Comparison between Data Encoding and Modulation:

	Data Encoding	Modulation
1.	Data encoding is the process of converting digital data into a specific format or code for transmission	It's a technique used to embed digital or Analog data into a carrier signal.
2.	It helps in error detection, correction, and efficient use of bandwidth.	It facilitates the transmission of information over a medium such as wire or air.
3.	Involves representing data using various coding schemes such as NRZ,RZ, HDB3, etc,	Involves various modulation techniques to vary the characteristics of carrier signal such as AM, ASK, QAM, etc
4.	Primarily operates in the digital domain, dealing with binary sequences and digital signals.	Operates in the Analog domain, as it involves varying the properties of a continuous carrier signal.
5.	Simple and less expensive	Complex and more expensive

Unit 8: Multiplexing and Switching

Multiplexing:

Whenever the bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices, the link can be shared. Multiplexing is the set of techniques that allows the simultaneous transmission of multiple signals across a single data link. As data and telecommunications use increases, so does traffic.

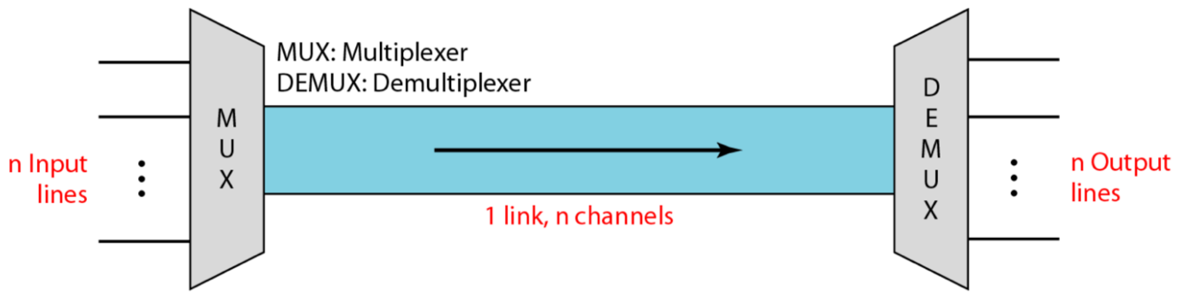
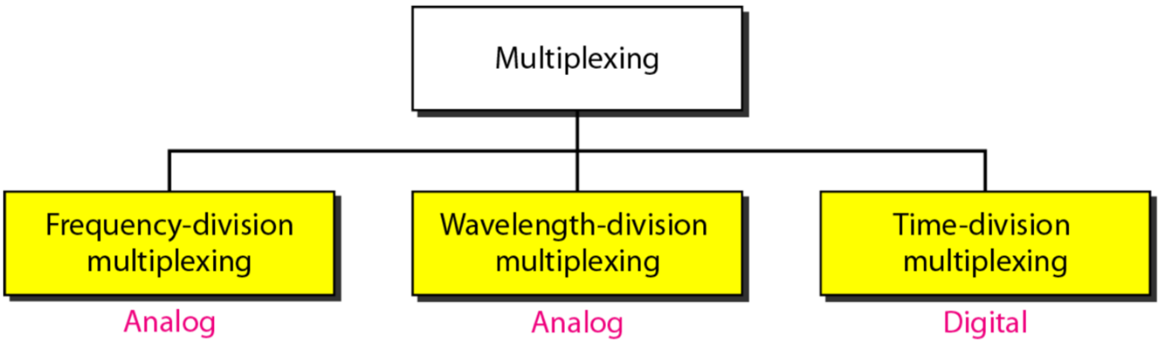


Fig.: Dividing a link in to channels

- In a multiplexed system, n lines share the bandwidth of one link.
- MUX – combines n-input lines into a single stream (many to one)
- DEMUX – separates single stream into n-output lines (one to many)

- Link – Physical path
- Channel – portion of a link that carries a transmission between a given pair of lines. One link can have many channels.

Types of Multiplexing



a) **Frequency Division Multiplexing (FDM):**

- Analog technique that can be applied when the bandwidth of a link (in Hertz) is greater than the combined bandwidths of the signals to be transmitted.
- In FDM, signal generated by each sending device modulate different carrier frequencies.
- These modulated signals are then combined into a single composite signal that can be transported by the link.
- Guard band are used to separate channels to prevent overlapping and interference.

Fig.: Frequency Division Multiplexing

- Application in telephone network, to maximize the efficiency of their infrastructure.
- Other applications; AM and FM radio broadcasting, TV broadcasting, etc

b) Wavelength Division Multiplexing (WDM):

- Designed to use the high data rate capability of fiber-optic cable
- WDM is an analog multiplexing technique to combine optical signals
- Conceptually, same as FDM, except that the multiplexing and demultiplexing involve optical signals transmitted through fiber optic channels.
- Very narrow bands of light from different sources are combined to make a wider band of light at the transmitter by MUX.
- At the receiver, the signals are separated by the DEMUX.

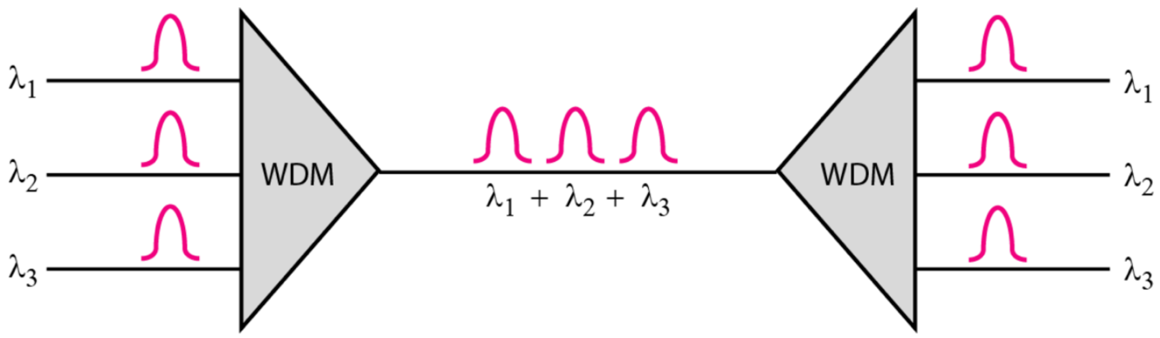


Fig.: Wavelength Division Multiplexing (WDM)

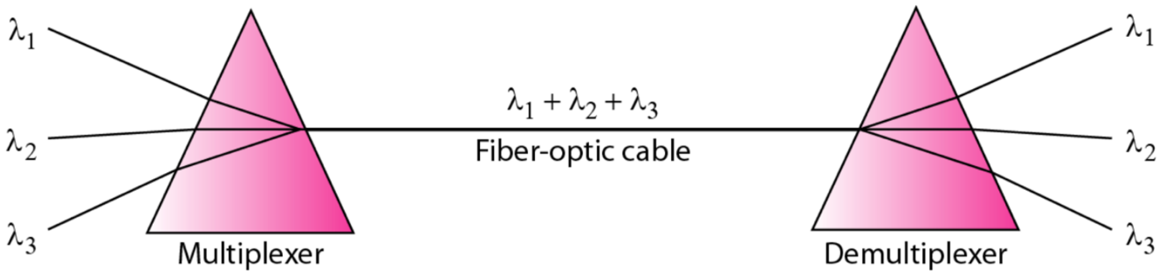


Fig.: Prisms in WDM MUX and DEMUX

- Application in the SONET (Synchronous Optical network), in which multiple optical lines are multiplexed and demultiplexed.
- Variation- Dense WDM (DWDM), can multiplex a very large number of channels by spacing channels very close to one another. Great efficiency than WDM.

c) Time Division Multiplexing (TDM):

- Digital data from different sources are combined into one timeshared link.
- Digital process that allows several connections to share the high bandwidth of a link.
- Each connection occupies a portion of time in the link.
- TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate one.

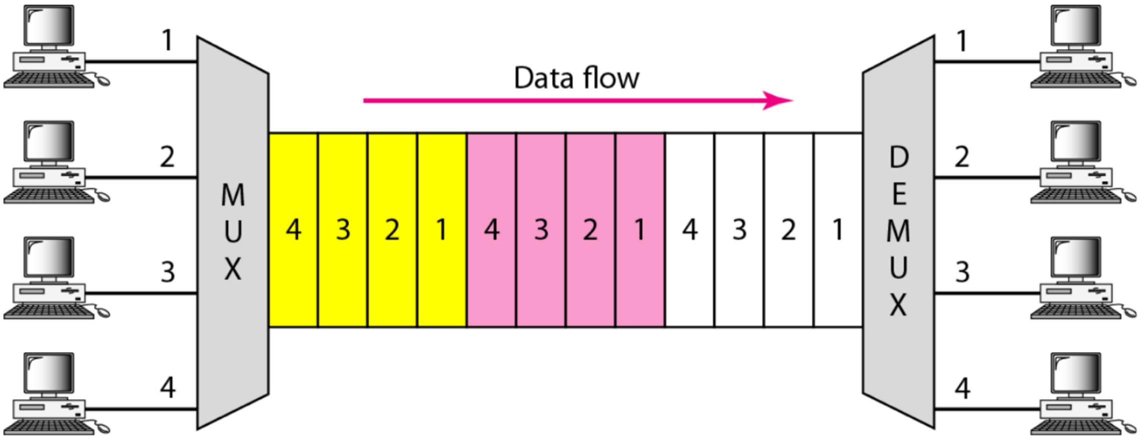


Fig.: TDM

- In fig., above we are concerned with only multiplexing, not switching. This means that all the data in a message from source 1 always go to one specific destination, be it 1,2,3, or 4. The delivery is fixed, unlike switching.

The Telephone System:

- To maximize the efficiency of their infrastructure, telephone companies have traditionally multiplexed signals from lower bandwidth lines onto higher bandwidth lines.
- In this way, many switched or leased lines can be combined into fewer but bigger channels.
- For analog lines, FDM is used.

Analog Hierarchy:

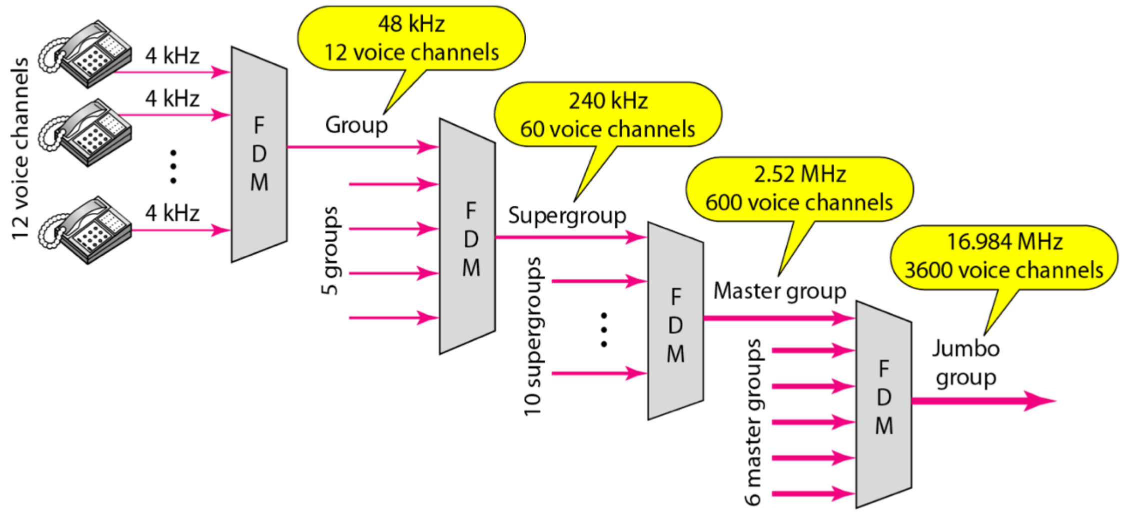


Fig.: Analog Hierarchy

In this Analog Hierarchy,

- **12 Voice channels** are multiplexed onto a higher bandwidth line to create **a group**. A group has 48 KHz of bandwidth and supports 12 voice channels.
- At the next level, upto **5 groups** can be multiplexed to create a composite signal called **a supergroup**. A supergroup has a bandwidth of 240 KHz and supports upto 60 voice channels.
- At the next level, **10 subgroups** are multiplexed to create **a master group.** A master group has a bandwidth of 2.52 MHz and supports upto 600 voice channels.
- Finally, **6 master groups** are multiplexed to create **a jumbo group**. A jumbo group has a bandwidth of 16.984 MHz and supports upto 3600 voice channels.

Digital Hierarchy:

- Telephone companies implement TDM through a hierarchy of digital signals, called **digital signal (DS) service** or **digital hierarchy**.

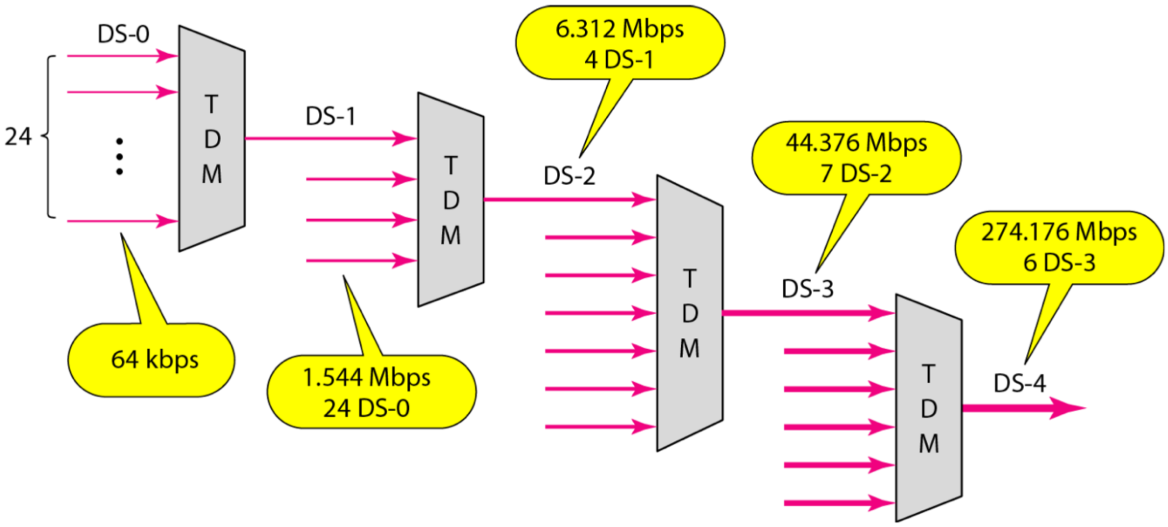


Fig.: Digital Hierarchy

- **DS-0** is a single digital channel of **64 kbps**.
- **DS-1** is a **1.544 Mbps** service. It can be used to multiplex **24 DS-0 channels.**
- **DS-2** is a **6.312 Mbps** service. It can be used to multiplex **4 DS-1 channels.**
- **DS-3** is a **44.376 Mbps** service. It can be used to multiplex **7 DS-2 channels.**
- **DS-4** is a **274.176 Mbps** service. It can be used to multiplex **6 DS-3 channels.**

Unit 7: Data link control and Protocol

Framing:

The data link layer needs to pack bits into frames, so that each frame is distinguishable from another. Our postal system practices a type of framing. The simple act of inserting a letter into an envelope separates one piece of information from another; the envelope serves as the delimiter.

- Framing in the data link layer separate a message from one source to a destination by adding a sender address and a destination address. The destination address defines where the packet is to go; the sender address helps the recipient acknowledge the receipt.
- Frames can be fixed or variable size.
  - Fixed size framing – no need for defining the boundaries of the frames; the size itself can be used as a delimiter.
  - Variable size framing – need a way to define the end of one frame and the beginning of the next.

Flow and Error control:

The most important responsibilities of the data link layer are **flow control** and **error control**. Collectively, these functions are known as **data link control**.

Flow Control:

- Flow control refers to a set of procedures used to restrict the amount of data that the sender can send before waiting for acknowledgment.
  - Preventing buffer overflow
- Transmission time
  - The time to emit all the bits of a frame onto the medium, proportional to the length of the frame.
- Propagation time
  - Time taken for a bit to traverse the link between source and destination.
- The figure below is a vertical time sequence diagram. It shows time dependencies and correct send-receive relationship. Each arrow represents a single frame transiting a data link between two stations. The data are sent in a sequence of frames, with each frame containing a portion of the data and some control information.

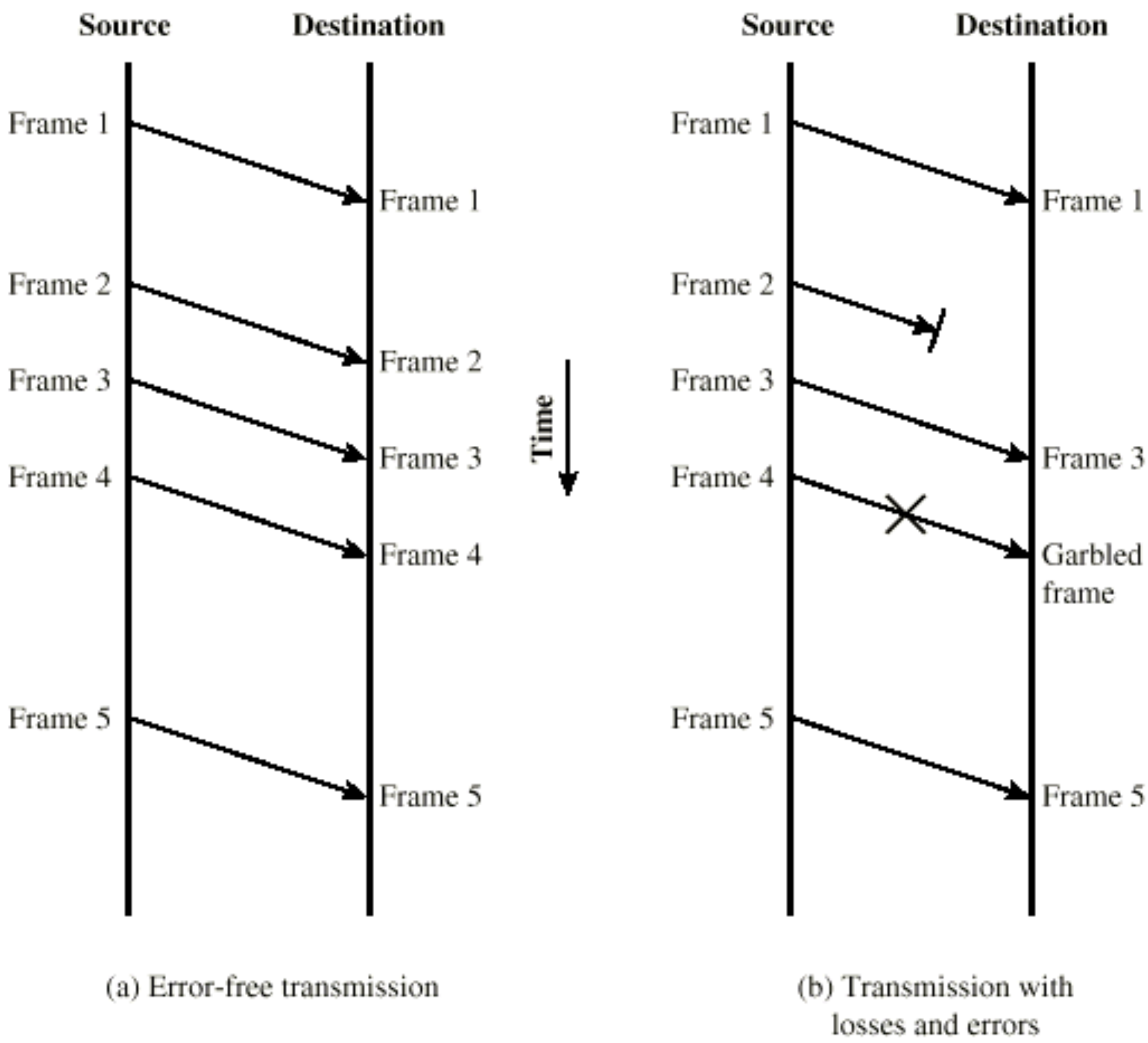
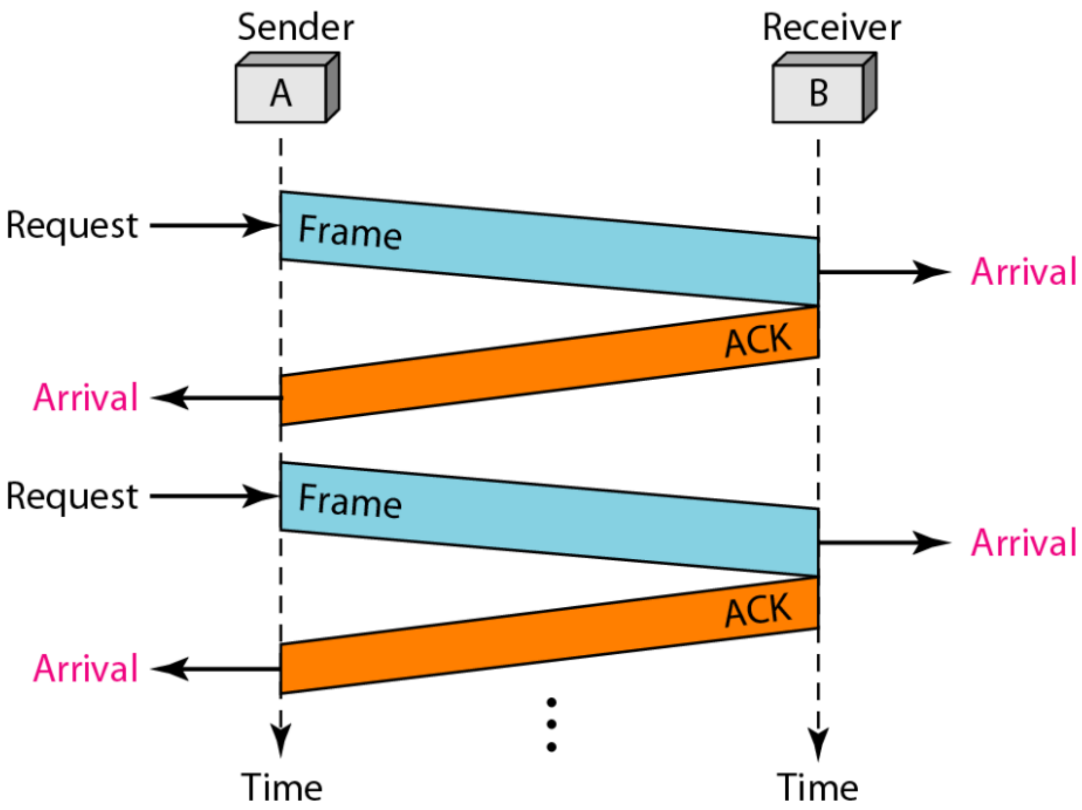


Fig.: Model of Frame Transmission

Types of Flow Control:

1. **Stop and Wait Flow Control**
    - A source entity transmits a frame.
    - After the destination entity receives the frame, it indicates its willingness to accept another frame by sending back an acknowledgement.
    - The source must wait until it receives the acknowledgement before sending the next frame.
    - The destination can thus stop the flow of data simply by withholding acknowledgement.

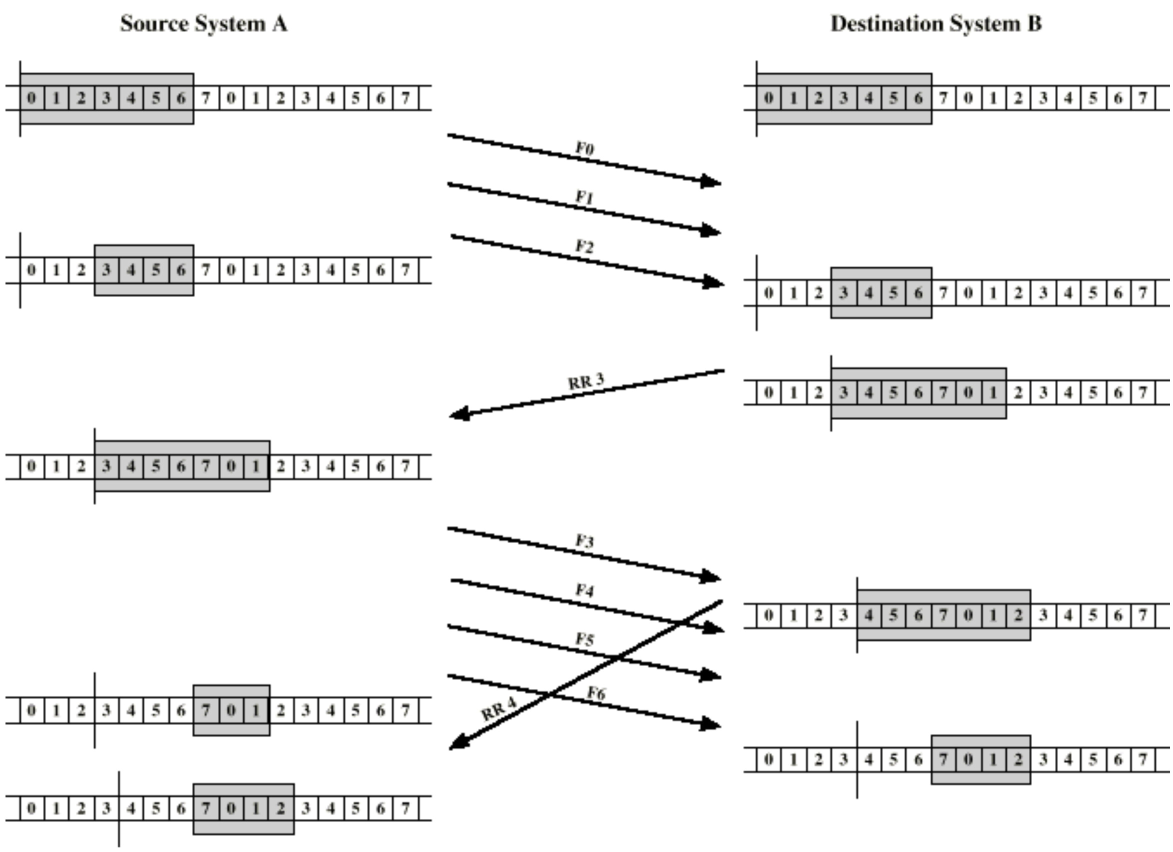


- the source will break up a large block of data into smaller blocks of data into smaller blocks and transmit the data in many frames for the following reasons:
  - limited buffer size
  - On error, retransmission of smaller frames is needed
  - Prevents one station occupying medium for long periods

  DEMERITS:
- With the use of multiple frames for a single message, the stop and wait procedure may be inadequate (essence of the problem: only one frame can be transmitted at a time)
- For very high data rates, for very long distances between sender and receiver, stop and wait flow control provides inefficient line utilization.

2. Sliding windows Flow Control

- Allows multiple frames to be in transit.
- Receiver allocates buffer space for W frames
- Receiver can accept W frames and Sender is allowed to send W frames without waiting for any acknowledgements.
- To keep track of which frames have been acknowledged, each is labelled with a sequence number.
- Receiver acknowledges a frame by sending an acknowledges that includes a sequence number of the next frame expected.
- This acknowledgement also implicitly announces that B is prepared to receive the next W frames, beginning with the number specified.
- This scheme can also be used to acknowledge multiple frames.
- Sender maintains a list of sequence numbers that it is allowed to send and the receiver maintains a list of sequence numbers that it is prepared to receive. Each of these lists can be thought of as a window of frames. So, the name sliding window.
- For a k-bit field, the range of sequence number is 0 through $2^k - 1$, and the frames are numbered modulo 2^k.



Error Control:

- Error control refers to mechanism to detect and correct errors that occur in the transmission of frames. Error may be caused by lost frame or damaged frame.
- The mechanism for error control includes: error detection, positive acknowledgement, retransmission after timeout and negative acknowledgement and retransmission.
- Collectively, these mechanisms are all referred to as **Automatic Repeat Request (ARQ).**

Types of ARQ:

1. **Stop and Wait ARQ**
    - Source transmits single frame
    - Wait for ACK
    - If received frame damaged, discard it
        - Transmitter has timeout
        - If no ACK within timeout, retransmit
    - If Ack damaged, transmitter will not recognize it
        - Transmitter will retransmit
        - Receiver gets two copies of frame
        - To avoid this problem, frames are alternately labelled with 0 and 1, and positive acknowledgements are of the form ACK0 and ACK1.
    - Simple but inefficient
    - Sliding window flow control can be adapted to provide more efficient line use
    - Also referred as **continuous ARQ**

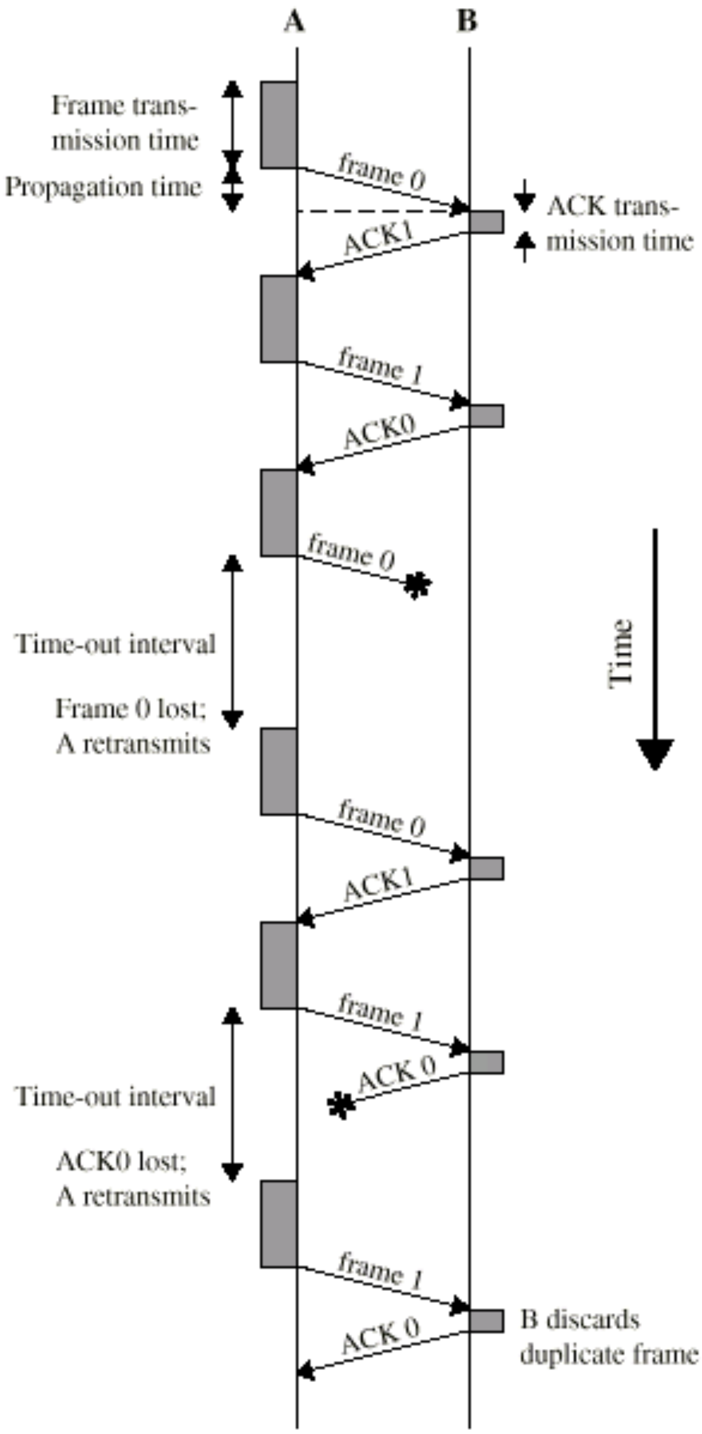


Fig.: Stop and Wait ARQ

2. Go – Back – N ARQ

- Based on sliding window
- If no error, the destination will acknowledge incoming frames as usual (RR = Receive Ready)
- If error detected, destination will send negative acknowledgement (REJ = Reject) for that frame.
    - Discard that frame and all future frames until error frame received correctly
    - Transmitter must go back and retransmit that frame and all subsequent frames.

The Go – Back – N ARQ takes into account the following contingencies:

I. Damaged Frame:

If the received frame is invalid, B discards the frame and takes no further action as the result of that frame. Two subclasses:

a) Within a reasonable period of time, A subsequently sends frame (i+1). B receives (I + 1) out pf order and sends a REJ i. A must retransmit frame I and all subsequent frames.

b) A does not soon send additional frames. B receives nothing and returns neither an RR nor a REJ. When A's timer expires, A just retransmit frame i.

II. Damaged RR:

Two subclasses

a) B gets frame i and send acknowledgement (i + 1) which is lost. Acknowledgement are cumulative, so next acknowledgement may arrive before A times out of frame i.

b) If A's timer expires, it sends acknowledgement with P bit timer. This can be repeated a number of times before a reset procedure is initiated.

III. Damaged REJ:

If a REJ is lost, this is equivalent to case I. b).

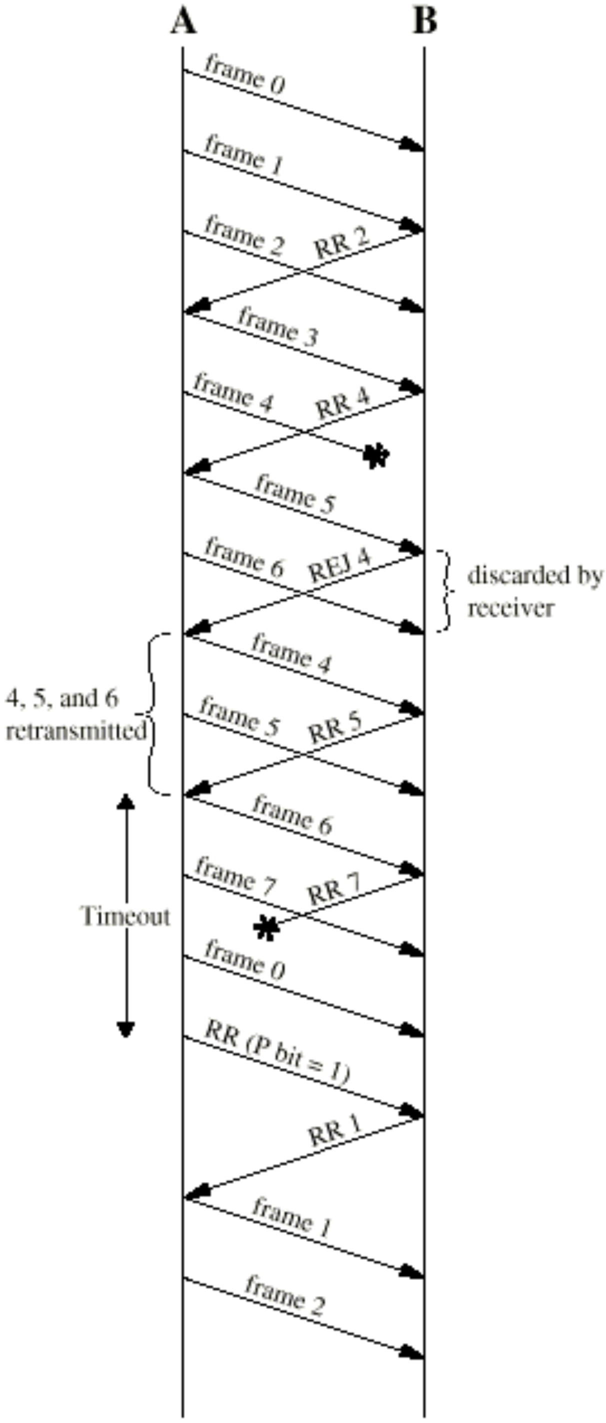


Fig.: Go-back_N ARQ

3. Selective – Reject ARQ

- Also called selective retransmission
- Only rejected frames are retransmitted
- Subsequent frames are accepted by the receiver and buffered
- Minimizes retransmission
- Receiver must maintain large enough buffer
- More complex logic in transmitter

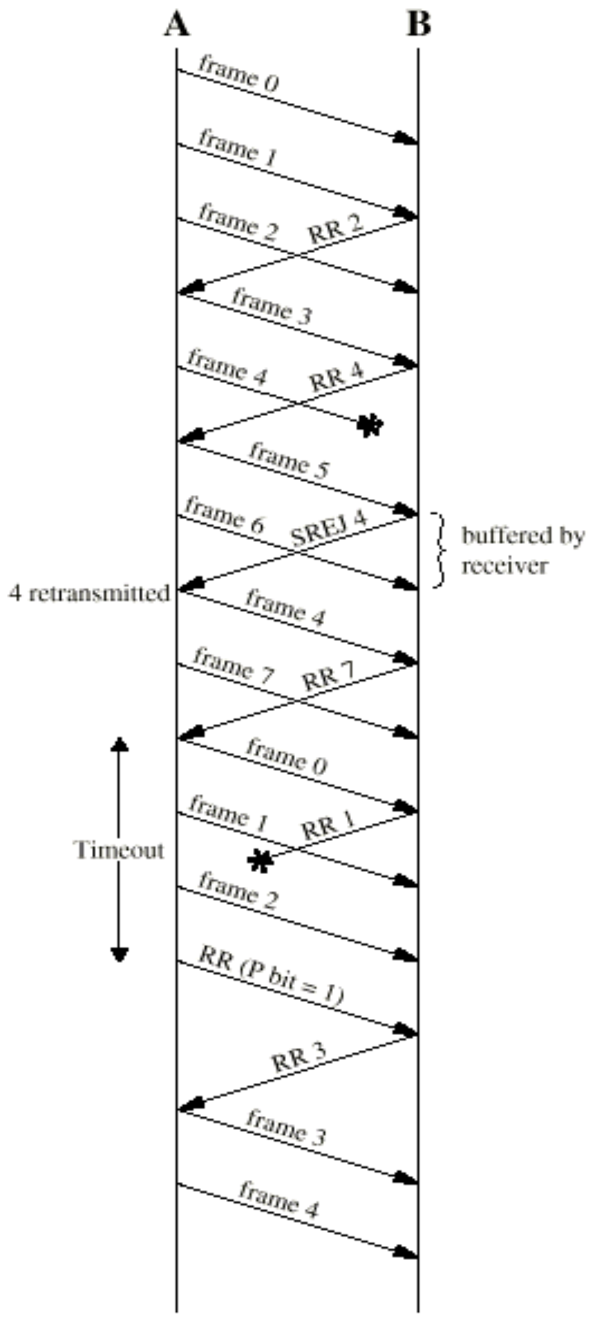


Fig.: Selective – Reject ARQ

Chapter 6:

Lossy compression – Predictive coding and Transform coding

Lossy Compression:

- Lossy compression is more effective than lossless compression
- Lossy compression is used to compress images, video and audio where we can afford to lose some information
- Accuracy is sacrificed to increase the compression rate

Two types:

1. **<u>Predictive Coding:</u>**
   - Predictive coding is used when we digitize an analog signal. Compression can be achieved in the quantization step by using predictive coding.
   - While quantizing, neighbouring quantized sample are closely related and have similar values.
   - In predictive coding, we use this similarity.
   - Instead of quantizing each sample separately (as in PCM), the differences are quantized.
   - The differences are smaller than the actual samples and thus require fewer bits.
   - Different algorithms based on this principle are: Delta Modulation (DM), Adaptive DM, Differential PCM (DPCM), Adaptive DPCM
   - Linear Predictive Coding (LPC):
     - The source determines the characteristics of the signals, such as frequencies in the sensitive range of frequencies, the power of each frequency and duration of each signal.
     - The source quantifies this information and sends to receiver.
     - With the help of signal synthesizer, receiver simulate a signal similar to original signal.
     - High level of compression
2. **Transform Coding:**
   - Mathematical transformation is applied.
   - Transformation needs to be invertible.
   - Changes the signal representation from one domain to another, which results in reducing the number of bits in encoding.
   - To achieve compression goals, quantization is added to make the whole process lossy.
   - Two – Dimensional DCT (Discrete Cosine Transform) is the most popular transformation used in multimedia compression for compressing images, audio and video.

Some examples of lossy compression techniques in different domain

1. **Audio Compression:**
    - MP3: One of the most popular audio compression formats, MP3 (MPEG Audio Layer III) discards certain frequencies that are considered less audible to the human ear.
    - AAC (Advanced Audio Coding): Another common audio compression format that offers better sound quality than MP3 at similar bit rates.
2. **Image Compression:**
    - JPEG (Joint Photographic Experts Group): JPEG is a widely used image compression format that employs lossy compression. It achieves compression by discarding some of the image's high-frequency information.
    - GIF(Graphics Interchange Format): lossy compression due to limited color palette (256 colors).
    - WebP: Developed by Google, WebP is a newer image format that uses both lossy and lossless compression.
3. **Video Compression:**
    - H.264 (Advanced Video Coding): Commonly used for video compression in applications like streaming, H.264 employs lossy compression techniques to reduce file sizes while maintaining acceptable video quality.
    - H.265 (High-Efficiency Video Coding or HEVC): A more recent standard than H.264, H.265 provides better compression efficiency, enabling higher quality at lower bit rates.
    - MPEG (Motion Picture Experts Group): designed to balance compression efficiency with video quality.