process.
3. If the frame or its ACK is lost, the sender **resends the same frame** after a short wait (timeout).
4. This method is **less efficient**, especially on slow or long-distance links, because the sender spends time waiting.

5. It needs **very little memory**, since only one frame has to be stored during the wait.

Go-Back-N ARQ:

1. The sender can send **many frames in a row** without waiting, up to a set limit called the **window size (N)**.
2. It needs a **larger sequence number range** to keep track of multiple frames (usually $\log_2(N+1)$ bits).
3. If a frame has an error or is lost, the sender **goes back and resends that frame and all the ones after it**.
4. It's **more efficient** because the sender keeps using the network while waiting for ACKs.
5. It needs **more memory**, since the sender has to store all the frames it sent but hasn't gotten ACKs for yet.

Discuss the concept of redundancy in error detection and correction.

Redundancy in error detection and correction means adding extra bits to the original data to help find and fix mistakes during transmission. These extra bits don't carry real data but are created using special rules. They are sent along with the main data. When the receiver gets the data, it checks these extra bits to see if anything went wrong on the way. If there's an error, the system can sometimes figure out where it happened and fix it using the extra bits. Common methods that use redundancy include parity check, checksum, Hamming code, and CRC (Cyclic Redundancy Check).

Design a parity check code where datawords are 3 bits and codewords are 4 bits. Use odd parity. How do you detect errors with this encoding? What are the limitations of this C(4, 3) scheme?

> Design a parity cheak code where dataword 3 bits and codeword 4 bits use parity ? i. How do you detect errors with this encoding ii) What are the limitation of this C(4,3) scheme ?

Autumn 2022 - 3(b)
Spring 2022 - 3(a)

i)

Dataword	Codeword
101	1011
110	1101
101	1011
110	1100

ODD parity

ii) To detect errors with this encoding the receiver calculate the parity bits (1110) again using the received 4 bit codeword. If the total number of "1's in the received codeword is even, it indicates there is an errors in the transmission, If the codeword is odd, there is no errors.

iii) limitation of this C(4,3) Scheme:

Single error ditection

No error correction

Inefficiency

Low Redundancy

Ineffective for burst errors

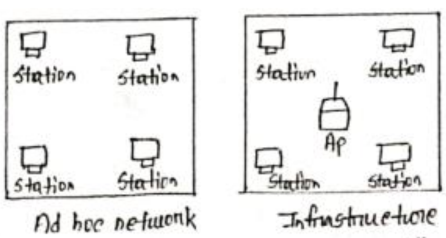
Multiple Access

Explain Basic Service Set and Extended Service Set in Wireless LAN with a figure.

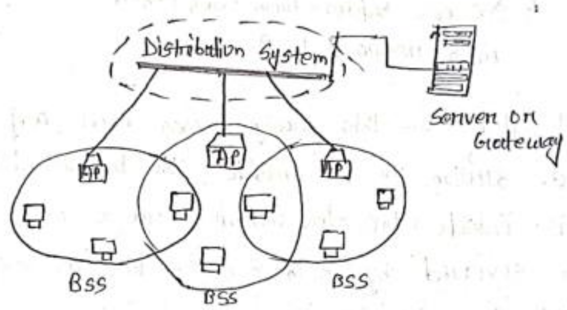
IEEE 802.11: IEEE has defined the specifications for a wireless LAN, called IEEE 802.11 which covers the physical & data link layers. The standard defines two kinds of services,

① The basic service set (BSS) &
② The extended service set (ESS)

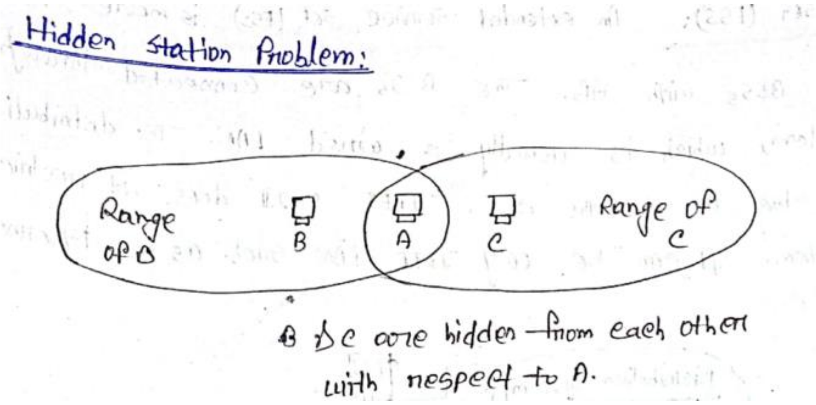
① The Basic Service Set (BSS): IEEE 802.11 defines the basic service set (BSS) as the building block of a wireless LAN. A basic service set is made of stationary or mobile wireless stations & an optional central base station known as access point (AP).



② Extended Service Set (ESS): An extended service set (ESS) is made up of two or more BSSs with APs. The BSSs are connected through a distribution system, which is usually a wired LAN. The distribution system connects the APs in the BSSs, IEEE 802.11 does not restrict the distribution system. It can be any IEEE LAN such as an Ethernet.



Describe the Hidden Station and Exposed Station problem with a necessary figure.



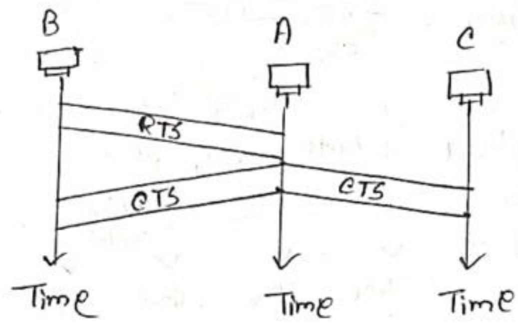
In this managed center, every station can hear signals from Station A. But Station B is out of range of Station C, and Station C is also out of B's range. However, Station A is within range of both B and C, so it can hear both of them.

Now, suppose Station B is sending data to Station A. In the middle of that, Station C also wants to send data to A. Since C can't detect B (because it's out of range), it thinks the channel is free and starts sending. This leads to a collision at Station A, as it receives data from both B and C at the same time.

This issue is called the Hidden Station Problem. Here, B and C are "hidden" from each other, even though they're both sending data to A. Their transmissions can clash, which lowers the network's efficiency.

The solution to the hidden station problem is the use of handshake frames, specifically RTS (Request to Send) and CTS (Clear to Send). These help make sure the channel is clear before any station starts sending data.

⇒ The CTS-frame in CSMA/CA handshake can prevent collision from a hidden station.

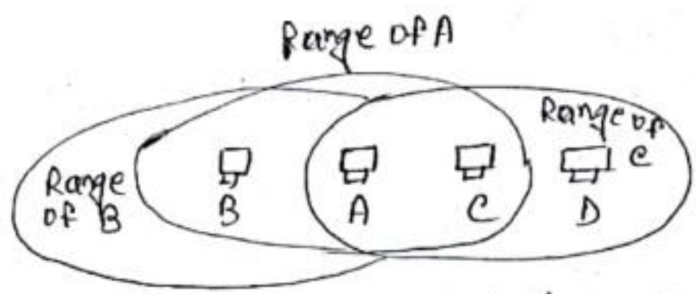


use of handshaking to prevent hidden station problem.

Exposed Station Problem:

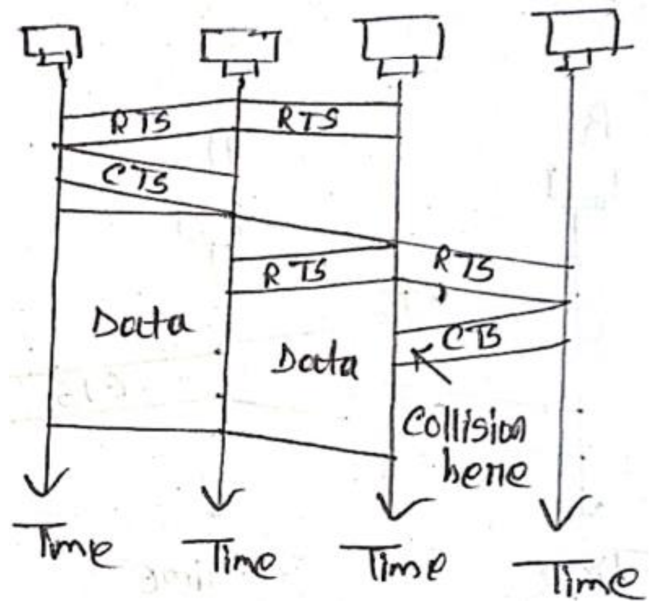
This is the opposite of the Hidden Station Problem. In this case, a station refrains from using the channel even though it could transmit without causing interference.

For example, suppose Station A is transmitting data to Station B. At the same time, Station C (which is within the range of Station A) wants to send data to Station D (which is outside the range of Station A and B). Since Station C can hear the transmission from Station A, it assumes the channel is busy and refrains from sending — even though its transmission to D would not interfere with A's transmission to B. As a result, Station C wastes channel capacity by being unnecessarily conservative.



C is exposed to transmission from A to B

The handshaking messages RTS and CTS cannot help in this case. Station C hears the RTS from A but doesn't hear the CTS from B. After waiting for a while, C sends an RTS to D. Both A and B hear it—A is busy sending data, so it ignores it, but B responds with a CTS.



The problem is here, if A has already started sending data, that signal can block C from hearing the CTS from D because of a collision. So C doesn't get the CTS and can't send its data. It stays exposed until A finishes sending.

How does the Ethernet address 1A:2B:3CAD:5E:6F appear on the line in binary? Compare a piconet and a scatternet.

The given address is 1A:2B:3C:AD:5E:6F. Each pair of hexadecimal digits represents one byte (8 bits), so the address consists of 6 bytes, totaling 48 bits.

Each hexadecimal digit corresponds to 4 bits in binary equivalent:

- 1A = 0001 1010
- 2B = 0010 1011
- 3C = 0011 1100
- AD = 1010 1101
- 5E = 0101 1110
- 6F = 0110 1111

So the complete Ethernet address 1A:2B:3C:AD:5E:6F appears on the line as:

000110100010101100111100101011010101111001101111

Piconet	Scatternet
1. A small Bluetooth network with one master.	1. A bigger network made by joining many piconets.
2. Has only one master device.	2. Has many masters (one in each piconet).
3. A device can be in just one piconet.	3. A device can be part of more than one piconet.
4. Data stays within one piconet.	4. Data can move between piconets.
5. Simple setup and easy to control.	5. More complex and harder to control.

Wireless WANs and Optical Networks

1. SONET:

What do you mean by SONET? Find the data rate of an STS-3 signal used in SONET.

SONET stands for **Synchronous Optical Network**. It defines a hierarchy of electrical signaling levels known as **Synchronous Transport Signals (STS)**.

Each STS level (from **STS-1** to **STS-192**) supports a specific data rate, measured in **megabits per second (Mbps)**. The corresponding optical signals are called optical carriers (OCs). SONET was developed by **ANSI (American National Standards Institute)**.

SONET transmission relies on three basic types of devices:

1. STS Multiplexers/Demultiplexers

2. Regenerators
3. Add/Drop Multiplexers and Terminals

Data Rate of STS-N = N × Data Rate of STS-1

Where: STS-1 rate = 51.84 Mbps & N = 3 (for STS-3)

Calculation: STS-3 Data Rate = 3 × 51.84 Mbps = 155.52 Mbps

Find the data rate of an STS-9 signal used in SONET.

Data Rate of STS-N = N × Data Rate of STS-1

Where: STS-1 rate = 51.84 Mbps & N = 9 (for STS-9)

Calculation: STS-3 Data Rate = 9 × 51.84 Mbps = 466.56 Mbps

2. Satellite Communication:

Which type of orbit does a GEO satellite have? Explain your answer. Find the period of the Moon, according to Kepler's law.

A GEO satellite has a **geostationary orbit**. It is called "Geostationary", because the satellite revolves around Earth at the same rate that Earth rotates, it appears stationary over a fixed point on the Earth's surface.

Explanation:

- A geostationary orbit is a circular orbit located approximately 22,000 miles above the Earth's equator.
- In this orbit, a satellite moves at the same speed the Earth spins. This makes the satellite look like it's staying still above one spot on Earth.
- Because it stays over the same place, the satellite can keep a steady line of sight with ground stations below.

- This steady connection is important for continuous communication since the sending and receiving antennas need to keep pointing at each other without losing track.
- There’s only one geostationary orbit because only at this height does the satellite’s speed match Earth’s rotation.
- But one GEO satellite can’t cover the whole Earth because of the planet’s curve, so at least three satellites spaced evenly (120° apart) are used to cover most of the globe.

In short, a GEO satellite stays fixed over one spot on the equator, which helps maintain constant communication with that area.

What is the period of the Moon, according to Kepler's law?

$$\text{Period} = C \times \text{distance}^{1.5}$$

Here C is a constant approximately equal to 1/100. The period is in seconds and the distance in kilometers.

Solution

The Moon is located approximately 384,000 km above the Earth. The radius of the Earth is 6378 km. Applying the formula, we get

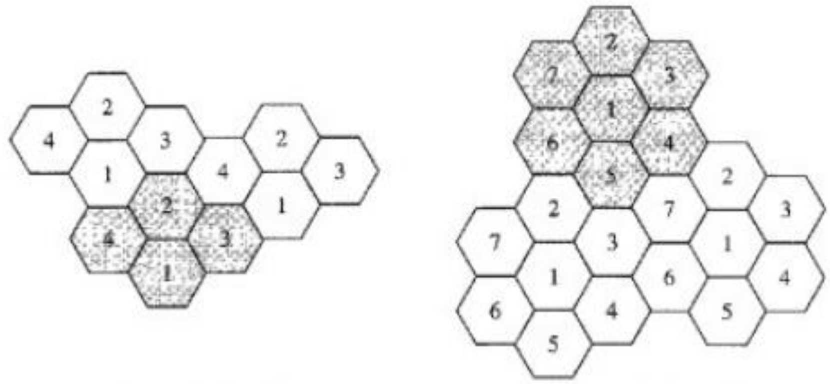
$$\text{Period} = \frac{1}{100}(384{,}000 + 6378)^{1.5} = 2{,}439{,}090 \text{ s} = 1 \text{ month}$$

Analyze Frequency Reuse in Cellular Networks with a figure.

Frequency reuse is a basic idea in cellular networks that helps us use the limited radio frequencies in a smart way. In these networks, the whole service area is split into small parts called cells. Each cell has its own base station that uses a certain set of frequencies to communicate with users.

Since there aren't many frequencies available, nearby cells cannot use the same frequencies at the same time because this would cause interference

To fix this, cellular networks use a frequency reuse pattern. A group of N cells, called a cluster, where each cell has a unique set of frequencies. Neighboring cells don't share frequencies, but cells far apart—called reuse cells—can use the same ones without interference.



a. Reuse factor of 4 b. Reuse factor of 7

A smaller N means more frequent reuse and higher capacity but more interference, while a larger N means less interference but less efficient use of frequencies.

Short Notes:

Satellite Communication: Satellites send and receive signals between stations on Earth across very long distances. They help in TV broadcasting, internet, and long-distance phone calls.

Mobile Communication: Mobile communication lets people talk and send data wirelessly using cell towers and mobile phones. It works for calls, texts, and internet while moving around.

Ethernet: Ethernet is a common way to connect computers with cables in a local area network (LAN). It gives fast and steady data transfer.

Wireless LAN: Wireless LAN (WLAN) lets devices connect to a network without wires using Wi-Fi. It is popular at homes, offices, and public places for internet access.

Bluetooth: Bluetooth is a short-range wireless method to link devices like phones, headphones, and speakers. It helps share data or stream sound without cables.

SONET: SONET (Synchronous Optical Network) is a fast fiber optic system that sends large amounts of data. It uses fixed-size frames to keep data synced.

STS: STS (Synchronous Transport Signal) is the type of signal used in SONET to carry digital data. Different levels like STS-1 or STS-3 mean different speeds.

PPP: PPP (Point-to-Point Protocol) helps create a direct connection between two devices. It's used for internet connections over phone lines or serial links.