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## Unit-4 Data Link Layer & Protocols

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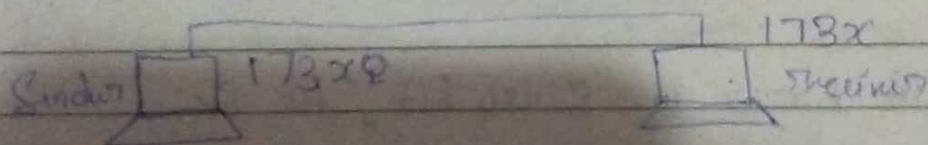
\* Error in Communication :- The message which is transmitted from Sender side, ~~as~~ if it is altered due to some reason between the path (reason may be electromagnetic field, external interference) then that alteration is known as error. This way receiver will not receive that data exactly.

Ex:   
 10110101   
 ↓   
 - This one alter   
 comes to '0'   
 then Error.

Types of Error

- (i) Single bit Error :- There is alteration of only one bit.
- (ii) Burst Error :- Multiple bits are affected simultaneously.

Now question is that, how receiver will know ~~error~~ there is error or bits are altered? How receiver will conclude this. At machine level it is important to know ~~whether~~ whether data is received properly or not.



\* approach of Error handling

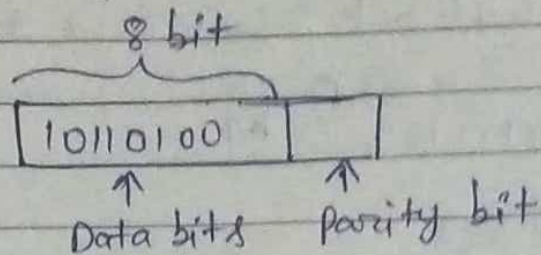
→ Error handling  
 → Error Recovery.



## \* Error Detection technique,

### ① Parity Check technique :-

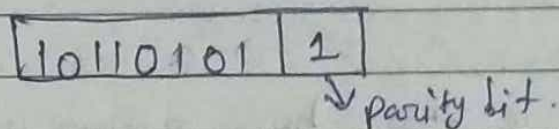
Data is transmitted frame by frame & frame is a logically arranged pattern of data. Bits are contained ~~in~~ in the frame. Frame may contain some control information.



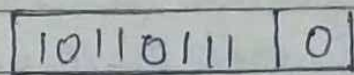
This technique is divided into two Subcategory :-

(i) Even Parity :- It says when a bit pattern is being transmitted the total no. of ones in the bit pattern should be even.

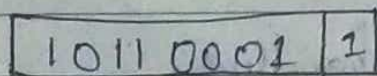
If bits are odd, ~~then~~ then put '1' in parity bit



If bits are already even then, put '0' in parity



when there is error in single bit, for Ex:



no. of 1's are not even so, error <sup>will be detected</sup> and data will be discarded.

When there will be error in two bits.

Ex:

$[10110000|1]$

NO. of 1's are even so error will not be detected.

Hence, we conclude that 50% of the error remains undetected.

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Checksum technique,  
Checking the sum based concept.

\* Binary addition :-

5  $\rightarrow$  101

6  $\rightarrow$  ~~110~~

1011

Carry

0+0=0

0+1=1

1+0=1

1+1=0 (Carry 1)

5  $\rightarrow$  101  
3  $\rightarrow$  + 011  
8  $\rightarrow$  1000

1's Complement : 1011  $\rightarrow$  0100 (1's complement)

101  
010 (1's complement)  
111

adding a binary no. to its 1's complement gives all bits 1.



- ① The binary bit which is being transmitted is divided into 'k' parts of 'n' bits each. for convenience we take 2 parts.
- ② add both the parts.
- ③ find the 1's complement of the sum
- ④ Send this 1's complement with binary data.

At the receiver side when receiver will find out the sum of the bits & 1's complement of the sum which is received with it, all '1' should be obtained.

Ex: Binary data  $\overbrace{1010\ 0001}^{\text{divide in 2 parts}}$

$$\begin{array}{r}
 1010 \\
 + 0001 \\
 \hline
 \text{Sum} \quad 1011 \\
 \text{1's complement} \quad 0100
 \end{array}$$

\* Data sent at receiver's side

10100001 0100

at receiver side,

$$\begin{array}{r}
 1010 \\
 0001 \\
 \hline
 \text{Sum} \rightarrow 1011 \\
 0100 \\
 \hline
 \text{Sum} \rightarrow 1111
 \end{array}$$

When MSB are exchanged?

- ① Receiver will again divide the bits in two parts
- ② Partwise sum is calculated.
- ③ Then calculated sum is added to the received sum which is actually 1's complement of the sum at sender's side
- ④ If all the bits come to be 1 then data will be received otherwise discarded.



## \* Flow & Error Control

Bandwidth help to find the bit rate of data. Sender se likar receiver tak Jo data travel karta h to jo media implemented hai ek hi bandwidth ka nahi hoga. Use bandwidth ka Specification alog alog ho Sakta hai. Also, It is possible that data sender is sending & receiver is receiving there architectural Configuration may be different. Also, Speed will be same, it is not at all necessary.

In the implementation of network, the architectural platform will be different for Station to Station. Sender jis Speed se data bhej raha hai receiver usi Speed se Propagate Kariga Smoothly, ye Jaruri nahi hai. Jo Ensure kar bhi to to receiver ke pass usi Speed se receive hoga ye bhi Jaruri nahi hai. Processing Speed may vary becoz many devices participate in network.

\* Flow Control  $\div$  It says control the flow of bits becoz the flow of data should be properly controlled. It is necessary becoz the sender which is sending the data should be properly received and processed by the receiver. Flow Control mean limit the flow of data, upto which amount that receiver can easily process in each processing cycle. Ya phir use amount ~~to~~ limit karu, jis amount me aage bandwidth allow kar rahi h data ko aage bhejne ke liye. Limit the Speed of data in any way.



## Error Control,

- (i) noiseless channel
  - (ii) noisy channel
- } Transmission channel.

(i) noiseless  $\div$  no external interference. So there is no chances of error. It requires only flow control mechanism. They don't exist in practical manner. It is only for logical understanding.

(ii) noisy channel  $\div$  External interference exists. Here the Probability of error, error bandwidth specification are different. Includes both flow & error control.

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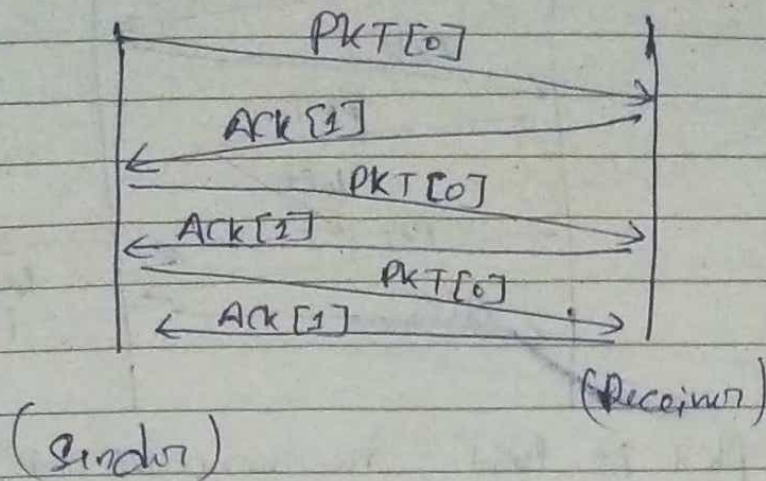
## Stop & Wait protocol,

It is protocol of flow control. Protocol means a set of rules. In this sender will send some data and wait upto the time the receiver is able to process data then receive data & it will become ready to receive further data from sender's side. So that channel do not get overloaded.

When Stop & wait protocol is implemented sender is allowed to send a packet, after transmission of 1<sup>st</sup> packet, sender will stop & then sender will wait for the acknowledgement from receiver's side.

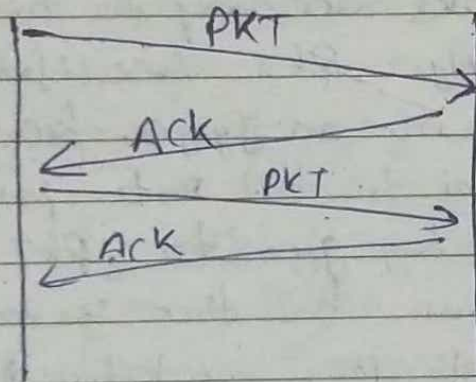
Acknowledgement (Ack) is that particular information that informs sender about the successful receiving & processing of the packet (PKT) and now receiver is ready to receive the next PKT. So sender is allowed to send the next ~~packet~~ PKT.





Since noisy channel is always busy. there may be probability of external interference. So, always there is a prob. that there is error in PKT, PKT loss etc.

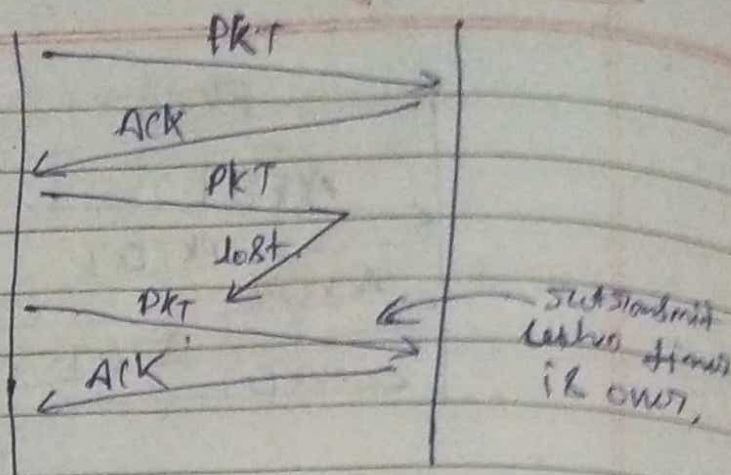
Case: 1  
ideal case



Sender ek PKT bhejta hai aur stop kr jata hai. Stop karke wait krta hai, kab tak, jab tak receiver ke side se ACK nahi aa jaye. To kyo aisa h ki sender wait hi krta rahuga kaafi samay tak jab tak receiver Acknowledge nahi kruga. To isme ek timer ka concept include hota hai.



Case: 2  
Packet loss

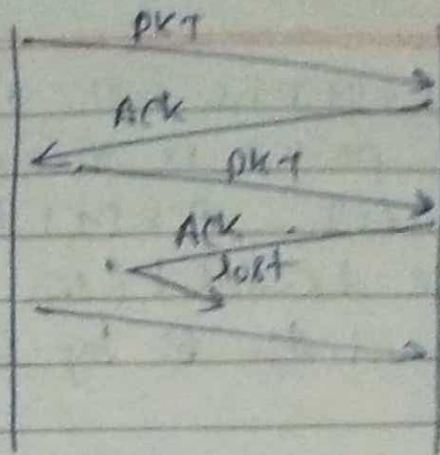


Now, when ~~for~~ PKT is lost, receiver will not get PKT so receiver will not Acknow. Sender has stop after sending PKT. <sup>But</sup> Sender ko nahi malum that in between path PKT is lost. Sender is waiting for ACK and Receiver is also waiting for PKT. ACK nhi pahuncha to Sender wait hi karke reh jayega. Is liye timer ka ek side h. We need a timer, if ACK doesn't come till this time. Timer works according to bandwidth & processing speed. Agar PKT successfully pahunch jayega to ACK itni time me aa jayega. becoz we know bandwidth known hai to bit rate pata lag jata hai. Agar bit rate pata lag gya to, PKT me bit & ka specification agar known hai then ye pata lag jayega, ki PKT ko receiver tak janne me kitna time lagiga. and same with the ACK. So if we add both the time of PKT and ACK including some marginal time we can get the timing for timer.

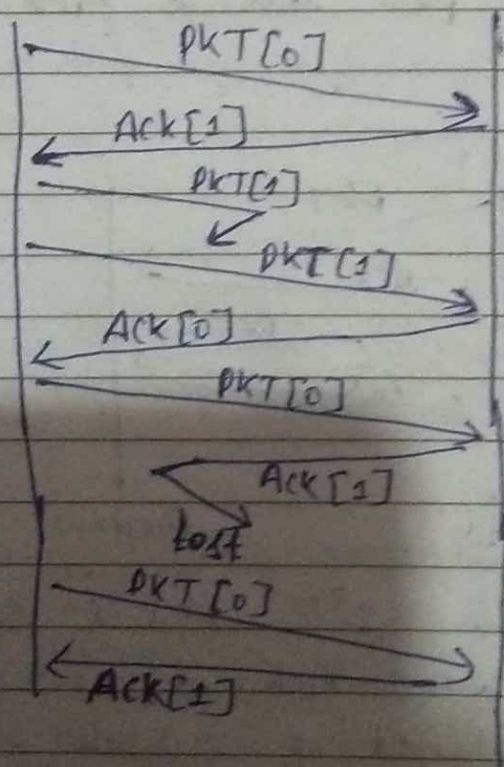
In case of PKT loss, Sender will send the PKT & wait till timer ~~is over~~ is over. If timer is over & still ACK is not received, Sender will assume that PKT is lost somewhere. So, Sender will retransmit the PKT. Again ACK will be received and transmission will continue.



Case: 3



In this Case, PKT is sent successfully at receiver side but ACK sent by receiver is lost. So when timer is over, sender will assume that PKT is not reached at receiver. So it will retransmit the same PKT bcoz ACK is not received but PKT is received at receiver side. So, there is a case of PKT duplication at the receiver's side. So we need a Strategy to manage PKT duplication.



When PKT[0] is transmitted, it is acknowledged by ACK[1]. It means receiver is expecting

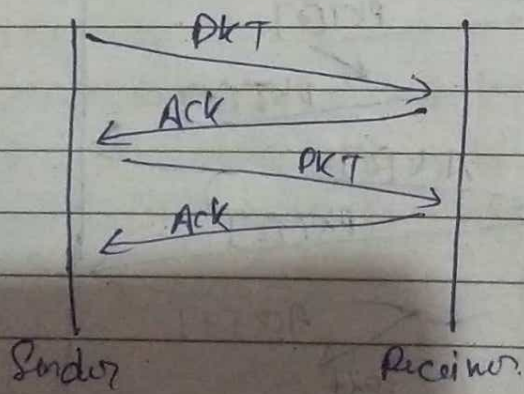


pkT[1] . pkT[1] is lost. So pkT[2] again sent. Ack[0] means packet is received. No, receiver is expecting pkT[0]. Ack[1] lost. Now pkT[0] again sent packet is discarded (No, pkT[0]). Now receiver ~~is~~ acknowledge ~~is~~ by Ack[1] and expecting pkT[2].

Ack[0] → informs receiver is expecting pkT[0]  
Ack[1] → receiver is expecting pkT[1].

- Q.) What is Stop & wait?
- Q.) What is flow control & role of Stop & wait in that?
- Q.) What are cases of Stop & wait?
- Q.) What is the use of acknowledging pkT[1] by Ack[0] & pkT[0] by Ack[1].
- Q.) How cases of pkT loss & Ack loss handled.

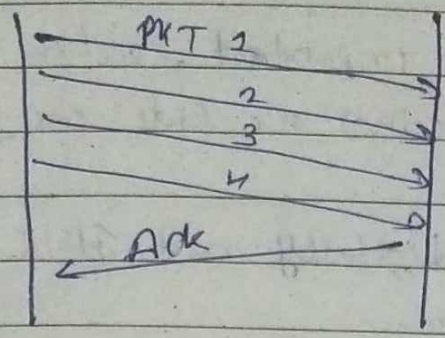
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In this case, when 1<sup>st</sup> pkT is sent we get 1<sup>st</sup> Ack and for 2<sup>nd</sup> pkT after Ack is sent. So there are four transmission out of which two are message part and rest two are acknowledgement part. Total 4 pkT bhai kaha the waha 2 hi bheje aur 2



Acknowledgment ke or liye use kiye. So bandwidth utilization reduced to 50% in ideal case and reduced more in case of PKT loss & Ack lost.

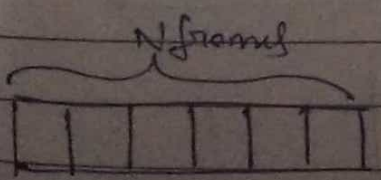
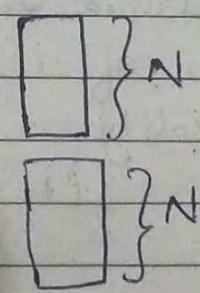


4 PKT send at one time & one Ack, then Efficiency increases to 80%.

So if we ~~use~~ acknowledge multiple PKT's with single Acknowledgment then bandwidth Efficiency can be improved.

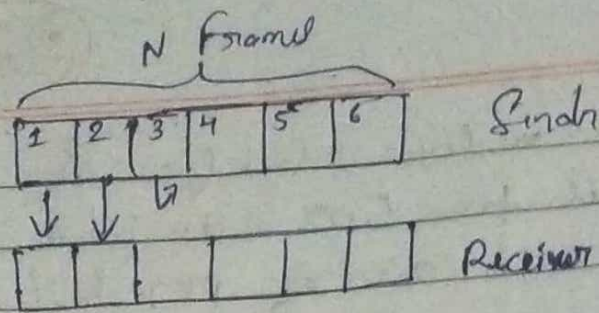
### \* GO-BACK-N ARQ (or sliding window protocol)

'N' no. of PKTs are transmitted in one time. 'N' no. of packets transmitted and we get a Cumulative acknowledgment <sup>which</sup> ~~that~~ acknowledge that 'N' no. of frames has reached.



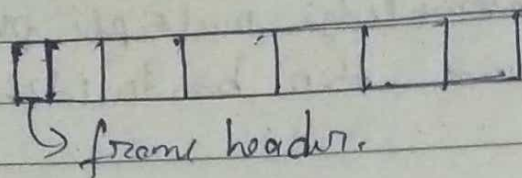
Send window + Logical arrangement of N frames is called Send window. It is an abstract concept. A kind of data structure, inside which frames are logically arranged. We will send our windows when we will get Ack for that proceed to next window.





Numbering of frame is important, when properly numbered then only receiver ke side me finally compare kar payega.

Transmission happens digitally and It is based on binary number system.



If we use 2 bit how many frames we can make.  
So, a window consist of 4 frames in this case. So, window size depends on the no. of bits used.

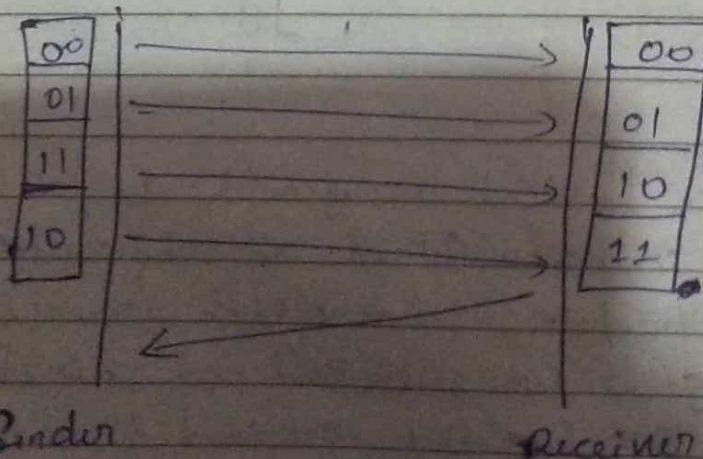
If we use,  $n$  bits in frame header, how many frames can be addressed uniquely.

So, size of window ~~should be~~ should be  $2^n$ .

But in reality,

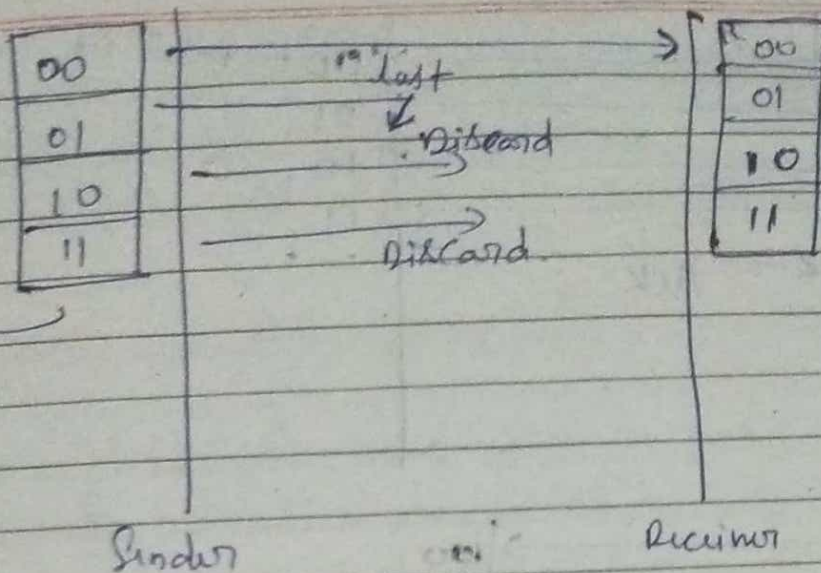
It is ~~2^n~~  $2^n - 1$

Ideal Case:

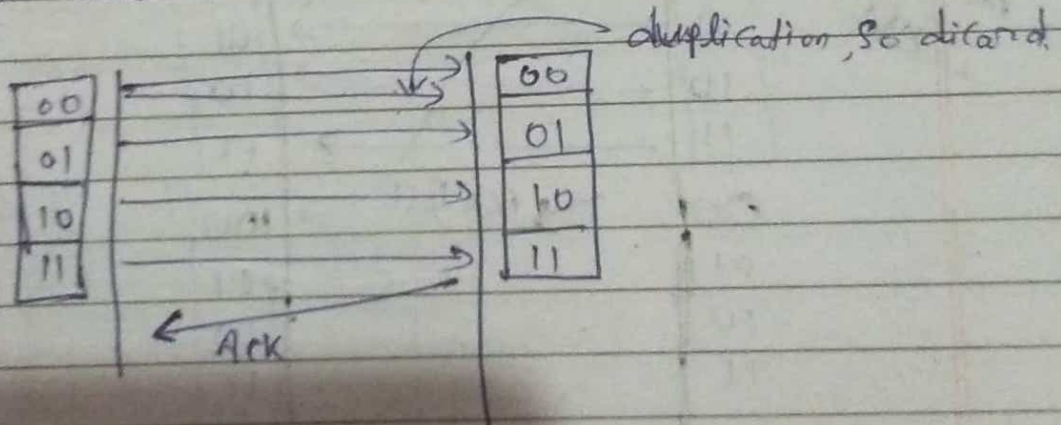




Case: 2  
Frame lost



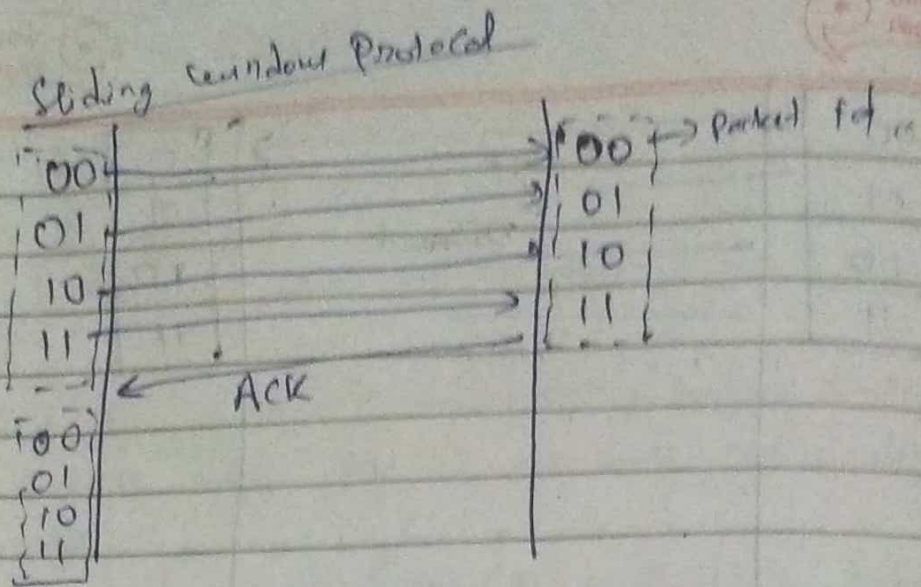
In this case, one frame is successfully received & 2<sup>nd</sup> frame is lost, now receiver is expecting frame with frame header 01. Sender doesn't know that ~~frame~~ frame is lost and will send frame [10] and [11] but receiver will discard them, and receiver will not acknowledge. Now, sender will wait for Ack till the timer is over. When timer is over and Ack is not received then sender will assume frame is lost. So, sender will Go-Back-N i.e., go to N<sup>th</sup> place.



It will again send frame from N<sup>th</sup> place and if receiver has already received that frame then it will discard that frame and transmit the next frame.

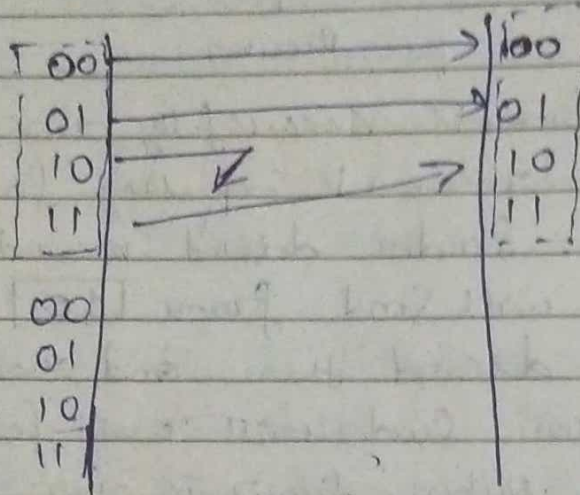


# Division Case: 1



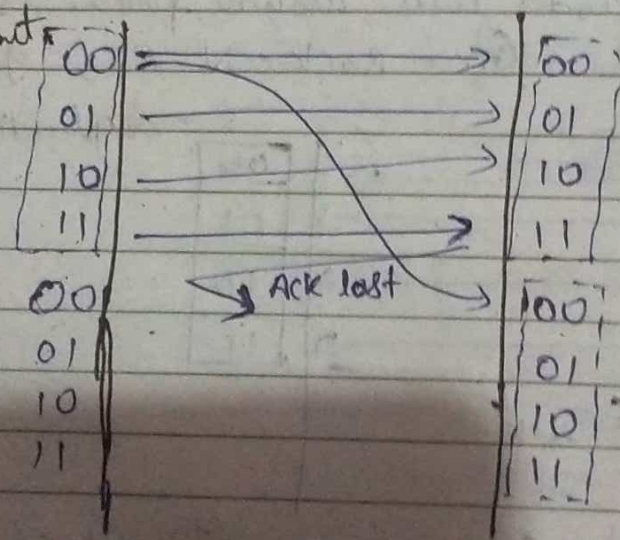
## Case: 2

Go-Back N



## Case: 3

Acknowledgment loss



when ACK is lost sender assume karaga ki packet loss. he gya. the sender will again go back to n<sup>th</sup> place and send 00 (pkt Id) of first window and will be received by the 2<sup>nd</sup> window. Since, pkt Id 00 (of 2<sup>nd</sup> window matched) then it will be received.

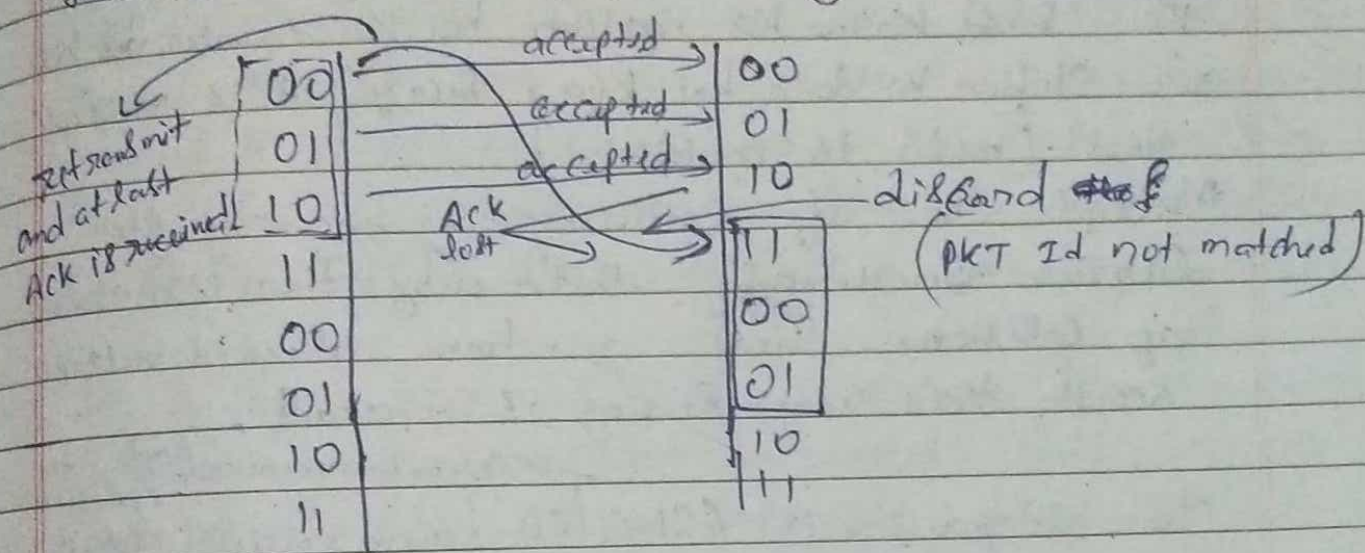


So there will be a wrong receiving. Sender is resubmitting the 1st window again.

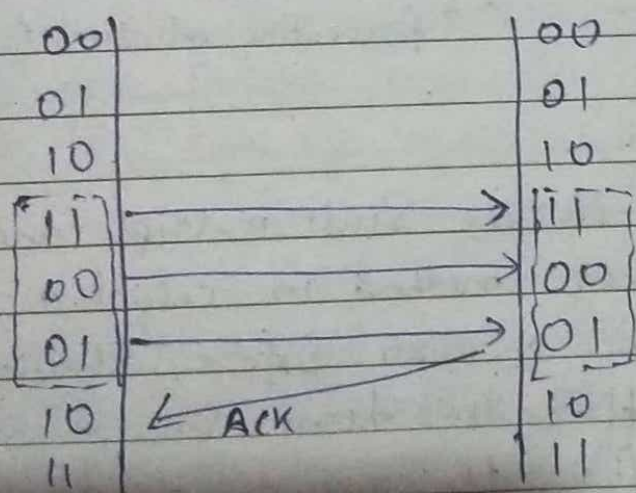
This is the drawback of Go-Back-N Protocol.

To overcome this drawback, :

If  $n$  bits are used for  $n$  pkt's,  $2^n$  <sup>Packets</sup> bits can be uniquely addressed as window size is  $2^n$ . Logically this is correct but, practically it is  $2^n - 1$



Slide to next window.



In case of Ack lost, if Ack is lost & we kept the size of window  $n$  then there will be duplication of packet and receiver will receive wrong packet sequence. Instead of  $2^n - 1$  it will correctly receive  $n$  packets.