COMPUTER NETWORKS

UNIT - 2: DATA LINK LAYER

Syllabus:

Data link layer: Design issues, Framing: fixed size framing, variable size framing, flow control, error control, error detection and correction codes, CRC, Checksum: idea, one's complement internet checksum services provided to Network Layer,

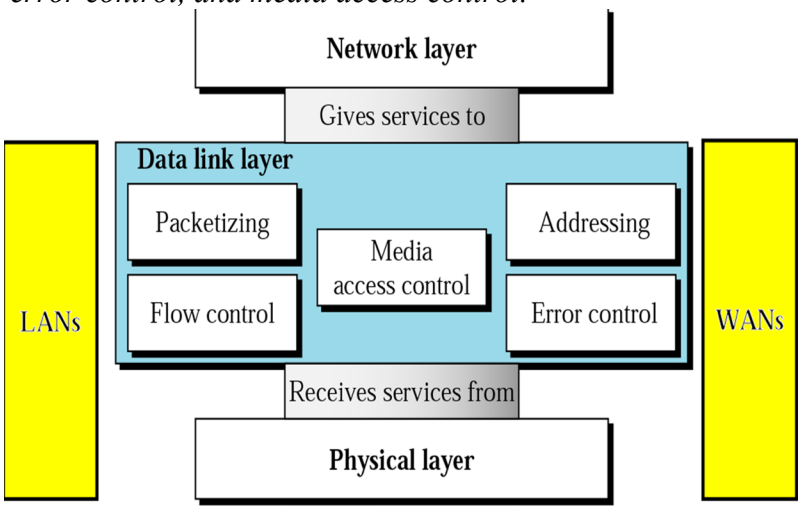
Elementary Data Link Layer protocols: simplex protocol, Simplex stop and wait, Simplex protocol for Noisy Channel.

Sliding window protocol: One bit, Go back N, Selective repeat-Stop and wait protocol, Data link layer in HDLC: configuration and transfer modes, frames, control field,

Point to point protocol (PPP): framing, transition phase, multiplexing, multi-link PPP.

DATA LINK LAYER

The data link layer transforms the physical layer, a raw transmission facility, to a link responsible for node-to-node (hop-to-hop) communication. Specific responsibilities of the data link layer include *framing, addressing, flow control, error control, and media access control.*

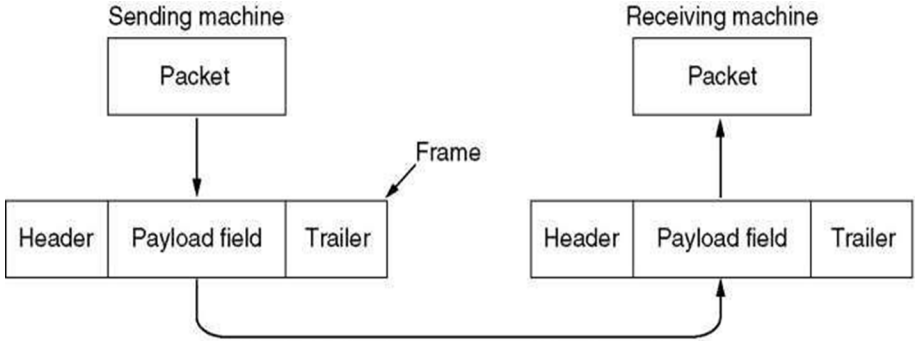


DATA LINK LAYER DESIGN ISSUES

The data link layer uses the services of the physical layer to send and receive bits over communication channels. It has a number of functions, including:

1. Providing a well-defined service interface to the network layer.
2. Dealing with transmission errors.
3. Regulating the flow of data so that slow receivers are not swamped by fast senders.

To accomplish these goals, the data link layer takes the packets it gets from the network layer and encapsulates them into frames for transmission. Each frame contains a frame header, a payload field for holding the packet, and a frame trailer, as illustrated in Fig.



SERVICES PROVIDED TO THE NETWORK LAYER

The function of the data link layer is to provide services to the network layer. The principal service is transferring data from the network layer on the source machine to the network layer on the destination machine.

The data link layer can be designed to offer various services. The actual services that are offered vary from protocol to protocol. Three reasonable possibilities that we will consider in turn are:

1. Unacknowledged connectionless service.
2. Acknowledged connectionless service.
3. Acknowledged connection-oriented service.

1. Unacknowledged connectionless service:

Unacknowledged connectionless service consists of having the source machine send independent frames to the destination machine without having the destination machine acknowledge them. Ethernet is a good example of a data link layer that provides this class of service. No logical connection is established beforehand or released afterward. If a frame is lost due to noise on the line attempt is made to detect the loss or recover from it in the data link layer. This class of service is appropriate when the error rate is very low, so recovery is left to higher layers.

2. Acknowledged connectionless service:

When this service is offered, there are still no logical connections used, but each frame sent is individually acknowledged. In this way, the sender knows whether a frame has arrived correctly or been lost. If it has not arrived within a specified time interval, it can be sent again. This service is useful over unreliable channels, such as wireless systems. 802.11 (WiFi) is a good example of this class of service.

3. Acknowledged connection-oriented service:

The most sophisticated service the data link layer can provide to the network layer is connection-oriented service. With this service, the source and destination machines establish a connection before any data are transferred. Each frame sent over the connection is numbered, and the data link layer guarantees that each frame sent is indeed received. Furthermore, it guarantees that each frame is received exactly once and that all frames are received in the right order.

When connection-oriented service is used, transfers go through three distinct phases. In the first phase, the connection is established by having both sides initialize variables and counters needed to keep track of which frames have been received and which ones have not. In the second phase, one or more frames are actually transmitted. In the third and final phase, the connection is released, freeing up the variables, buffers, and other resources used to maintain the connection.

FRAMING

To provide service to the network layer, the data link layer must use the service provided to it by the physical layer. What the physical layer does is accept a raw bit stream and attempt to deliver it to the destination. If the channel is noisy, as it is for most wireless and some wired links, the physical layer will add some redundancy to its signals to reduce the bit error rate to a tolerable level. However, the bit stream received by the data link layer is not guaranteed to be error free. Some bits may have different values and the number of bits received may be less than, equal to, or more than the number of bits transmitted. It is up to the data link layer to detect and, if necessary, correct errors.

The data link layer to break up the bit stream into discrete frames, compute a short token called a checksum for each frame, and include the checksum in the frame when it is transmitted.

When a frame arrives at the destination, the checksum is recomputed. If the newly computed checksum is different from the one contained in the frame, the data link layer knows that an error has occurred and takes steps to deal with it.

DLL translates the physical layer's raw bit stream into discrete units (messages) called **frames.**

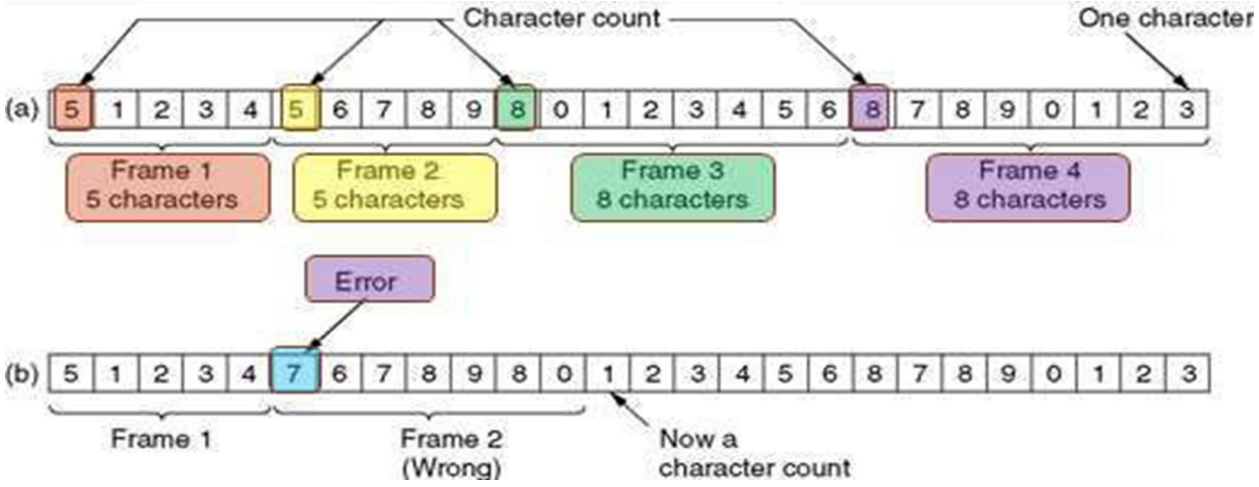
A good design must make it easy for a receiver to find the start of new frames while using little of the channel bandwidth. We will look at **four methods:**

1. **Byte count.**
2. **Flag bytes with byte stuffing.**
3. **Flag bits with bit stuffing.**
4. **Physical layer coding violations.**

1. Byte count (Character Count) :

This framing method uses a field in the header to specify the number of bytes in the frame. When the data link layer at the destination sees the byte count, it knows how many bytes follow and hence where the end of the frame is. This technique is shown in Fig.(a) For four small example frames of sizes 5, 5, 8, and 8 bytes, respectively.

The trouble with this algorithm is that the count can be garbled by a transmission error. For example, if the byte count of 5 in the second frame of Fig.(b) becomes a 7 due to a single bit flip, the destination will get out of synchronization. It will then be unable to locate the correct start of the next frame.

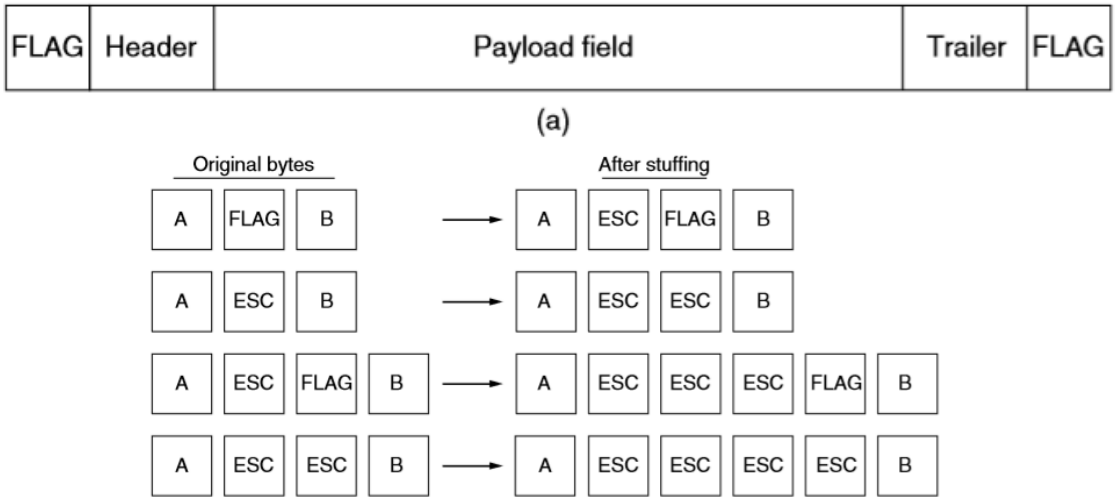


2. Flag bytes with byte stuffing:

This framing method gets around the problem of resynchronization after an error by having each frame start and end with special bytes. Often the same byte, called a **flag byte**, is used as both the starting and ending delimiter. This byte is shown in Fig.(a) as **FLAG.** Two consecutive flag bytes indicate the end of one frame and the start of the next. Thus, if the receiver ever loses synchronization it can just search for two flag bytes to find the end of the current frame and the start of the next frame.

However, there is a still a problem we have to solve. It may happen that the flag byte occurs in the data, especially when binary data such as photographs or songs are being transmitted. This situation would interfere with the framing. One way to solve this problem is to have the sender's data link layer insert a special escape byte (ESC) just before each ''accidental'' flag byte in the data.

The data link layer on the receiving end removes the escape bytes before giving the data to the network layer. This technique is called **byte stuffing.**



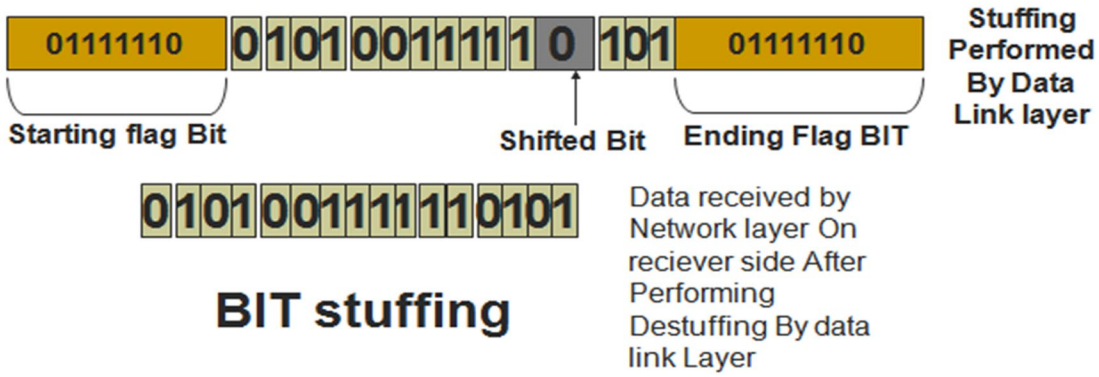
Four examples of byte sequences before and after byte stuffing.

3. Flag bits with bit stuffing:

Framing can be also be done at the bit level, so frames can contain an arbitrary number of bits made up of units of any size. It was developed for the once very popular **HDLC (Highlevel Data Link Control)**

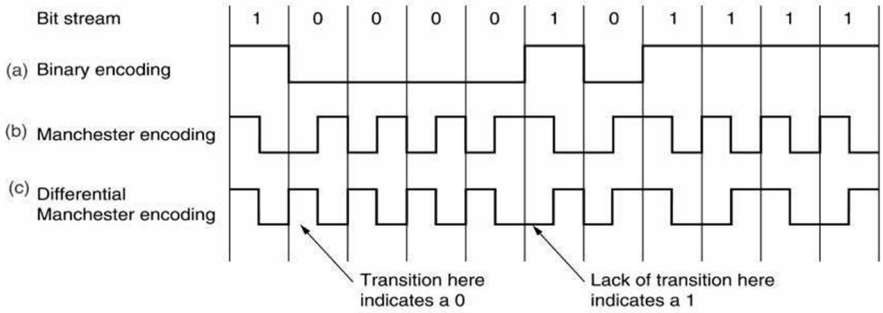
protocol. Each frame begins and ends with a special bit pattern, 01111110 or 0x7E in hexadecimal. This pattern is a flag byte. Whenever the sender's data link layer encounters five consecutive 1s in the data, it automatically stuffs a 0 bit into the outgoing bit stream. This **bit stuffing** is analogous to byte stuffing, in which an escape byte is stuffed into the outgoing character stream before a flag byte in the data. It also ensures a minimum density of transitions that help the physical layer maintain synchronization. USB (Universal Serial Bus) uses bit stuffing for this reason.

When the receiver sees five consecutive incoming 1 bits, followed by a 0 bit, it automatically destuffs (i.e., deletes) the 0 bit. Just as byte stuffing is completely transparent to the network layer in both computers, so is bit stuffing. If the user data contain the flag pattern, 01111110, this flag is transmitted as 011111010 but stored in the receiver's memory as 01111110. Figure gives an example of bit stuffing.



4. Physical layer coding violations:

- This Framing Method is used only in those networks in which Encoding on the Physical Medium contains some redundancy.
- Some LANs encode each bit of data by using two Physical Bits i.e. Manchester coding is used. Here, Bit 1 is encoded into high- low (10) pair and Bit 0 is encoded into low-high (01) pair.
- The scheme means that every data bit has a transition in the middle, making it easy for the receiver to locate the bit boundaries. The combinations high-high and low-low are not used for data but are used for delimiting frames in some protocols.



ERROR CONTROL:

Error control is concerned with ensuring that all frames are eventually delivered (possibly in order) to a destination. How? Three items are required.

- **Acknowledgements:** Typically, reliable delivery is achieved using the "acknowledgments with retransmission" paradigm, whereby the receiver. returns a special acknowledgment (ACK) frame to the sender indicating the correct receipt of a frame. In some systems, the receiver also returns a negative acknowledgment (NACK) for incorrectly-received frames. This is nothing more than a hint to the sender so that it can retransmit a frame right away without waiting for a timer to expire.
- **Timers:** One problem that simple ACK/NACK schemes fail to address is recovering from a frame that is lost? Retransmission timers are used to resend frames that don't produce an ACK. When sending a frame, schedule a timer to expire at some time after the ACK should have been returned. If the timer goes o, retransmit the frame.

- **Sequence Numbers:** Retransmissions introduce the possibility of duplicate frames. To suppress duplicates, add sequence numbers to each frame, so tha1t 9a receiver can distinguish between new frames and old copies.

FLOW CONTROL:

Flow control deals with controlling the speed of the sender to match that of the receiver.

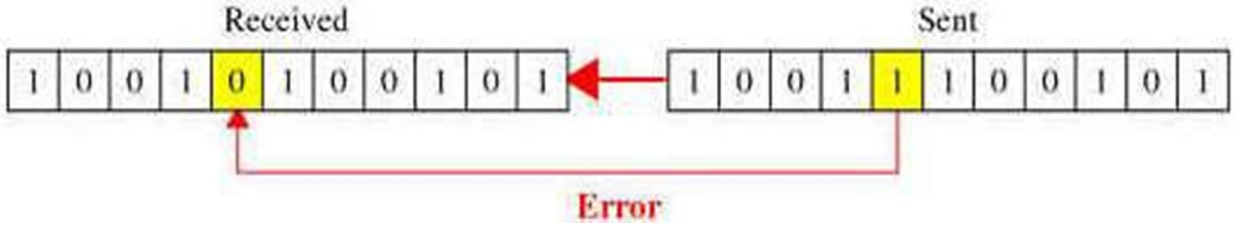
Two Approaches:

- **feedback-based flow control**, the receiver sends back information to the sender giving it permission to send more data or at least telling the sender how the receiver is doing
- **rate-based flow control**, the protocol has a built-in mechanism that limits the rate at which senders may transmit data, without using feedback from the receiver.

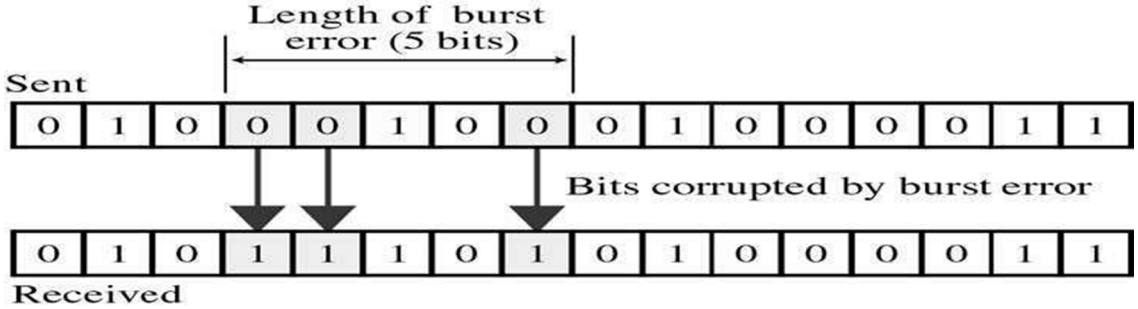
TYPES OF ERRORS:

There are two main types of errors in transmissions:

1. **Single bit error:** It means only one bit of data unit is changed from 1 to 0 or from 0 to 1.



2. **Burst error:** It means two or more bits in data unit are changed from 1 to 0 from 0 to 1. In burst error, it is not necessary that only consecutive bits are changed. The length of burst error is measured from first changed bit to last changed bit

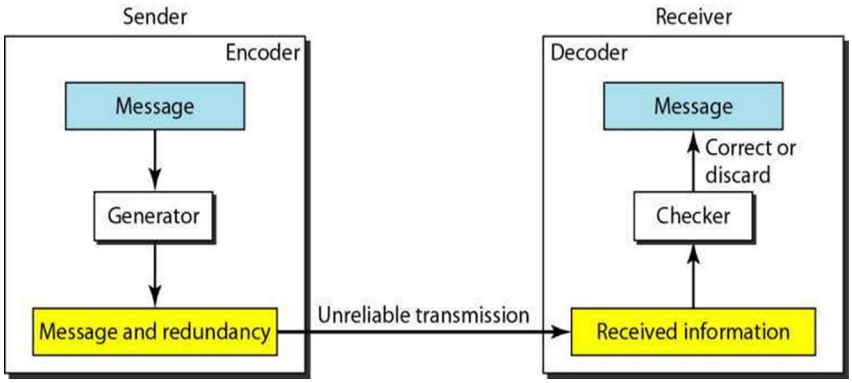


ERROR DETECTION AND CORRECTION:

Network designers have developed two basic strategies for dealing with errors. Both add redundant information to the data that is sent. One strategy is to include enough redundant information to enable the receiver to deduce what the transmitted data must have been. The other is to include only enough redundancy to allow the receiver to deduce that an error has occurred and have it request a retransmission. The former strategy uses **error-correcting codes** and the latter uses **error-detecting codes**.

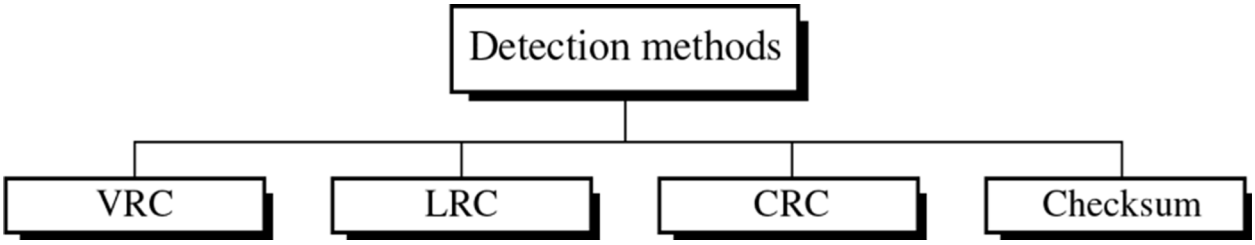
Error Detecting Codes: Include enough redundancy bits to *detect* errors and use ACKs and retransmissions to recover from the errors.

Error Correcting Codes: Include enough redundancy to detect and correct errors.



ERROR-DETECTING CODES:

Error detection means to decide whether the received data is correct or not without having a copy of the original message. Error detection uses the concept of redundancy, which means adding extra bits for detecting errors at the destination.



1. **Vertical Redundancy Check(VRC):**

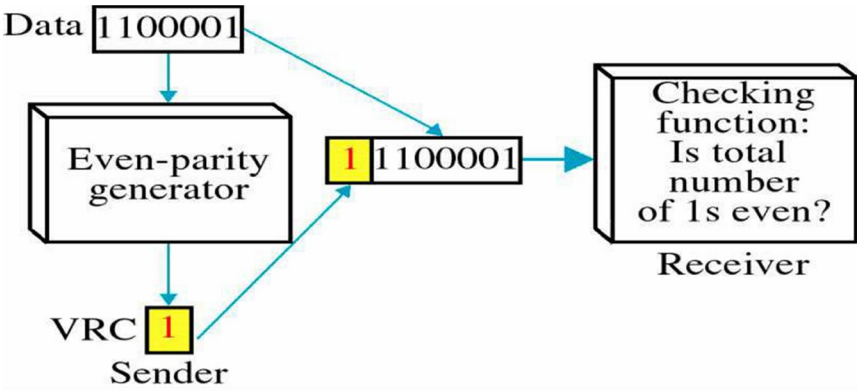
Append a single bit at the end of data block such that the number of one's is even

→À Even Parity (odd parity is similar)

0110011 →01100110

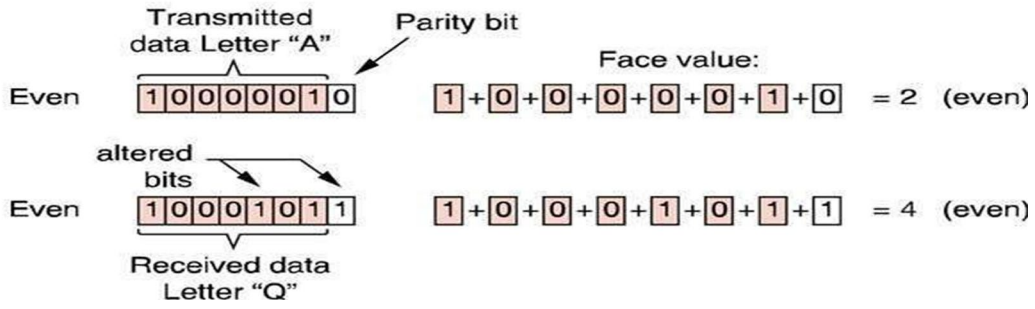
0110001 → 01100011

VRC is also known as **Parity Check**. Detects all odd-number errors in a data block.



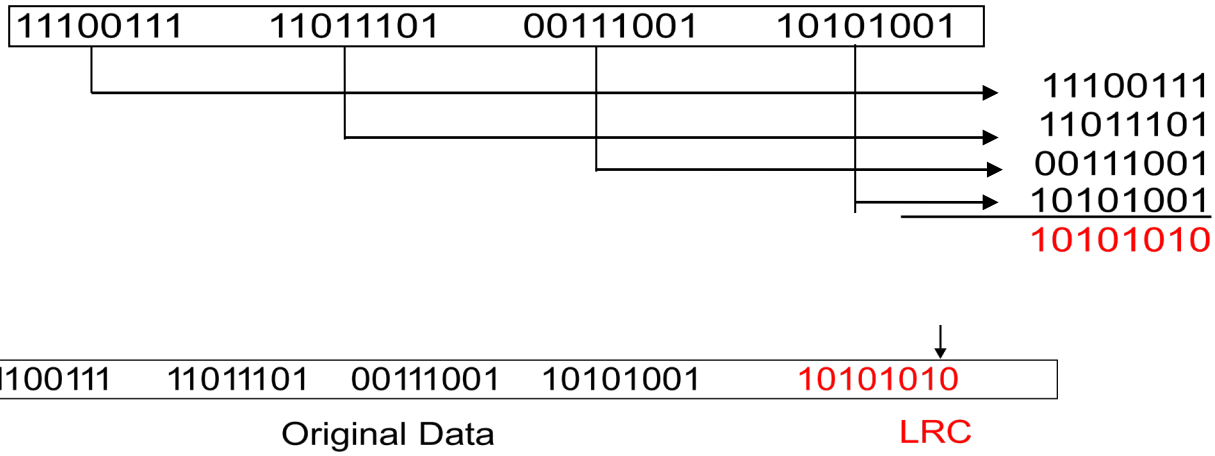
The problem with parity is that it can only detect odd numbers of bit substitution errors, i.e. 1 bit, 3bit, 5, bit, etc. Errors. If there two, four, six, etc. bits which are transmitted in error, using VRC will not be able to detect the error.

Multiple bit errors/even parity



2. Longitudinal Redundancy Check(LRC):

- Longitudinal Redundancy Checks (LRC) seek to overcome the weakness of simple, bit-oriented, one-directional parity checking.
- LRC adds a new character (instead of a bit) called the Block Check Character (BCC) to each block of data. Its determined like parity, but counted longitudinally through the message (also vertically)
- Its has better performance over VRC as it detects 98% of the burst errors (>10 errors) but less capable of detecting single errors



3. Cyclic Redundancy Check(CRC):

The cyclic redundancy check, or CRC, is a technique for detecting errors in digital data, but not for making corrections when errors are detected. The CRC (**Cyclic Redundancy Check**), also known as a **polynomial code**

Polynomial codes are based upon treating bit strings as representations of polynomials with coefficients of 0 and 1 only. For example, 110001 has 6 bits and thus represents a six-term polynomial with coefficients 1, 1, 0, 0, 0, and 1: **1x5 + 1x4 + 0x3 + 0x2 + 0x1 + 1x0.**

Polynomial arithmetic is done modulo 2, according to the rules of algebraic field theory. It does not have carries for addition or borrows for subtraction. Both addition and subtraction are identical to **exclusive OR.** For example:

10011011	**00110011**	**11110000**	**01010101**
+ 11001010	**+ 11001101**	**− 10100110**	**− 10101111**
01010001	**11111110**	**01010110**	**11111010**

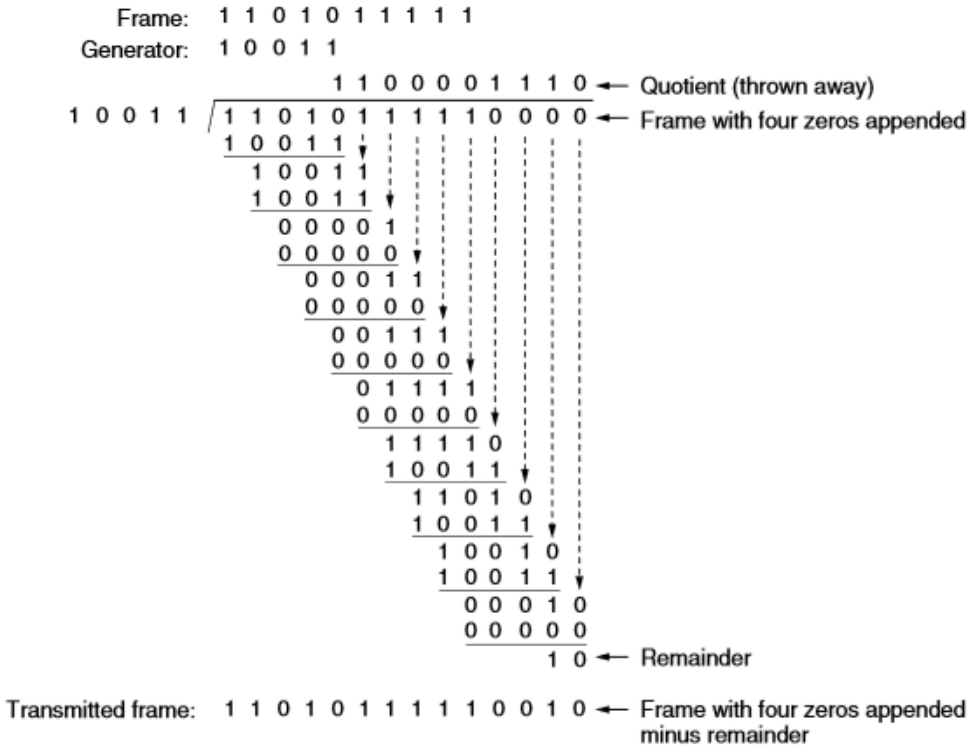
When the polynomial code method is employed, the sender and receiver must agree upon a **generator polynomial, G(x),** in advance. Both the high- and low order bits of the generator must be 1. To compute the CRC for some frame with m bits corresponding to the polynomial M(x), the frame must be longer than the generator polynomial. The idea is to append a CRC to the end of the frame in such a way that the polynomial represented by the check summed frame is divisible by G(x). When the receiver gets the check summed frame, it tries dividing it by G(x). If there is a remainder, there has been a transmission error.

The **algorithm for computing the CRC is as follows:**

1. Let r be the degree of G(x). Append r zero bits to the low-order end of the frame so it now contains m + r bits and corresponds to the polynomial $x^rM(x)$.
2. Divide the bit string corresponding to G(x) into the bit string corresponding to $x^rM(x)$, using modulo 2 divisions.

3. Subtract the remainder (which is always r or fewer bits) from the bit string corresponding to $x^rM(x)$ using modulo 2 subtractions. The result is the check summed frame to be transmitted. Call its polynomial T(x).

Below figure illustrates the calculation for a frame 1101011111 using the generator $G(x) = x4 + x + 1$.



Example calculation of the CRC.

It should be clear that T(x) is divisible (modulo 2) by G(x). In any division problem, if you diminish the dividend by the remainder, what is left over is divisible by the divisor.

Example of CRC:

- Consider a message 110010 represented by the polynomial $M(x) = x^5 + x^4 + x$
  Consider a *generating polynomial* $G(x) = x^3 + x^2 + 1$ (1101)

This is used to generate a 3 bit CRC = C(x) to be appended to M(x).

Steps:

1. Multiply M(x) by x^3 (highest power in G(x)). i.e. Add 3 zeros. 110010000
2. Divide the result by G(x). The remainder = C(x).

1101 long division into 110010000 (with subtraction mod 2)

= 100100 remainder **100**

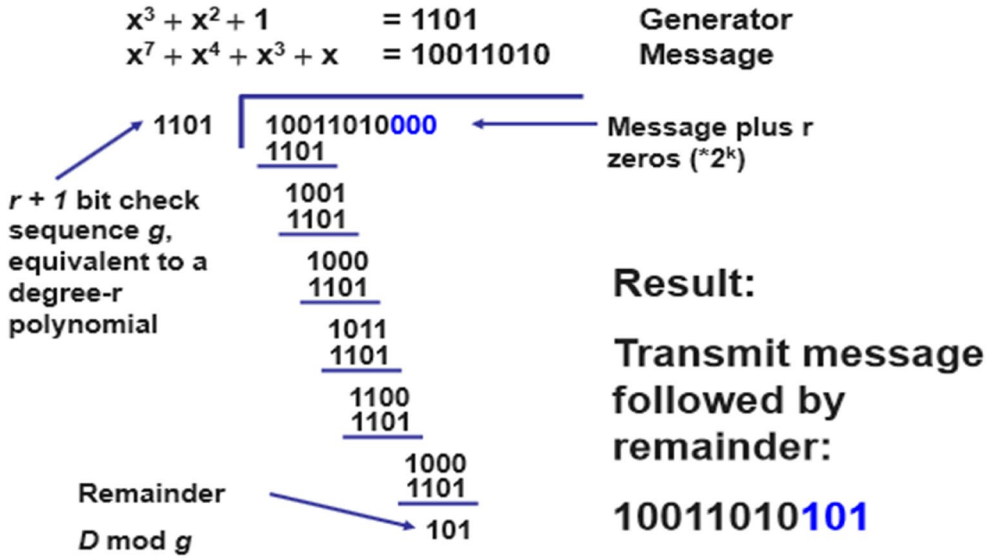
3. Transmit 110010000 + 100

To be precise, transmit: $T(x) = x^3M(x) + C(x)$

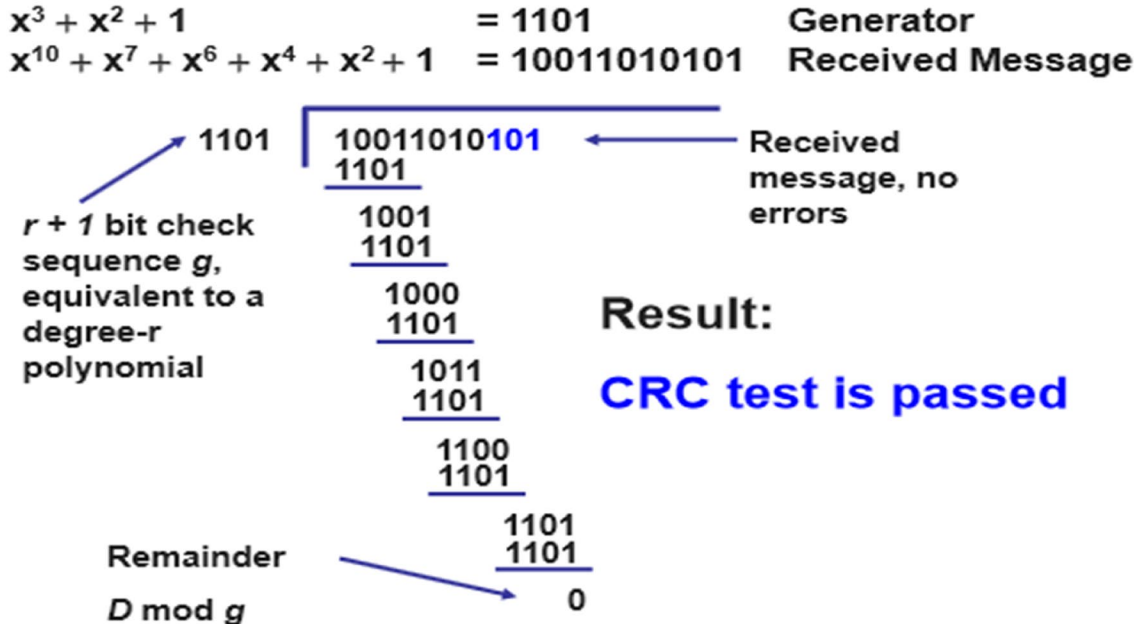
= 110010100

4. Receiver end: Receive T(x). Divide by G(x), should have remainder 0.

At sender side calculation of CRC:

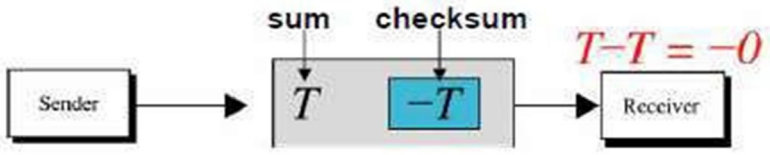


At receiver side calculation of CRC:



4. CHECKSUM:

- Checksum is the error detection scheme used in IP, TCP & UDP.
- Here, the data is divided into k segments each of n bits. In the sender's end the segments are added using 1's complement arithmetic to get the sum. The sum is complemented to get the checksum. The checksum segment is sent along with the data segments
- At the receiver's end, all received segments are added using 1's complement arithmetic to get the sum. The sum is complemented. If the result is zero, the received data is accepted; otherwise discarded
- The checksum detects all errors involving an odd number of bits. It also detects most errors involving even number of bits.



Checksum procedure at sender and receiver end:

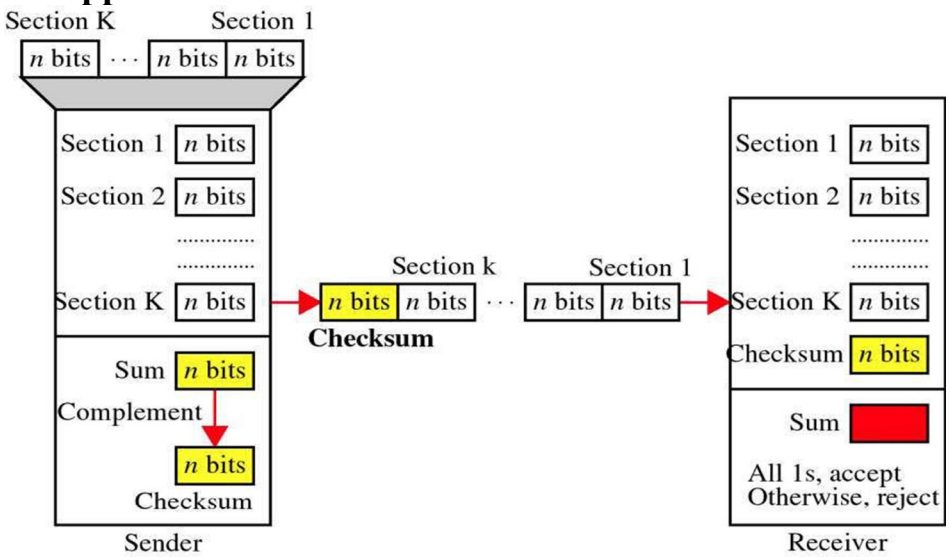
Sender:

- data is divided into k sections each n bits long
- all sections are added using 1-s complement to get the sum
- the sum is bit-wise complemented and becomes the checksum
- the checksum is sent with the data

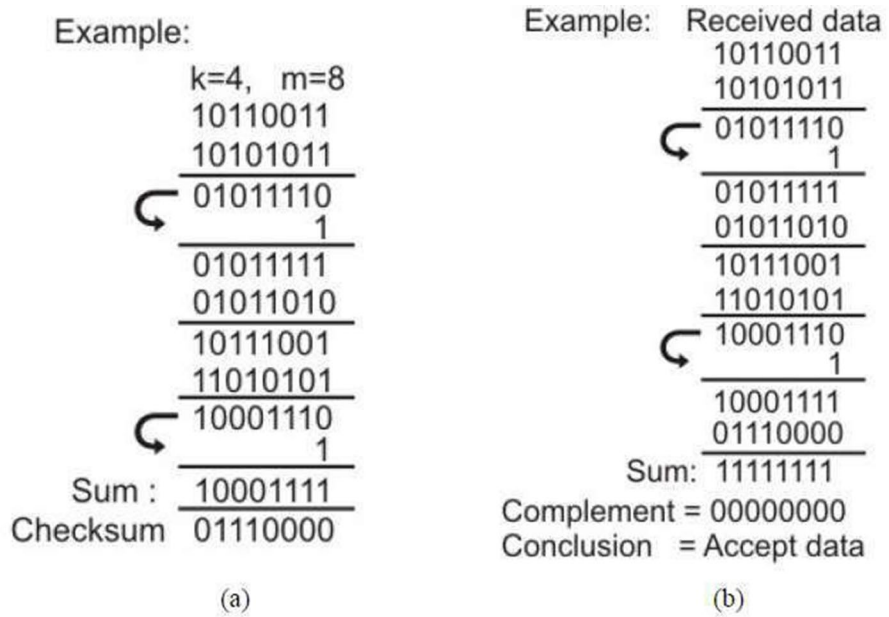
Receiver:

- data is divided into k sections each n bits long
- all sections are added using 1-s complement to get the sum
- the sum is bit-wise complemented
- if the result is zero, the data is accepted, otherwise it is rejected

Diagrammatic approach:



Example for Checksum:

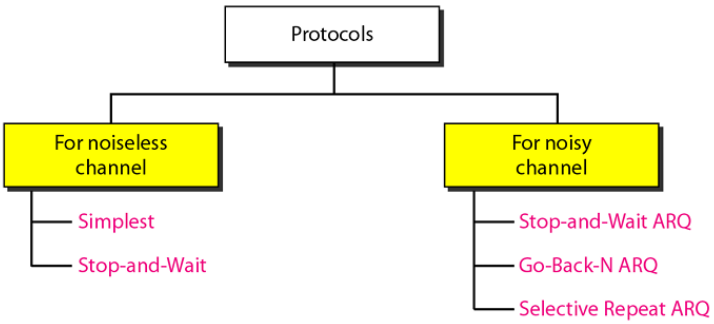


(a) Sender's end for the calculation of the checksum. (b) Receiving end for checking the checksum

Elementary Data Link Layer protocols:

Now let us see how the data link layer can combine framing, flow control, and error control to achieve the delivery of data from one node to another. The protocols are normally implemented in software by using one of the common programming languages.

We divide the discussion of protocols into those that can be used for noiseless (error-free) channels and those that can be used for noisy (error-creating) channels. The protocols in the first category cannot be used in real life, but they serve as a basis for understanding the protocols of noisy channels.



NOISELESS CHANNELS:

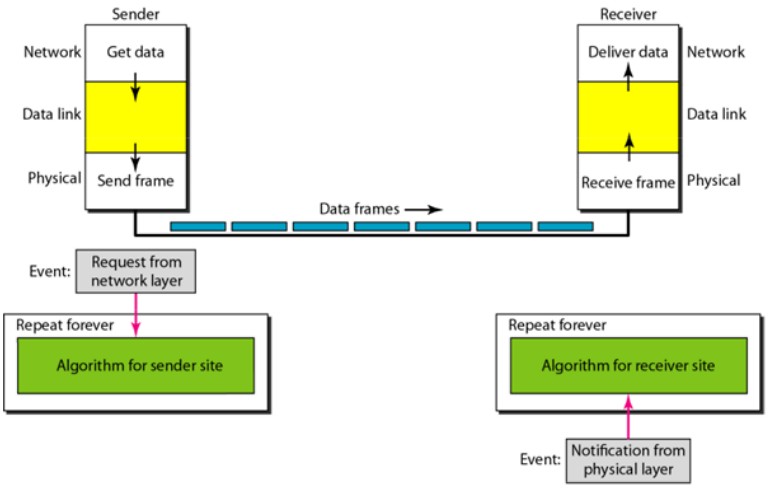
Let us first assume we have an ideal channel in which no frames are lost, duplicated, or corrupted. We introduce two protocols for this type of channel. The first is a protocol that does not use flow control; the second is the one that does. Of course, neither has error control because we have assumed that the channel is a perfect noiseless channel.

Simplest Protocol:

Our first protocol, which we call the Simplest Protocol for lack of any other name, is one that has no flow or error control. Like other protocols we will discuss in this chapter, it is a unidirectional protocol in which data frames are traveling in only one direction-from the sender to receiver. We assume that the receiver can immediately handle any frame it receives with a processing time that is small enough to be negligible. The data link layer of the receiver immediately removes the header from the frame and hands the data packet to its network layer, which can also accept the packet immediately. In other words, the receiver can never be overwhelmed with incoming frames.

Design

There is no need for flow control in this scheme. The data link layer at the sender site gets data from its network layer, makes a frame out of the data, and sends it. The data link layer at the receiver site receives a frame from its physical layer, extracts data from the frame, and delivers the data to its network layer. The data link layers of the sender and receiver provide transmission services for their network layers.



Algorithms

Sender-site algorithm for the simplest protocol

```
1  while(true)                          // Repeat forever
2  {
3    WaitForEvent();                    // Sleep until an event occurs
4    if(Event(RequestToSend))           //There is a packet to send
5    {
6       GetData();
7       MakeFrame();
8       SendFrame();                    //Send the frame
9    }
10 }
```

Analysis The algorithm has an infinite loop, which means lines 3 to 9 are repeated forever once the program starts. The algorithm is an event-driven one, which means that it *sleeps* (line 3) until an event *wakes* it *up* (line 4). This means that there may be an undefined span of time between the execution of line 3 and line 4; there is a gap between these actions. When the event, a request from the network layer, occurs, lines 6 through 8 are executed. The program then repeats The loop and again sleeps at line 3 until the next occurrence of the event. We have written pseudo code for the main process. We do not show any details for the modules Get Data, Make Frame, and Send Frame.

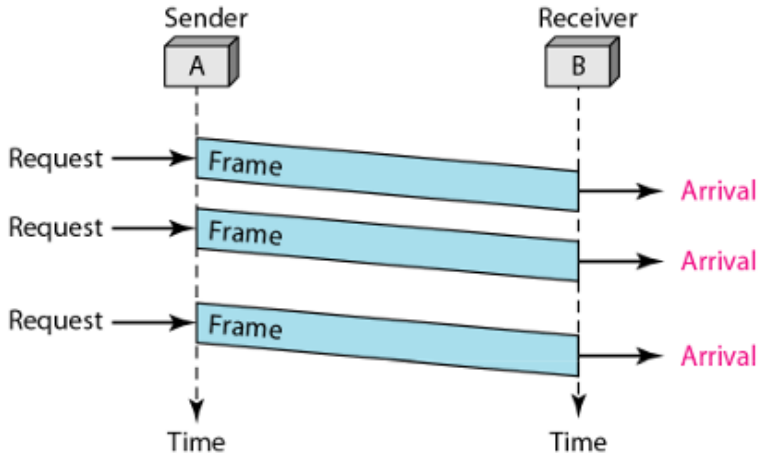
Receiver-site algorithm for the simplest protocol

```
1  while(true)                          // Repeat forever
2  {
3    WaitForEvent();                    // Sleep until an event occurs
4    if(Event(ArrivalNotification))     //Data frame arrived
5    {
6       ReceiveFrame();
7       ExtractData();
8       DeliverData();                  //Deliver data to network layer
9    }
10 }
```

Analysis This algorithm has the same format as above Algorithm except that the direction of the frames and data is upward. The event here is the arrival of a data frame. After the event occurs, the data link layer receives the frame from the physical layer using the ReceiveFrame() process, extracts the data from the frame using the ExtractData() process, and delivers the data to the network layer using the DeliverData() process. Here, we also have an event-driven algorithm because the algorithm never knows when the data frame will arrive.

Example:

Below Figure shows an example of communication using this protocol. It is very simple. The sender sends a sequence of frames without even thinking about the receiver. To send three frames, three events occur at the sender site and three events at the receiver site. Note that the data frames are shown by tilted boxes; the height of the box defines the transmission time difference between the first bit and the last bit in the frame.



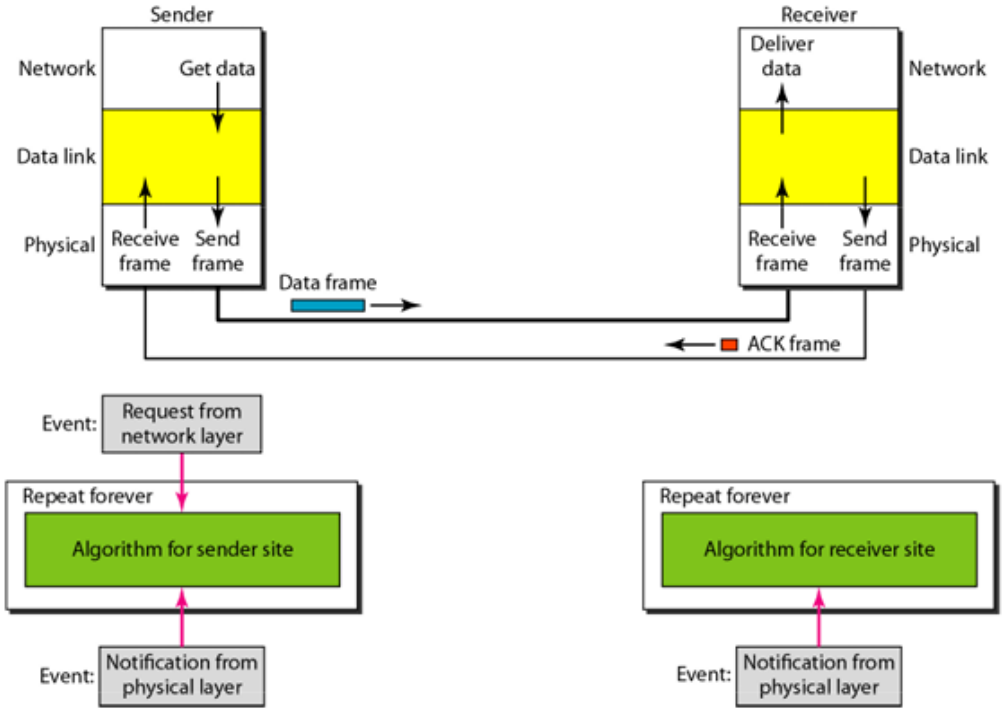
Stop-and-Wait Protocol:

If data frames arrive at the receiver site faster than they can be processed, the frames must be stored until their use. Normally, the receiver does not have enough storage space, especially if it is receiving data from many sources. This may result in either the discarding of frames or denial of service. To prevent the receiver from becoming overwhelmed with frames, we somehow need to tell the sender to slow down. There must be feedback from the receiver to the sender.

The protocol we discuss now is called the Stop-and-Wait Protocol because the sender sends one frame, stops until it receives confirmation from the receiver (okay to go ahead), and then sends the next frame. We still have unidirectional communication for data frames, but auxiliary ACK frames (simple tokens of acknowledgment) travel from the other direction. We add flow control to our previous protocol.

Design

We can see the traffic on the forward channel (from sender to receiver) and the reverse channel. At any time, there is either one data frame on the forward channel or one ACK frame on the reverse channel. We therefore need a half-duplex link.



Algorithms

Sender-site algorithm for Stop-and- Wait Protocol

```
1  while(true)                          //Repeat forever
2  canSend = true                       //Allow the first frame to go
3  {
4   WaitForEvent();                     // Sleep until an event occurs
5   if(Event(RequestToSend) AND canSend)
6   {
7      GetData();
8      MakeFrame();
9      SendFrame();                     //Send the data frame
10     canSend = false;                 //Cannot send until ACK arrives
11  }
12  WaitForEvent();                     // Sleep until an event occurs
13  if(Event(ArrivalNotification)       // An ACK has arrived
14   {
15     ReceiveFrame();                  //Receive the ACK frame
16     canSend = true;
17   }
18 }
```

Analysis Here two events can occur: a request from the network layer or an arrival notification from the physical layer. The responses to these events must alternate. In other words, after a frame is sent, the algorithm must ignore another network layer request until that frame is acknowledged. We know that two arrival events cannot happen one after another because the channel is error-free and does not duplicate the frames. The requests from the network layer, however, may happen one after another without an arrival event in between. We need somehow to prevent the immediate sending of the data frame. Although there are several methods, we have used a simple *canSend* variable that can either be true or false. When a frame is sent, the variable is set to false to indicate that a new network request cannot be sent until *canSend* is true. When an ACK is received, canSend is set to true to allow the sending of the next frame.

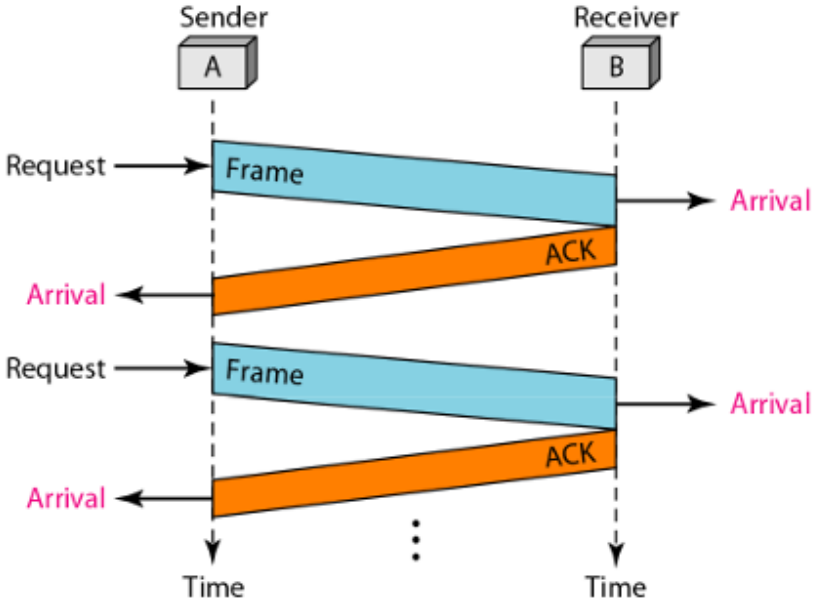
Receiver-site algorithm for Stop-and-Wait Protocol

```
1  while(true)                          //Repeat forever
2  {
3    WaitForEvent();                    // Sleep until an event occurs
4    if(Event(ArrivalNotification))     //Data frame arrives
5    {
6       ReceiveFrame();
7       ExtractData();
8       Deliver(data);                  //Deliver data to network layer
9       SendFrame();                    //Send an ACK frame
10   }
11 }
```

Analysis This is very similar to above Algorithm with one exception. After the data frame arrives, the receiver sends an ACK frame (line 9) to acknowledge the receipt and allow the sender to send the next frame.

Example:

Below Figure shows an example of communication using this protocol. It is still very simple. The sender sends one frame and waits for feedback from the receiver. When the ACK arrives, the sender sends the next frame. Note that sending two frames in the protocol involves the sender in four events and the receiver in two events.



NOISY CHANNELS:

Although the Stop-and-Wait Protocol gives us an idea of how to add flow control to its predecessor, noiseless channels are nonexistent. We can ignore the error (as we sometimes do), or we need to add error control to our protocols. We discuss three protocols in this section that use error control.

Stop-and-Wait Automatic Repeat Request:

Our first protocol, called the Stop-and-Wait Automatic Repeat Request (Stop-and Wait ARQ), adds a simple error control mechanism to the Stop-and-Wait Protocol. Let us see how this protocol detects and corrects errors.

To detect and correct corrupted frames, we need to add redundancy bits to our data frame. When the frame arrives at the receiver site, it is checked and if it is corrupted, it is silently discarded. The detection of errors in this protocol is manifested by the silence of the receiver.

Lost frames are more difficult to handle than corrupted ones. In our previous protocols, there was no way to identify a frame. The received frame could be the correct one, or a duplicate, or a frame out of order. The solution is to number the frames. When the receiver receives a data frame that is out of order, this means that frames were either lost or duplicated.

The lost frames need to be resent in this protocol. If the receiver does not respond when there is an error, how can the sender know which frame to resend? To remedy this problem, the sender keeps a copy of the sent frame. At the same time, it starts a timer. If the timer expires and there is no ACK for the sent frame, the frame is resent, the copy is held, and the timer is restarted. Since the protocol uses the stop-and-wait mechanism, there is only one specific frame that needs an ACK even though several copies of the same frame can be in the network.

Sequence Numbers

As we discussed, the protocol specifies that frames need to be numbered. This is done by using sequence numbers. A field is added to the data frame to hold the sequence number of that frame.

One important consideration is the range of the sequence numbers. Since we want to minimize the frame size, we look for the smallest range that provides unambiguous communication. The sequence numbers of course can wrap around. For example, if we decide that the field is *m* bits long, the sequence numbers start from 0, go to 2^m - 1, and then are repeated.

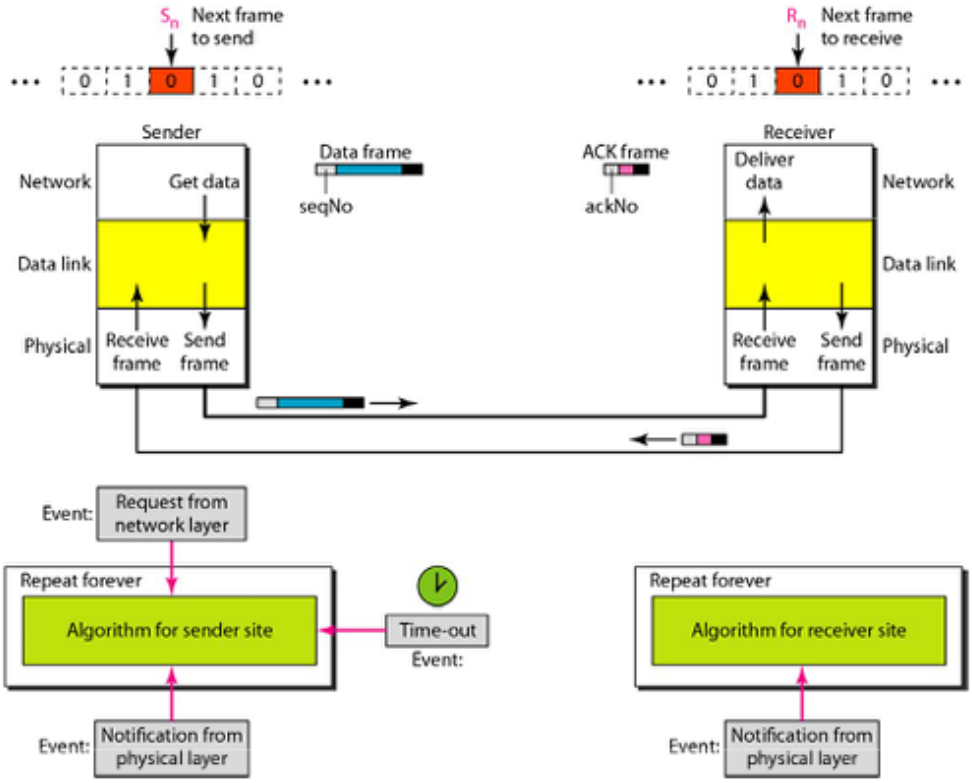
Acknowledgment Numbers

Since the sequence numbers must be suitable for both data frames and ACK frames, we use this convention: The acknowledgment numbers always announce the sequence number of the next frame expected by the receiver. For example, if frame 0 has arrived safe and sound, the receiver sends an ACK frame with acknowledgment 1 (meaning frame 1 is expected next). If frame 1 has arrived safe and sound, the receiver sends an ACK frame with acknowledgment 0 (meaning frame 0 is expected).

Design

Below Figure shows the design of the Stop-and-WaitARQ Protocol. The sending device keeps a copy of the last frame transmitted until it receives an acknowledgment for that frame. A data frames uses a seqNo (sequence number); an ACK frame uses an ackNo (acknowledgment number). The sender has a control variable, which we call *Sn* (sender, next frame to send), that holds the sequence number for the next frame to be sent (0 or 1).

The receiver has a control variable, which we call *Rn* (receiver, next frame expected), that holds the number of the next frame expected. When a frame is sent, the value of *Sn* is incremented (modulo-2), which means if it is 0, it becomes 1 and vice versa. When a frame is received, the value of *Rn* is incremented (modulo-2), which means if it is 0, it becomes 1 and vice versa. Three events can happen at the sender site; one event can happen at the receiver site. Variable *Sn* points to the slot that matches the sequence number of the frame that has been sent, but not acknowledged; *Rn* points to the slot that matches the sequence number of the expected frame.



Algorithms

Sender-site algorithm for Stop-and- Wait ARQ

```
1  Sn = 0;                          // Frame 0 should be sent first
2  canSend = true;                  // Allow the first request to go
3  while(true)                      // Repeat forever
4  {
5    WaitForEvent();                // Sleep until an event occurs
6    if(Event(RequestToSend) AND canSend)
7    {
8       GetData();
9       MakeFrame(Sn);                    //The seqNo is Sn
10      StoreFrame(Sn);                   //Keep copy
11      SendFrame(Sn);
12      StartTimer();
13      Sn = Sn + 1;
14      canSend = false;
15   }
16   WaitForEvent();                       // Sleep
17      if(Event(ArrivalNotification)      // An ACK has arrived
18      {
19        ReceiveFrame(ackNo);             //Receive the ACK frame
20        if(not corrupted AND ackNo == Sn) //Valid ACK
21          {
22            Stoptimer();
23            PurgeFrame(Sn-1);            //Copy is not needed
24            canSend = true;
25          }
26       }
27
28      if(Event(TimeOut)                  // The timer expired
29      {
30       StartTimer();
31       ResendFrame(Sn-1);                //Resend a copy check
32      }
33 }
```

Analysis We first notice the presence of *Sn'* the sequence number of the next frame to be sent. This variable is initialized once (line 1), but it is incremented every time a frame is sent (line 13) in preparation for the next frame. However, since this is modulo-2 arithmetic, the sequence numbers are 0, 1, 0, 1, and so on. Note that the processes in the first event (SendFrame, Store Frame, and Purge Frame) use an *Sn* defining the frame sent out. We need at least one buffer to hold this frame until we are sure that it is received safe and sound. Line 10 shows that before the frame is sent, it is stored. The copy is used for resending a corrupt or lost frame. We are still using the canSend variable to prevent the network layer from making a request before the previous frame is received safe and sound. If the frame is not corrupted and the ackNo of theACK frame matches the sequence number of the next frame to send, we stop the timer and purge the copy of the data frame we saved. Otherwise, we just ignore this event and wait for the next event to happen. After each frame is sent, a timer is started. When the timer expires (line 28), the frame is resent and the timer is restarted.

Receiver-site algorithm for Stop-and-WaitARQ Protocol

```
1  Rn = 0;                        // Frame 0 expected to arrive first
2  while(true)
3  {
4    WaitForEvent();              // Sleep until an event occurs
5    if(Event(ArrivalNotification))  //Data frame arrives
6    {
7       ReceiveFrame();
8       if(corrupted(frame));
9          sleep();
10      if(seqNo == Rn)              //Valid data frame
11      {
12       ExtractData();
13        DeliverData();             //Deliver data
14        Rn = Rn + 1;
15      }
16       SendFrame(Rn);              //Send an ACK
17   }
18 }
```

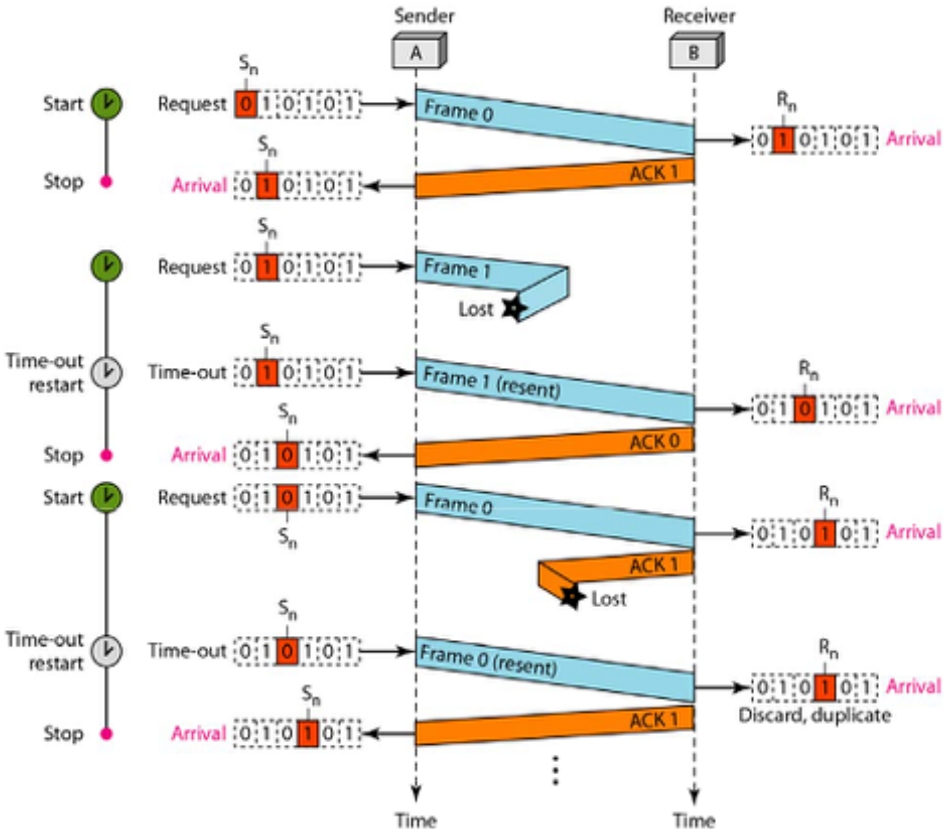
Analysis This is noticeably different from Algorithm 11.4. First, all arrived data frames that are corrupted are ignored. If the seqNo of the frame is the one that is expected *(Rn),* the frame is accepted, the data are delivered to the network layer, and the value of *Rn* is incremented. However, there is one subtle point here. Even if the sequence number of the data frame does not match the next frame expected, an ACK is sent to the sender. This ACK, however, just reconfirms the previous ACK instead of confirming the frame received. This is done because the receiver assumes that the previous ACK might have been lost; the receiver is sending a duplicate frame. The resent ACK may solve the problem before the time-out does it.

Efficiency

The Stop-and-Wait ARQ discussed in the previous section is very inefficient if our channel is *thick* and *long.* By *thick,* we mean that our channel has a large bandwidth; by *long,* we mean the round-trip delay is long. The product of these two is called the bandwidth delay product, as we discussed in Chapter 3. We can think of the channel as a pipe. The bandwidth-delay product then is the volume of the pipe in bits. The pipe is always there. If we do not use it, we are inefficient. The bandwidth-delay product is a measure of the number of bits we can send out of our system while waiting for news from the receiver.

Example

Below Figure shows an example of Stop-and-Wait ARQ. Frame **a** is sent and acknowledged. Frame 1 is lost and resent after the time-out. The resent frame 1 is acknowledged and the timer stops. Frame **a** is sent and acknowledged, but the acknowledgment is lost. The sender has no idea if the frame or the acknowledgment is lost, so after the time-out, it resends frame 0, which is acknowledged.



Pipelining

In networking and in other areas, a task is often begun before the previous task has ended. This is known as pipelining. There is no pipelining in Stop-and-Wait ARQ because we need to wait for a frame to reach the destination and be acknowledged before the next frame can be sent. However, pipelining does apply to our next two protocols because several frames can be sent before we receive news about the previous frames. Pipelining improves the efficiency of the transmission if the number of bits in transition is large with respect to the bandwidth-delay product.

Go-Back-N Automatic Repeat Request:

To improve the efficiency of transmission (filling the pipe), multiple frames must be in transition while waiting for acknowledgment. In other words, we need to let more than one frame be outstanding to keep the channel busy while the sender is waiting for acknowledgment.

Go-Back-N Automatic Repeat Request protocol we can send several frames before receiving acknowledgments; we keep a copy of these frames until the acknowledgments arrive.

Sequence Numbers

Frames from a sending station are numbered sequentially. However, because we need to include the sequence number of each frame in the header, we need to set a limit. If the header of the frame allows *m* bits for the sequence number, the sequence numbers range from 0 to *2m* - 1. For example, if *m* is 4, the only sequence numbers are 0 through 15 inclusive. However, we can repeat the sequence. So the sequence numbers are

0, 1,2,3,4,5,6, 7,8,9, 10, 11, 12, 13, 14, 15,0, 1,2,3,4,5,6,7,8,9,10, 11, ...

In other words, the sequence numbers are modulo-2^m

.

Sliding Window

In this protocol the sliding window is an abstract concept that defines the range of sequence numbers that is the concern of the sender and receiver. In other words, the sender and receiver need to deal with only part of the possible sequence numbers. The range which is the concern of the sender is called the send sliding window; the range that is the concern of the receiver is called the receive sliding window.

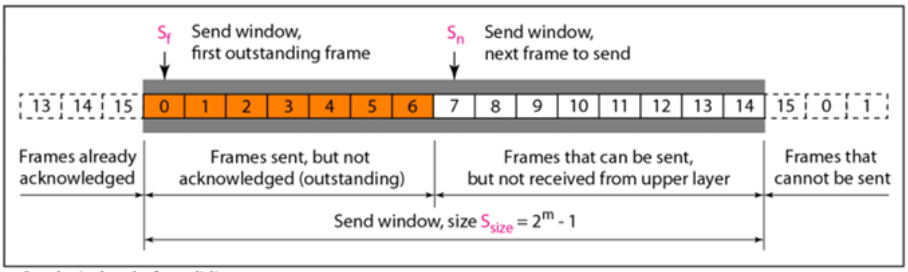
The send window is an imaginary box covering the sequence numbers of the data frames which can be in transmit. In each window position, some of these sequence numbers define the frames that have been sent;

others define those that can be sent. The maximum size of the window is 2^m - 1 we let the size be fixed and set to the maximum value, below figure **a** shows a sliding window of size 15 *(m* =4).

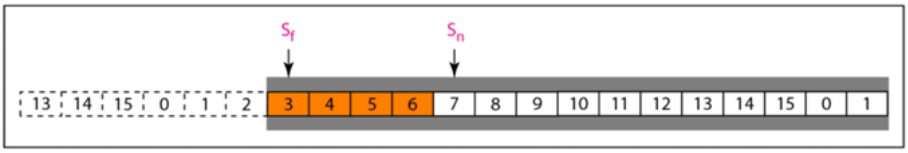
The window at any time divides the possible sequence numbers into four regions. The first region, from the far left to the left wall of the window, defines the sequence numbers belonging to frames that are already acknowledged. The sender does not worry about these frames and keeps no copies of them. The second region, colored in Figure **a**, defines the range of sequence numbers belonging to the frames that are sent and have an unknown status. The sender needs to wait to find out if these frames have been received or were lost. We call these outstanding frames. The third range, white in the figure, defines the range of sequence numbers for frames that can be sent; however, the corresponding data packets have not yet been received from the network layer. Finally, the fourth region defines sequence numbers that cannot be used until the window slides, as we see next.

Below Figure **b** shows how a send window can slide one or more slots to the right when an acknowledgment arrives from the other end. As we will see shortly, the acknowledgments in this protocol are cumulative, meaning that more than one frame can be acknowledged by an ACK frame. In Figure **b**, frames 0, I, and 2 are acknowledged, so the window has slid to the right three slots. Note that the value of *Sf* is 3 because frame 3 is now the first outstanding frame.

The window itself is an abstraction; three variables define its size and location at any time. We call these variables *Sf(send* window, the first outstanding frame), *Sn* (send window, the next frame to be sent), and Ssize (send window, size). The variable *Sf* defines the sequence number of the first (oldest) outstanding frame. The variable *Sn* holds the sequence number that will be assigned to the next frame to be sent. Finally, the variable Ssize defines the size of the window, which is fixed in our protocol.



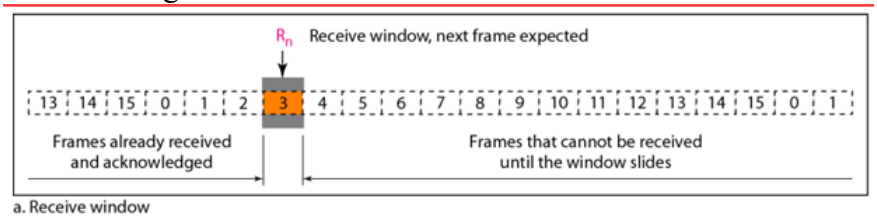
a. Send window before sliding



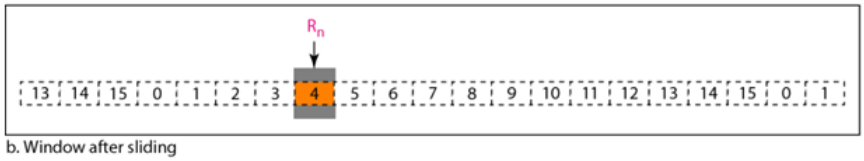
b. Send window after sliding

The receive window makes sure that the correct data frames are received and that the correct acknowledgments are sent. The size of the receive window is always 1. The receiver is always looking for the arrival of a specific frame. Any frame arriving out of order is discarded and needs to be resent. Below figure shows the receive window.

We need only one variable *Rn* (receive window, next frame expected) to define this abstraction. The sequence numbers to the left of the window belong to the frames already received and acknowledged; the sequence numbers to the right of this window define the frames that cannot be received. Any received frame with a sequence number in these two regions is discarded. Only a frame with a sequence number matching the value of *Rn* is accepted and acknowledged.



a. Receive window



b. Window after sliding

Timers

Although there can be a timer for each frame that is sent, in our protocol we use only one. The reason is that the timer for the first outstanding frame always expires first; we send all outstanding frames when this timer expires.

Acknowledgment

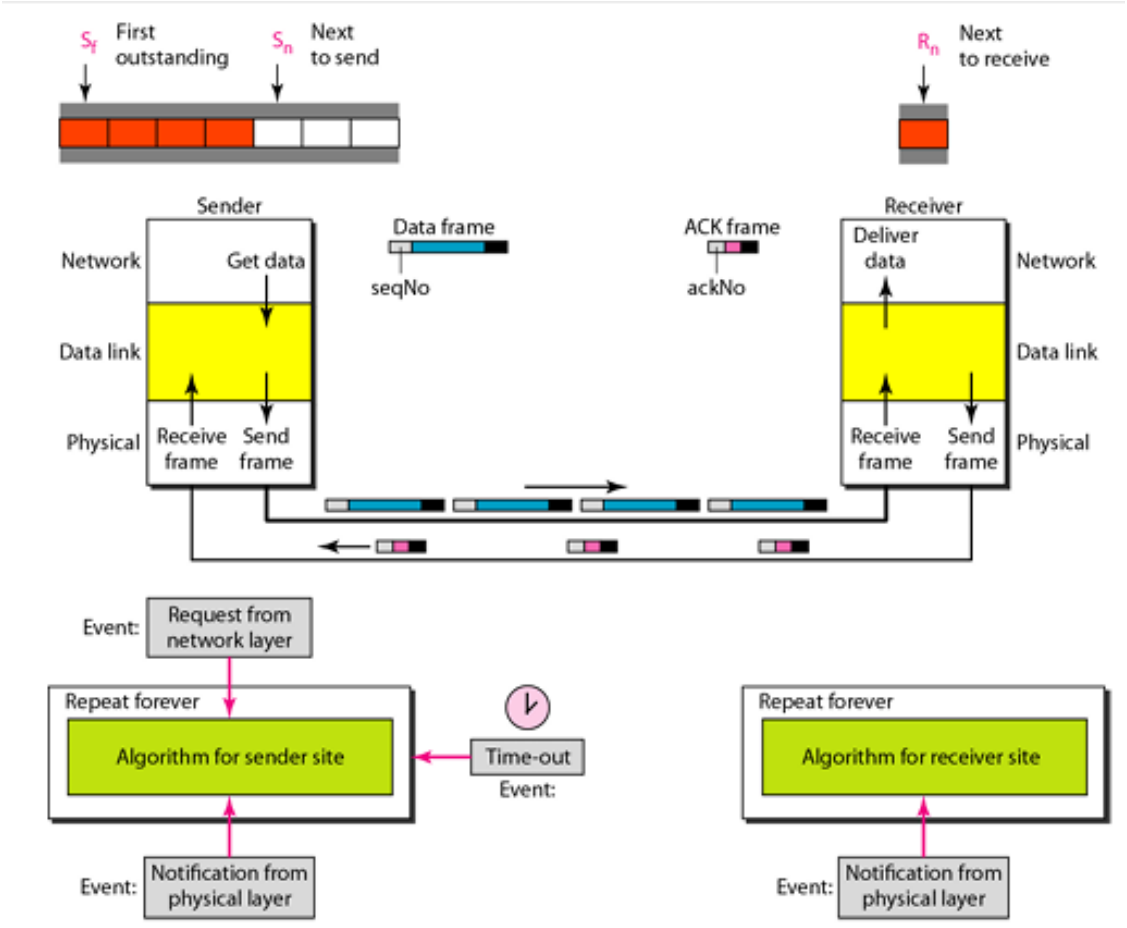
The receiver sends a positive acknowledgment if a frame has arrived safe and sound and in order. If a frame is damaged or is received out of order, the receiver is silent and will discard all subsequent frames until it receives the one it is expecting. The silence of the receiver causes the timer of the unacknowledged frame at the sender site to expire. This, in turn, causes the sender to go back and resend all frames, beginning with the one with the expired timer. The receiver does not have to acknowledge each frame received. It can send one cumulative acknowledgment for several frames.

Resending a Frame

When the timer expires, the sender resends all outstanding frames. For example, suppose the sender has already sent frame 6, but the timer for frame 3 expires. This means that frame 3 has not been acknowledged; the sender goes back and sends frames 3, 4,5, and 6 again. That is why the protocol is called *Go-Back-N* ARQ.

Design

Below Figure shows the design for this protocol. As we can see, multiple frames can be in transit in the forward direction, and multiple acknowledgments in the reverse direction. The idea is similar to Stop-and-Wait ARQ; the difference is that the send window allows us to have as many frames in transition as there are slots in the send window.



Algorithms
Go-Back-N sender algorithm

```
1  Sw = 2^m - 1;
2  Sf = 0;
3  Sn = 0;
4
5  while (true)                          //Repeat forever
6  {
7   WaitForEvent();
8    if(Event(RequestToSend))            //A packet to send
9    {
10      if(Sn-Sf >= Sw)                  //If window is full
11            Sleep();
12      GetData();
13      MakeFrame(Sn);
14      StoreFrame(Sn);
15      SendFrame(Sn);
16      Sn = Sn + 1;
17      if(timer not running)
18           StartTimer();
19   }
20
21   if(Event(ArrivalNotification)) //ACK arrives
22   {
23      Receive(ACK);
24      if(corrupted(ACK))
25           Sleep();
26      if((ackNo>Sf)&&(ackNo<=Sn))  //If a valid ACK
27      While(Sf <= ackNo)
28       {
29        PurgeFrame(Sf);
30        Sf = Sf + 1;
31       }
32       StopTimer();
33   }
34
35   if(Event(TimeOut))                  //The timer expires
36   {
37    StartTimer();
38    Temp = Sf;
39    while(Temp < Sn);
40     {
41      SendFrame(Sf);
42      Sf = Sf + 1;
43     }
44   }
45 }
```

Analysis this algorithm first initializes three variables. Unlike Stop-and-Wait ARQ, this protocol allows several requests from the network layer without the need for other events to occur; we just need to be sure that the window is not full (line 12). In our approach, if the window is full, the request is just ignored and the network layer needs to try again. Some implementations use other methods such as enabling or disabling the network layer. The handling of the arrival event is more complex than in the previous protocol. If we receive a corrupted ACK, we ignore it.

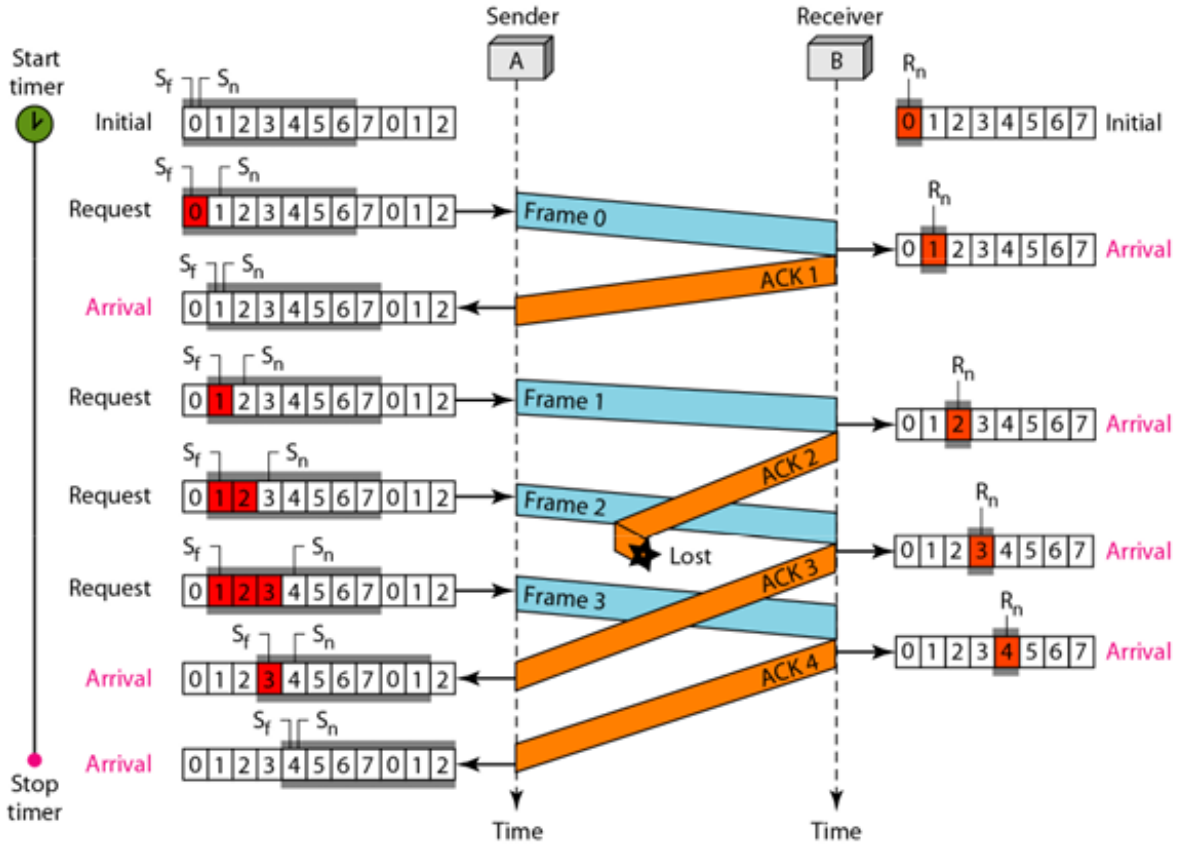
```
1  Rn = 0;
2
3  while (true)                          //Repeat forever
4  {
5    WaitForEvent();
6
7    if(Event(ArrivalNotification)) /Data frame arrives
8    {
9       Receive(Frame);
10      if(corrupted(Frame))
11           Sleep();
12      if(seqNo == Rn)                  //If expected frame
13      {
14        DeliverData();                 //Deliver data
15        Rn = Rn + 1;                   //Slide window
16        SendACK(Rn);
17      }
18   }
19 }
```

Analysis This algorithm is simple. We ignore a corrupt or out-of-order frame. If a frame arrives with an expected sequence number, we deliver the data, update the value of *Rn,* and send an ACK with the ackNa showing the next frame expected.

Example

Below Figure shows an example of Go-Back-*N.* This is an example of a case where the forward channel is reliable, but the reverse is not. No data frames are lost, but some ACKs are delayed and one is lost. The example also shows how cumulative acknowledgments can help if acknowledgments are delayed or lost.

After initialization, there are seven sender events. Request events are triggered by data from the network layer; arrival events are triggered by acknowledgments from the physical layer. There is no time-out event here because all outstanding frames are acknowledged before the timer expires. Note that although ACK 2 is lost, ACK 3 serves as both ACK 2 and ACK3.

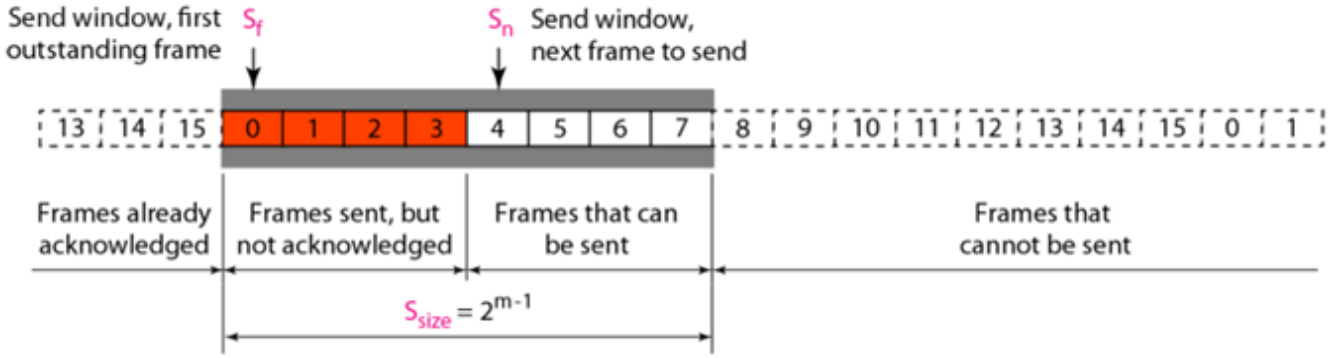


Selective Repeat Automatic Repeat Request:

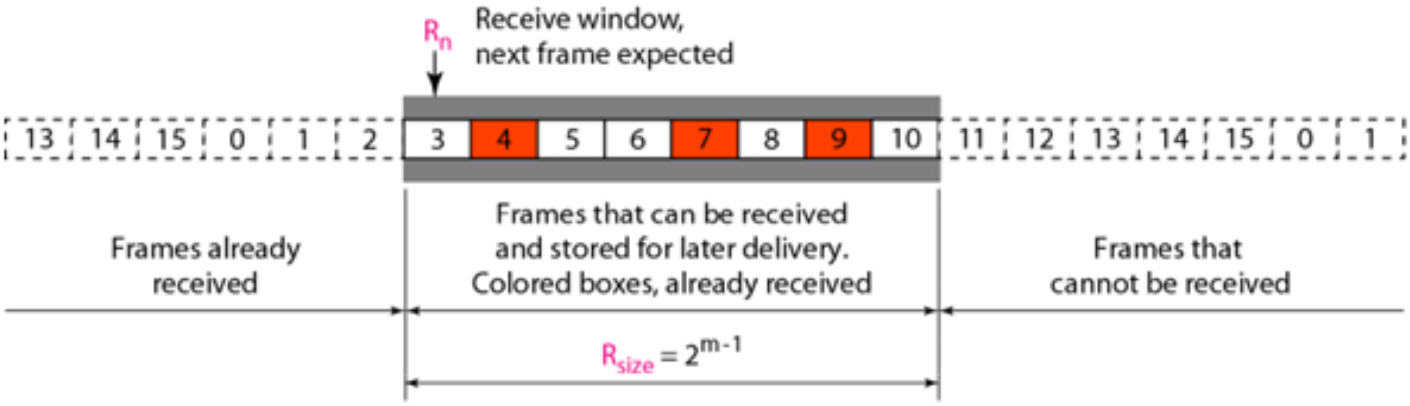
Go-Back-N ARQ simplifies the process at the receiver site. The receiver keeps track of only one variable, and there is no need to buffer out-of-order frames; they are simply discarded. However, this protocol is very inefficient for a noisy link. In a noisy link a frame has a higher probability of damage, which means the resending of multiple frames. This resending uses up the bandwidth and slows down the transmission. For noisy links, there is another mechanism that does not resend *N* frames when just one frame is damaged; only the damaged frame is resent. This mechanism is called Selective RepeatARQ.

Windows

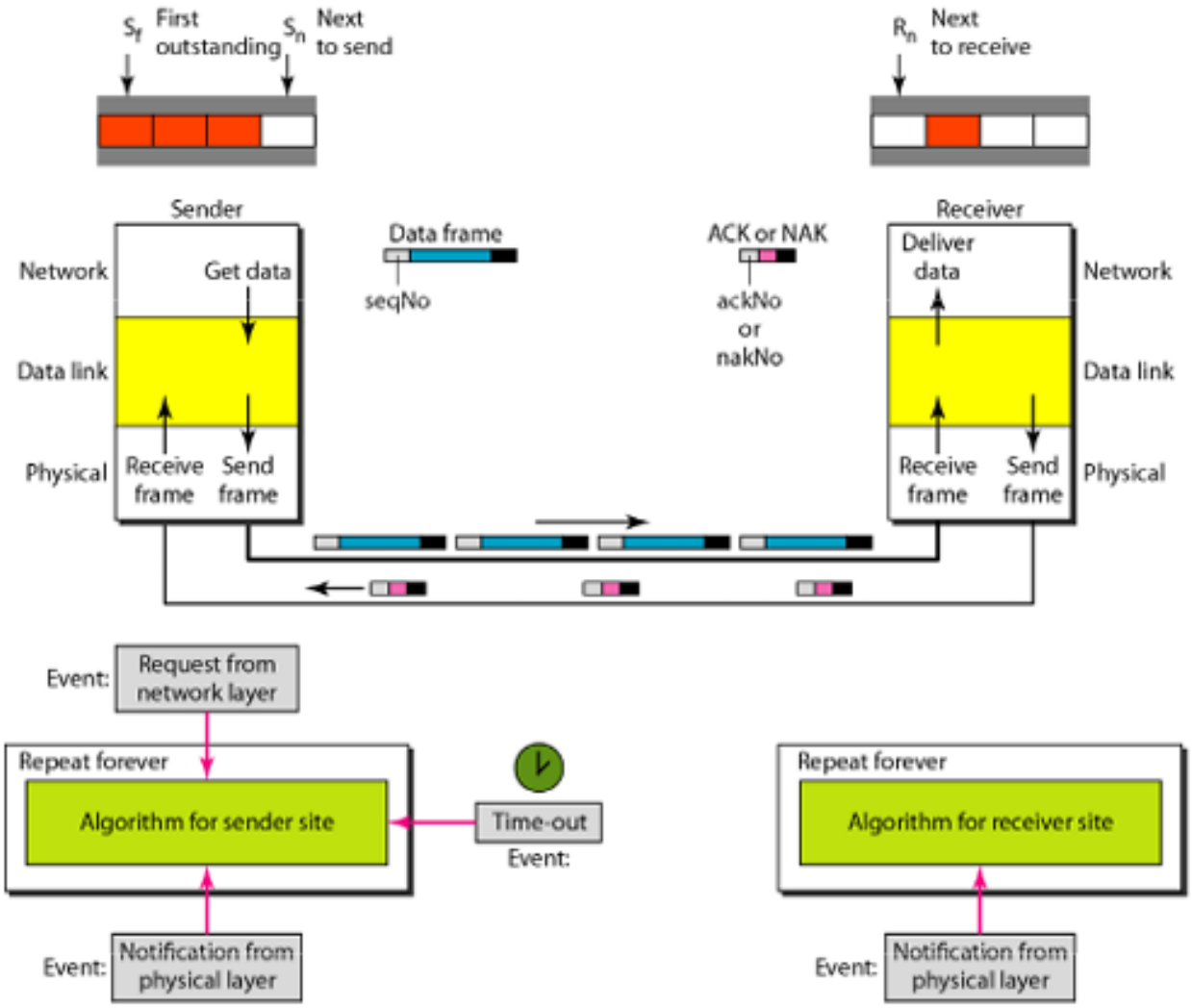
The Selective Repeat Protocol also uses two windows: a send window and a receive window. However, there are differences between the windows in this protocol and the ones in Go-Back-N. First, the size of the send window is much smaller; it is $2^m - 1$. The reason for this will be discussed later. Second, the receive window is the same size as the send window. The send window maximum size can be $2^m - 1$. For example, if $m = 4$, the sequence numbers go from 0 to 15, but the size of the window is just 8



The receive window in Selective Repeat is totally different from the one in GoBack-N. First, the size of the receive window is the same as the size of the send window $(2^m - 1)$. The Selective Repeat Protocol allows as many frames as the size of the receive window to arrive out of order and be kept until there is a set of in-order frames to be delivered to the network layer. Because the sizes of the send window and receive window are the same.



Design



Algorithms

Sender-side Selective Repeat algorithm

```
1  Sw = 2^(m-1) ;
2  Sf = 0;
3  Sn = 0;
4
5  while (true)                          //Repeat forever
6  {
7    WaitForEvent();
8    if(Event(RequestToSend))            //There is a packet to send
9    {
10     if(Sn-Sf >= Sw)                   //If window is full
11           Sleep();
12     GetData();
13     MakeFrame(Sn);
14     StoreFrame(Sn);
15     SendFrame(Sn);
16     Sn = Sn + 1;
17     StartTimer(Sn);
18   }
19
```

```
20   if(Event(ArrivalNotification)) //ACK arrives
21   {
22      Receive(frame);              //Receive ACK or NAK
23      if(corrupted(frame))
24           Sleep();
25      if (FrameType == NAK)
26         if (nakNo between Sf and Sn)
27         {
28          resend(nakNo);
29          StartTimer(nakNo);
30         }
31      if (FrameType == ACK)
32         if (ackNo between Sf and Sn)
33         {
34           while(sf < ackNo)
35           {
36            Purge(sf);
37            StopTimer(sf);
38            Sf = Sf + 1;
39           }
40         }
41   }
42
43   if(Event(TimeOut(t)))          //The timer expires
44   {
45    StartTimer(t);
46    SendFrame(t);
47   }
48 }
```

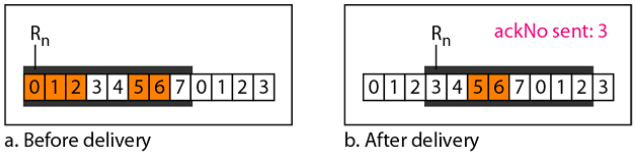
Analysis The handling of the request event is similar to that of the previous protocol except that one timer is started for each frame sent. The arrival event is more complicated here. An ACK or a NAK frame may arrive. If a valid NAK frame arrives, we just resend the corresponding frame. If a valid ACK arrives, we use a loop to purge the buffers, stop the corresponding timer. and move the left wall of the window. The time-out event is simpler here; only the frame which times out is resent.

Receiver-site Selective Repeat algorithm

```
1  Rn = 0;
2  NakSent = false;
3  AckNeeded = false;
4  Repeat(for all slots)
5      Marked(slot) = false;
6
7  while (true)                              //Repeat forever
8  {
9    WaitForEvent();
10
11   if(Event(ArrivalNotification))          /Data frame arrives
12   {
13      Receive(Frame);
14      if(corrupted(Frame))&& (NOT NakSent)
15      {
16       SendNAK(Rn);
17       NakSent = true;
18       Sleep();
19      }
20      if(seqNo <> Rn)&& (NOT NakSent)
21      {
22       SendNAK(Rn);
```

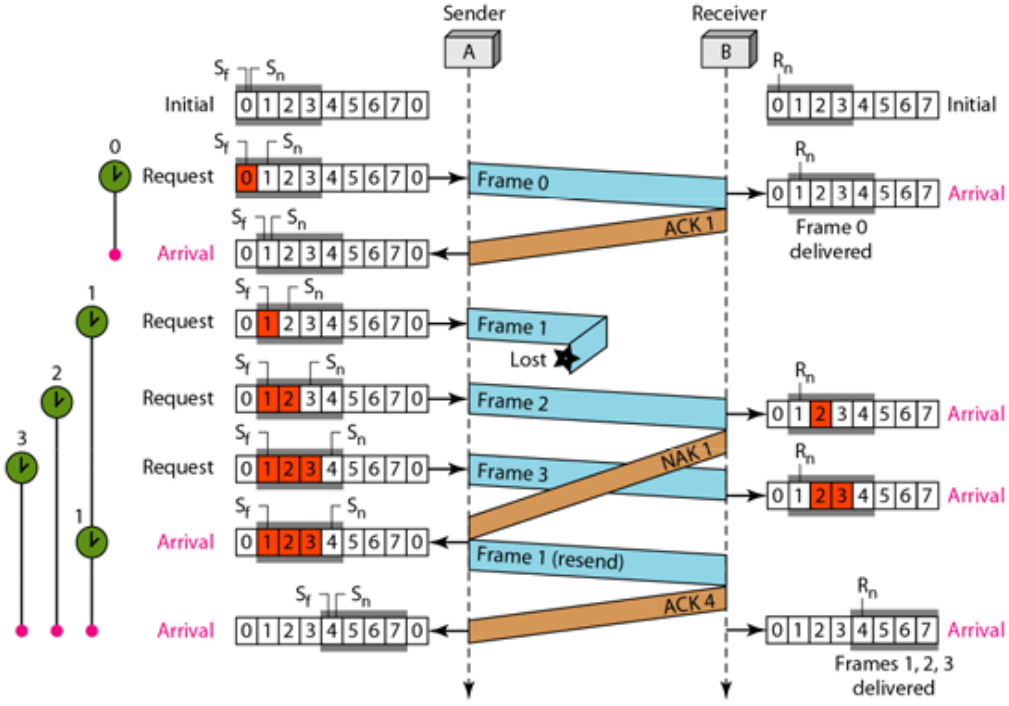
```
23        NakSent = true;
24        if ((seqNo in window)&&(!Marked(seqNo))
25        {
26         StoreFrame(seqNo)
27         Marked(seqNo)= true;
28         while(Marked(Rn))
29         {
30          DeliverData(Rn);
31          Purge(Rn);
32          Rn = Rn + 1;
33          AckNeeded = true;
34         }
35          if(AckNeeded);
36          {
37          SendAck(Rn);
38          AckNeeded = false;
39          NakSent = false;
40          }
41        }
42       }
43   }
44 }
```

Analysis Here we need more initialization. In order not to overwhelm the other side with NAKs, we use a variable called NakSent. To know when we need to send an ACK, we use a variable called AckNeeded. Both of these are initialized to false. We also use a set of variables to mark the slots in the receive window once the corresponding frame has arrived and is stored. If we receive a corrupted frame and a NAK has not yet been sent, we send a NAK to tell the other site that we have not received the frame we expected. If the frame is not corrupted and the sequence number is in the window, we store the frame and mark the slot. If contiguous frames, starting from *Rn* have been marked, we deliver their data to the network layer and slide the window. Below Figure shows this situation.



Example

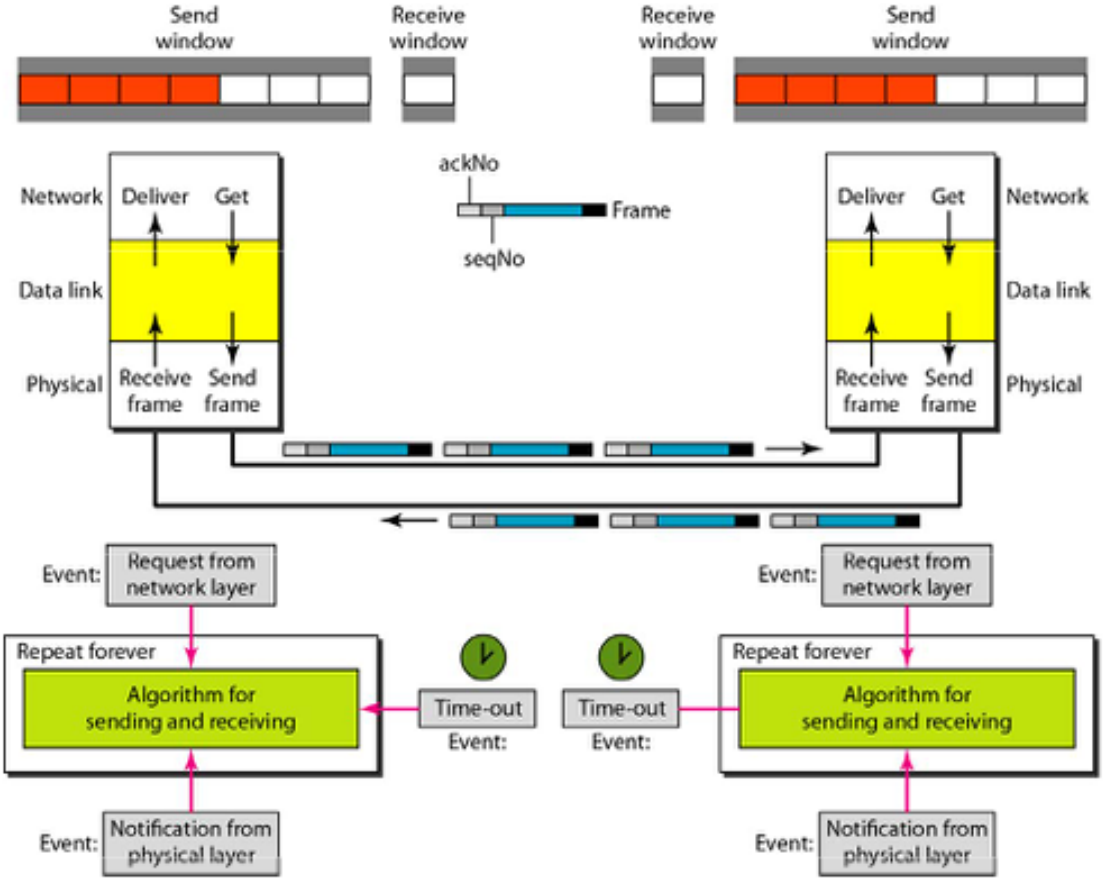
This example is similar to go back N Example in which frame 1 is lost. We show how Selective Repeat behaves in this case. Below Figure shows the situation.



Piggybacking:

The three protocols we discussed in this section are all unidirectional: data frames flow in only one direction although control information such as ACK and NAK frames can travel in the other direction. In real life, data frames are normally flowing in both directions: from node A to node B and from node B to node A. This means that the control information also needs to flow in both directions. A technique called **piggybacking** is used to improve the efficiency of the bidirectional protocols. When a frame is carrying data from A to B, it can also carry control information about arrived (or lost) frames from B; when a frame is carrying data from B to A, it can also carry control information about the arrived (or lost) frames from A.

We show the design for a Go-Back-N ARQ using piggybacking in below Figure. Note that each node now has two windows: one send window and one receive window. Both also need to use a timer. Both are involved in three types of events: request, arrival, and time-out. However, the arrival event here is complicated; when a frame arrives, the site needs to handle control information as well as the frame itself. Both of these concerns must be taken care of in one event, the arrival event. The request event uses only the send window at each site; the arrival event needs to use both windows.



HDLC:

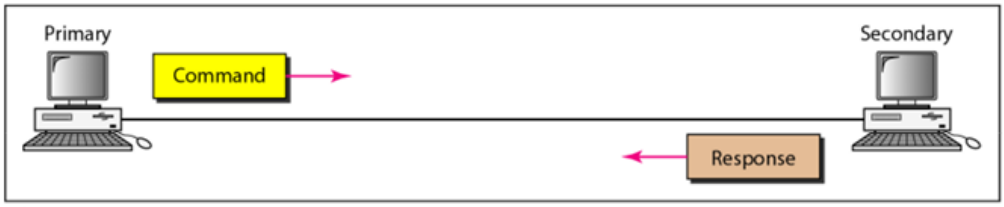
High-level Data Link Control (HDLC) is a bit-oriented protocol for communication over point-to-point and multipoint links. It implements the ARQ mechanisms

Configurations and Transfer Modes

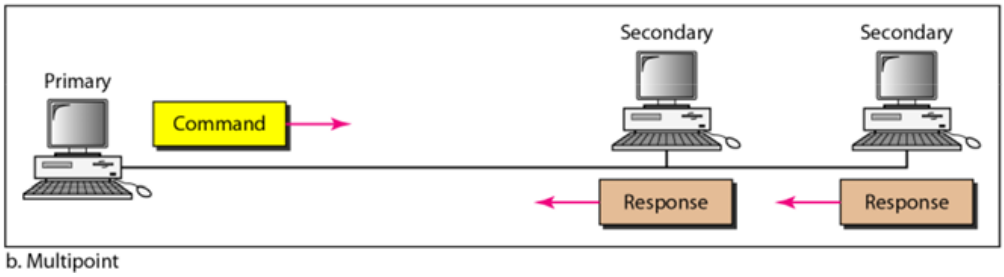
HDLC provides two common transfer modes that can be used in different configurations: normal response mode (NRM) and asynchronous balanced mode (ABM).

Normal Response Mode

In normal response mode (NRM), the station configuration is unbalanced. We have one primary station and multiple secondary stations. A primary station can send commands; a secondary station can only respond. The NRM is used for both point-to-point and multiple-point links, as shown in below Figure.



a. Point-to-point



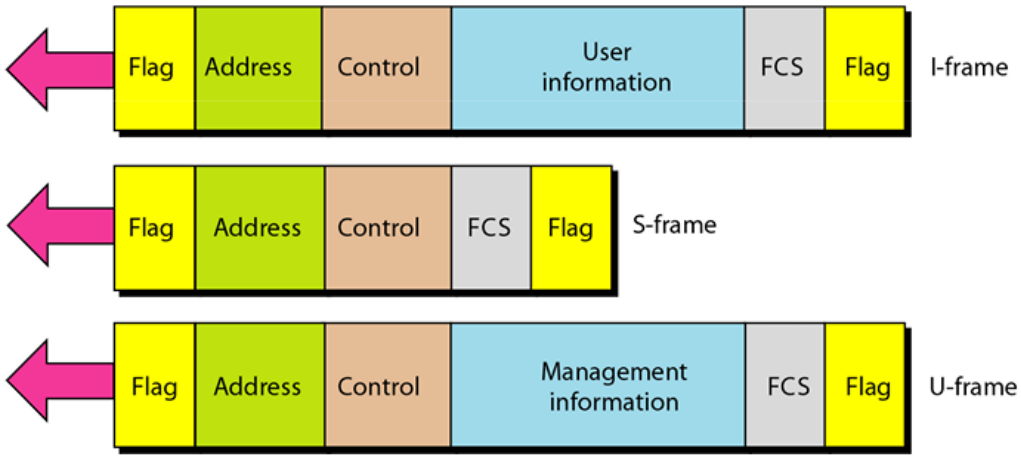
b. Multipoint

Asynchronous Balanced Mode

In asynchronous balanced mode (ABM), the configuration is balanced. The link is point-to-point, and each station can function as a primary and a secondary (acting as peers), as shown in below Figure. This is the common mode today.

Frames

To provide the flexibility necessary to support all the options possible in the modes and configurations just described, HDLC defines three types of frames: information frames (I-frames), supervisory frames (S-frames), and unnumbered frames (V-frames). Each type of frame serves as an envelope for the transmission of a different type of message. I-frames are used to transport user data and control information relating to user data (piggybacking). S-frames are used only to transport control information. V-frames are reserved for system management. Information carried by V-frames is intended for managing the link itself.



Fields

Let us now discuss the fields and their use in different frame types.

Flag field. The flag field of an HDLC frame is an 8-bit sequence with the bit pattern 01111110 that identifies both the beginning and the end of a frame and serves as a synchronization pattern for the receiver.

Address field. The second field of an HDLC frame contains the address of the secondary station. If a primary station created the frame, it contains a *to* address. If a secondary creates the frame, it contains *afrom* address. An address field can be 1 byte or several bytes long, depending on the needs of the network. One byte can identify up to 128 stations (l bit is used for another purpose). Larger networks require multiple-byte address

fields. If the address field is only 1 byte, the last bit is always a 1. If the address is more than 1 byte, all bytes but the last one will end with 0; only the last will end with 1. Ending each intermediate byte with 0 indicates to the receiver that there are more address bytes to come.

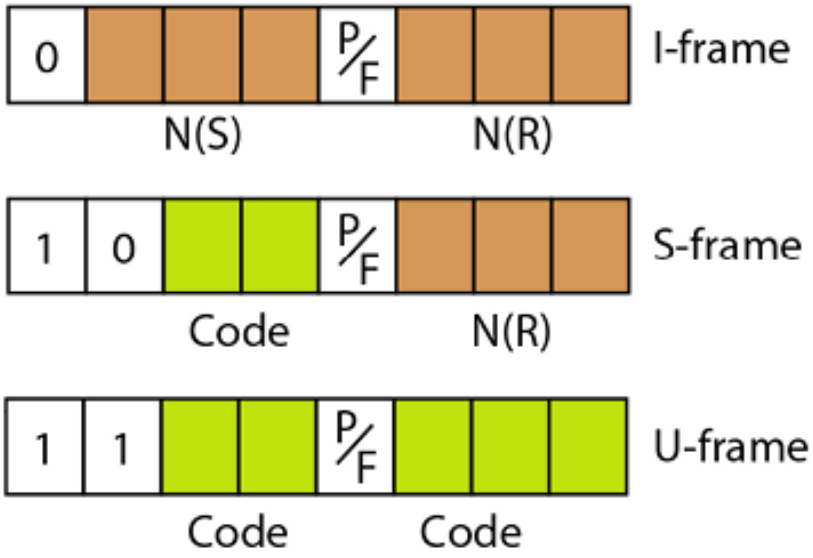
Control field. The control field is a 1- or 2-byte segment of the frame used for flow and error control. The interpretation of bits in this field depends on the frame type. We discuss this field later and describe its format for each frame type.

Information field. The information field contains the user's data from the network layer or management information. Its length can vary from one network to another.

FCS field. The frame check sequence (FCS) is the HDLC error detection field. It can contain either a 2- or 4-byte ITU-T CRC.

Control Field

The control field determines the type of frame and defines its functionality. So let us discuss the format of this field in greater detail. The format is specific for the type of frame, as shown in below Figure.



Control Field for I-Frames

I-frames are designed to carry user data from the network layer. In addition, they can include flow and error control information (piggybacking). The subfields in the control field are used to define these functions. The first bit defines the type. If the first bit of the control field is 0, this means the frame is an I-frame. The next 3 bits, called *N(S),* define the sequence number of the frame. Note that with 3 bits, we can define a sequence number between 0 and 7; but in the extension format, in which the control field is 2 bytes, this field is larger. The last 3 bits, called *N(R),* correspond to the acknowledgment number when piggybacking is used. The single bit between *N(S)* and *N(R)* is called the *PIF* bit. The *PIP* field is a single bit with a dual purpose. It has meaning only when it is set (bit = 1) and can mean poll or final. It means *poll* when the frame is sent by a primary station to a secondary (when the address field contains the address of the receiver). It means *final* when the frame is sent by a secondary to a primary (when the address field contains the address of the sender).

Control Field for S-Frames

Supervisory frames are used for flow and error control whenever piggybacking is either impossible or inappropriate (e.g., when the station either has no data of its own to send or needs to send a command or response other than an acknowledgment). S-frames do not have information fields. If the first 2 bits of the control field is 10, this means the frame is an S-frame. The last 3 bits, called *N(R),* corresponds to the acknowledgment number (ACK) or negative acknowledgment number (NAK) depending on the type of S-frame. The 2 bits called code is used to define the type of S-frame itself. With 2 bits, we can have four types of S-frames, as described below:

Receive ready (RR). If the value of the code subfield is 00, it is an RR S-frame. This kind of frame acknowledges the receipt of a safe and sound frame or group of frames. In this case, the value *N(R)* field defines the acknowledgment number.

Receive not ready (RNR). If the value of the code subfield is 10, it is an RNR S-frame. This kind of frame is an RR frame with additional functions. It acknowledges the receipt of a frame or group of frames, and it announces that the receiver is busy and cannot receive more frames. It acts as a kind of congestion control mechanism by asking the sender to slow down. The value of *NCR)* is the acknowledgment number.
Reject (REJ). If the value of the code subfield is 01, it is a REJ S-frame. This is a NAK frame, but not like the one used for Selective Repeat ARQ. It is a NAK that can be used in *Go-Back-N* ARQ to improve the efficiency of the process by informing the sender, before the sender time expires, that the last frame is lost or damaged. The value of *NCR)* is the negative acknowledgment number.
Selective reject (SREJ). If the value of the code subfield is 11, it is an SREJ S-frame. This is a NAK frame used in Selective Repeat ARQ. Note that the HDLC Protocol uses the term *selective reject* instead of *selective repeat.* The value of *N(R)* is the negative acknowledgment number.

Control Field for V-Frames

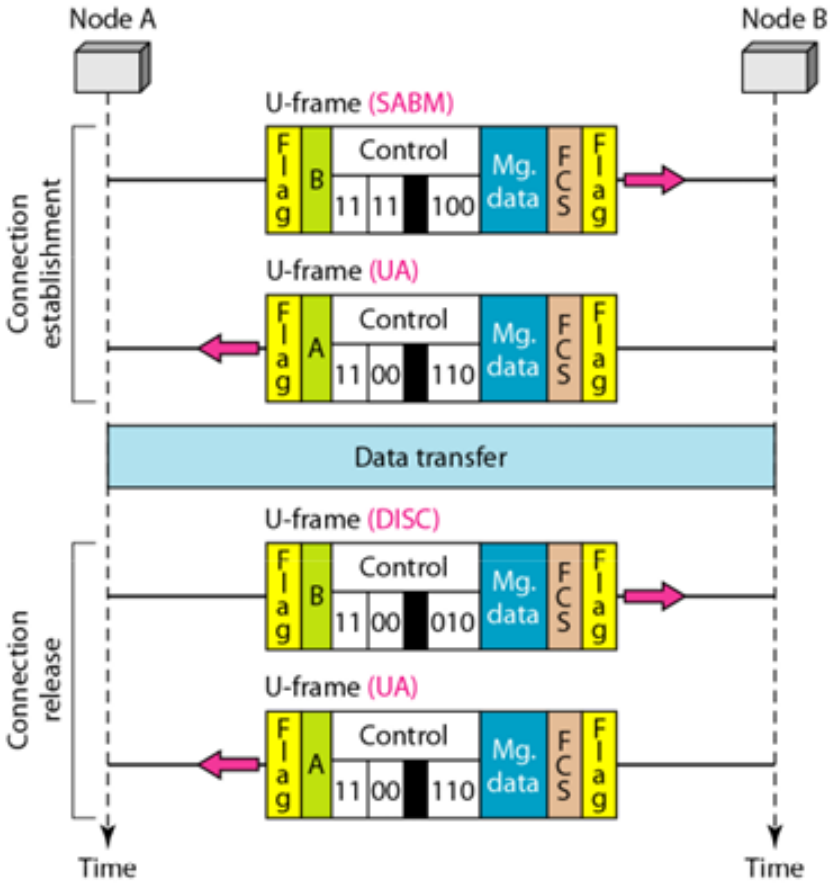
Unnumbered frames are used to exchange session management and control information between connected devices. Unlike S-frames, U-frames contain an information field, but one used for system management information, not user data. As with S-frames, however, much of the information carried by U-frames is contained in codes included in the control field. U-frame codes are divided into two sections: a 2-bit prefix before the P/F bit and a 3-bit suffix after the P/F bit. Together, these two segments (5 bits) can be used to create up to 32 different types of U-frames. Some of the more common types are shown in below Table.

Uframe control command and response

Code	*Command*	*Response*	*Meaning*
00 001	SNRM		Set normal response mode
11 011	SNRME		Set normal response mode, extended
11 100	SABM	**DM**	Set asynchronous balanced mode or **disconnect mode**
11 110	SABME		Set asynchronous balanced mode, extended
00 000	UI	**UI**	Unnumbered information
00 110		**UA**	**Unnumbered acknowledgment**
00 010	DISC	**RD**	Disconnect or **request disconnect**
10 000	SIM	**RIM**	Set initialization mode or **request information mode**
00 100	UP		Unnumbered poll
11 001	RSET		Reset
11 101	XID	**XID**	Exchange ID
10 001	FRMR	**FRMR**	Frame reject

Example: *Connection/Disconnection*

Below Figure shows how V-frames can be used for connection establishment and connection release. Node A asks for a connection with a set asynchronous balanced mode (SABM) frame; node B gives a positive response with an unnumbered acknowledgment (VA) frame. After these two exchanges, data can be transferred between the two nodes (not shown in the figure). After data transfer, node A sends a DISC (disconnect) frame to release the connection; it is confirmed by node B responding with a VA (unnumbered acknowledgment).



POINT-TO-POINT PROTOCOL:

Although HDLC is a general protocol that can be used for both point-to-point and multipoint configurations, one of the most common protocols for point-to-point access is the Point-to-Point Protocol (PPP). Today, millions of Internet users who need to connect their home computers to the server of an Internet service provider use PPP. The majority of these users have a traditional modem; they are connected to the Internet through a telephone line, which provides the services of the physical layer. But to control and manage the transfer of data, there is a need for a point-to-point protocol at the data link layer. PPP is by far the most common.

PPP provides several services:

1. PPP defines the format of the frame to be exchanged between devices.
2. PPP defines how two devices can negotiate the establishment of the link and the exchange of data.
3. PPP defines how network layer data are encapsulated in the data link frame.
4. PPP defines how two devices can authenticate each other.
5. PPP provides multiple network layer services supporting a variety of network layer protocols.
6. PPP provides connections over multiple links.
7. PPP provides network address configuration. This is particularly useful when a home user needs a temporary network address to connect to the Internet.

On the other hand, to keep PPP simple, several services are missing:

I. PPP does not provide flow control. A sender can send several frames one after another with no concern about overwhelming the receiver.

2. PPP has a very simple mechanism for error control. A CRC field is used to detect errors. If the frame is corrupted, it is silently discarded; the upper-layer protocol needs to take care of the problem. Lack of error control and sequence numbering may cause a packet to be received out of order.

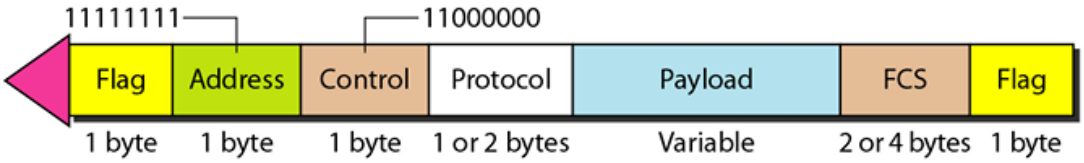
3. PPP does not provide a sophisticated addressing mechanism to handle frames in a multipoint configuration.

Framing:

PPP is a byte-oriented protocol. Framing is done according to the discussion of byte oriented protocols.

Frame Format

Below Figure shows the format of a PPP frame. The description of each field follows:



Flag. A PPP frame starts and ends with a I-byte flag with the bit pattern 01111110. Although this pattern is the same as that used in HDLC, there is a big difference. PPP is a byte-oriented protocol; HDLC is a bit-oriented protocol. The flag is treated as a byte, as we will explain later.

Address. The address field in this protocol is a constant value and set to 11111111(broadcast address). During negotiation (discussed later), the two parties may agree to omit this byte.

Control. This field is set to the constant value 11000000 (imitating unnumbered frames in HDLC). As we will discuss later, PPP does not provide any flow control. Error control is also limited to error detection. This means that this field is not needed at all, and again, the two parties can agree, during negotiation, to omit this byte.

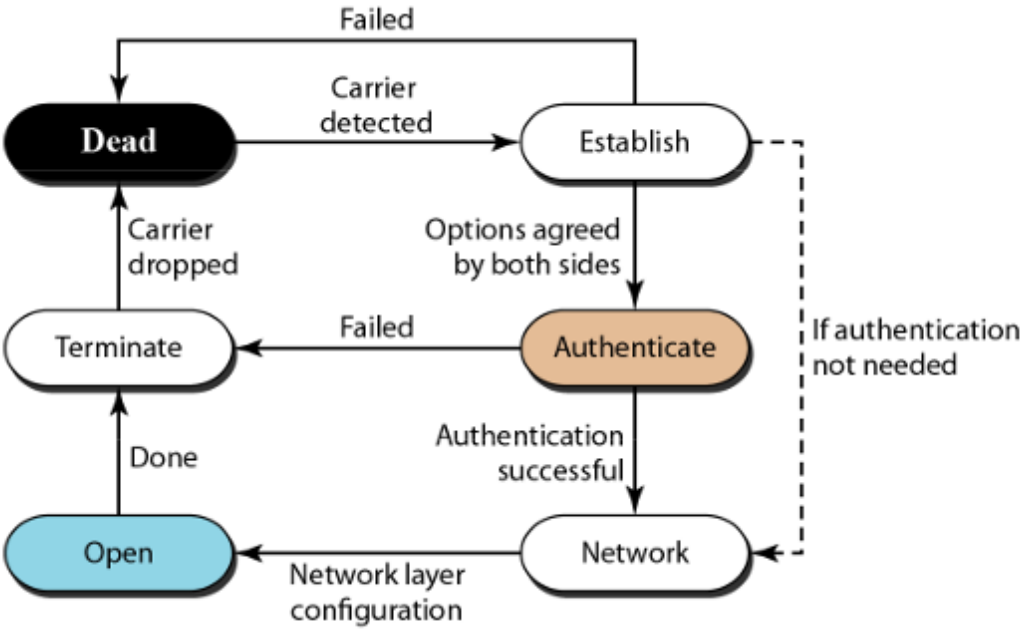
Protocol. The protocol field defines what is being carried in the data field: either user data or other information. We discuss this field in detail shortly. This field is by default 2 bytes long, but the two parties can agree to use only 1 byte.

Payload field. This field carries either the user data or other information that we will discuss shortly. The data field is a sequence of bytes with the default of a maximum of 1500 bytes; but this can be changed during negotiation. The data field is byte stuffed if the flag byte pattern appears in this field. Because there is no field defining the size of the data field, padding is needed if the size is less than the maximum default value or the maximum negotiated value.

FCS. The frame check sequence (FCS) is simply a 2-byte or 4-byte standard CRC.

Transition Phases

A PPP connection goes through phases which can be shown in a transition phase diagram.



Dead. In the dead phase the link is not being used. There is no active carrier (at the physical layer) and the line is quiet.

Establish. When one of the nodes starts the communication, the connection goes into this phase. In this phase, options are negotiated between the two parties. If the negotiation is successful, the system goes to the

authentication phase (if authentication is required) or directly to the networking phase. The link control protocol packets, discussed shortly, are used for this purpose. Several packets may be exchanged here.

Authenticate. The authentication phase is optional; the two nodes may decide, during the establishment phase, not to skip this phase. However, if they decide to proceed with authentication, they send several authentication packets, discussed later. If the result is successful, the connection goes to the networking phase; otherwise, it goes to the termination phase.

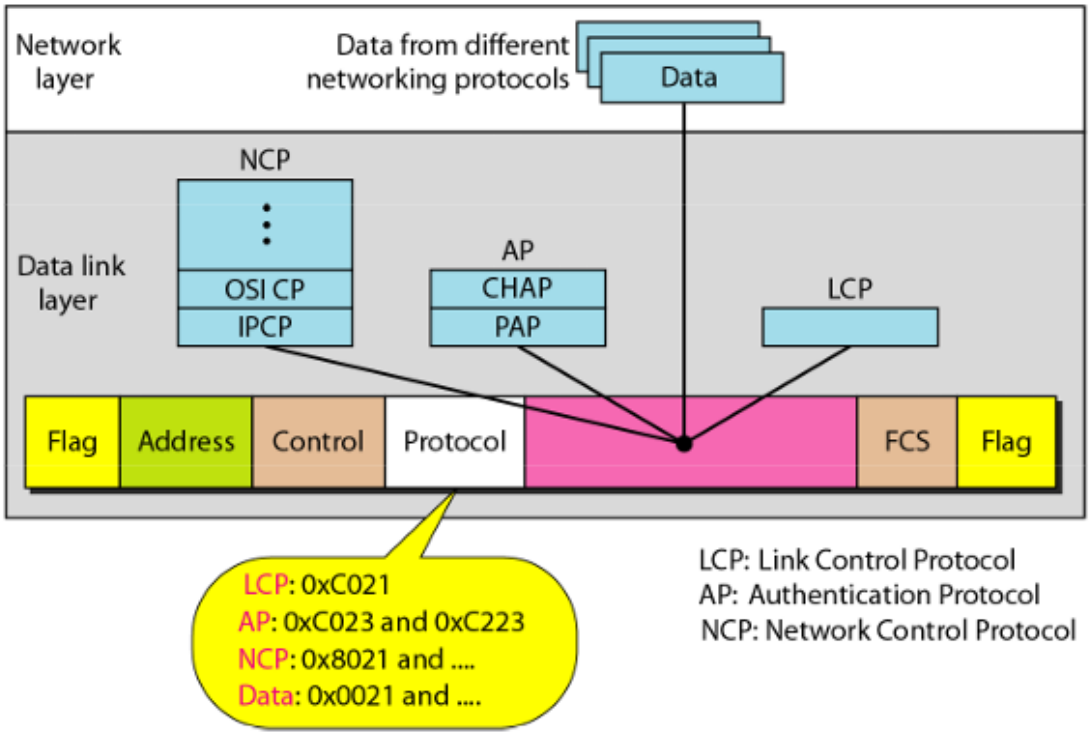
Network. In the network phase, negotiation for the network layer protocols takes place. PPP specifies that two nodes establish a network layer agreement before data at the network layer can be exchanged. The reason is that PPP supports multiple protocols at the network layer. If a node is running multiple protocols simultaneously at the network layer, the receiving node needs to know which protocol will receive the data.

Open. In the open phase, data transfer takes place. When a connection reaches this phase, the exchange of data packets can be started. The connection remains in this phase until one of the endpoints wants to terminate the connection.

Terminate. In the termination phase the connection is terminated. Several packets are exchanged between the two ends for house cleaning and closing the link.

Multiplexing

Although PPP is a data link layer protocol, PPP uses another set of other protocols to establish the link, authenticate the parties involved, and carry the network layer data. Three sets of protocols are defined to make PPP powerful: the Link Control Protocol (LCP), two Authentication Protocols (APs), and several Network Control Protocols (NCPs). At any moment, a PPP packet can carry data from one of these protocols in its data field, as shown in below Figure. Note that there is one LCP, two APs, and several NCPs. Data may also come from several different network layers.

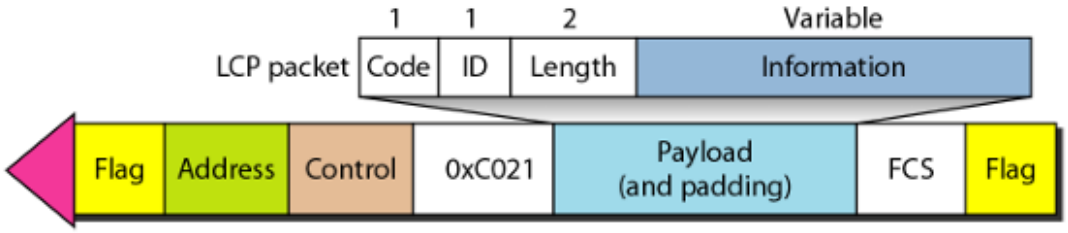


Link Control Protocol

The **Link Control Protocol** (LCP) is responsible for establishing, maintaining, configuring, and terminating links. It also provides negotiation mechanisms to set options between the two endpoints. Both endpoints of the link must reach an agreement about the options before the link can be established.

All LCP packets are carried in the payload field of the PPP frame with the protocol field set to C021 in hexadecimal.

The code field defines the type of LCP packet. There are 11 types of packets as shown in below Table.



Code	*Packet Type*	*Description*
0x01	Configure-request	Contains the list of proposed options and their values
0x02	Configure-ack	Accepts all options proposed
0x03	Configure-nak	Announces that some options are not acceptable
0x04	Configure-reject	Announces that some options are not recognized
0x05	Terminate-request	Request to shut down the line
0x06	Terminate-ack	Accept the shutdown request
0x07	Code-reject	Announces an unknown code
0x08	Protocol-reject	Announces an unknown protocol
0x09	Echo-request	A type of hello message to check if the other end is alive
0x0A	Echo-reply	The response to the echo-request message
0x0B	Discard-request	A request to discard the packet

There are three categories of packets. The first category, comprising the first four packet types, is used for link configuration during the establish phase. The second category, comprising packet types 5 and 6, is used for link termination during the termination phase. The last five packets are used for link monitoring and debugging.

The ID field holds a value that matches a request with a reply. One endpoint inserts a value in this field, which will be copied into the reply packet. The length field defines the length of the entire LCP packet. The information field contains information, such as options, needed for some LCP packets.

There are many options that can be negotiated between the two endpoints. Options are inserted in the information field of the configuration packets. In this case, the information field is divided into three fields: option type, option length, and option data. We list some of the most common options in below Table.

Option	*Default*
Maximum receive unit (payload field size)	1500
Authentication protocol	None
Protocol field compression	Off
Address and control field compression	Off

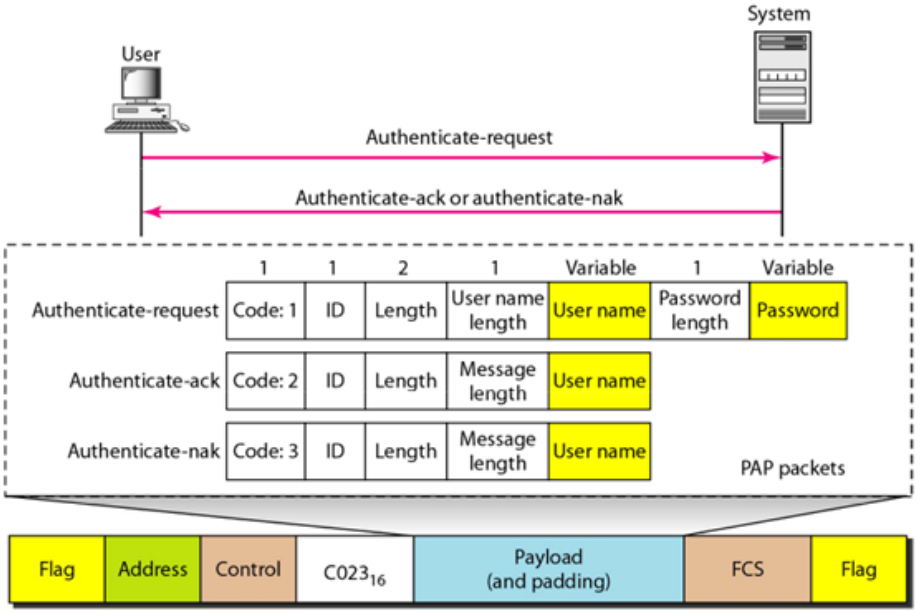
Authentication Protocols

Authentication plays a very important role in PPP because PPP is designed for use over dial-up links where verification of user identity is necessary. **Authentication** means validating the identity of a user who needs to access a set of resources. PPP has created two protocols for authentication: Password Authentication Protocol and Challenge Handshake Authentication Protocol.

PAP The **Password Authentication Protocol (PAP)** is a simple authentication procedure with a two-step process:

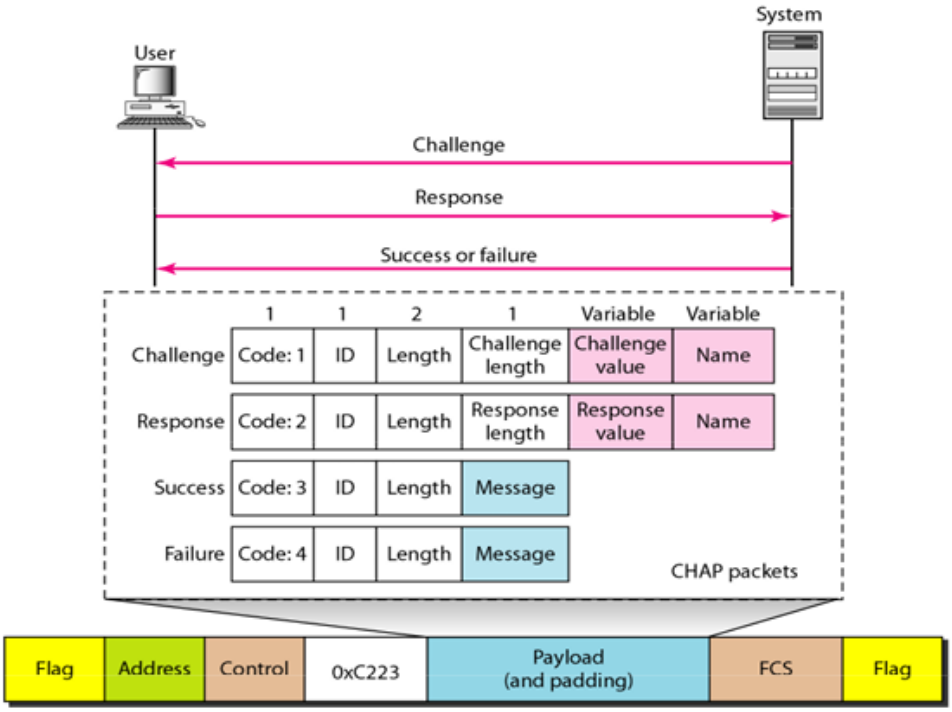
1. The user who wants to access a system sends an authentication identification (usually the user name) and a password.
2. The system checks the validity of the identification and password and either accepts or denies connection.

Below Figure shows the three types of packets used by PAP and how they are actually exchanged. When a PPP frame is carrying any PAP packets, the value of the protocol field is OxC023. The three PAP packets are authenticate-request, authenticate-ack, and authenticate-nak. The first packet is used by the user to send the user name and password. The second is used by the system to allow access. The third is used by the system to deny access.



CHAP The **Challenge Handshake Authentication Protocol (CHAP)** is a three-way hand-shaking authentication protocol that provides greater security than PAP. **In** this method, the password is kept secret; it is never sent online.

1. The system sends the user a challenge packet containing a challenge value, usually a few bytes.
2. The user applies a predefined function that takes the challenge value and the user's own password and creates a result. The user sends the result in the response packet to the system.
3. The system does the same. It applies the same function to the password of the user (known to the system) and the challenge value to create a result. If the result created is the same as the result sent in the response packet, access is granted; otherwise, it is denied. CHAP is more secure than PAP, especially if the system continuously changes the challenge value. Even if the intruder learns the challenge value and the result, the password is still secret. Below Figure shows the packets and how they are used.

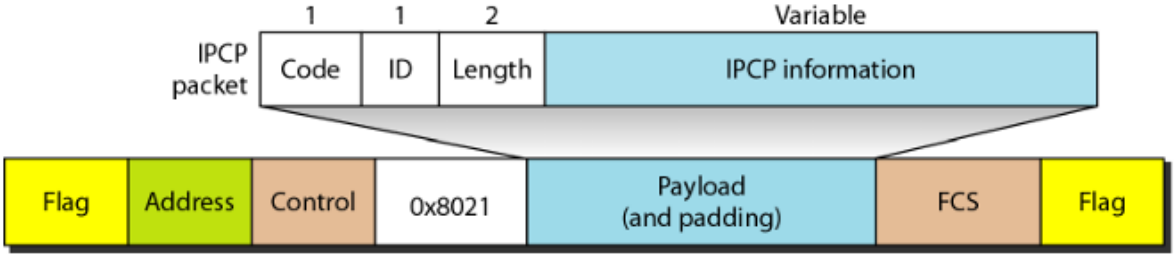


CHAP packets are encapsulated in the PPP frame with the protocol value C223 in hexadecimal. There are four CHAP packets: challenge, response, success, and failure. The first packet is used by the system to send the challenge value. The second is used by the user to return the result of the calculation. The third is used by the system to allow access to the system. The fourth is used by the system to deny access to the system.

Network Control Protocols

PPP is a multiple-network layer protocol. It can carry a network layer data packet from protocols defined by the Internet, OSI, Xerox, DECnet, AppleTalk, Novel, and so on. To do this, PPP has defined a specific Network Control Protocol for each network protocol. For example, IPCP (Internet Protocol Control Protocol) configures the link for carrying IP data packets. Xerox CP does the same for the Xerox protocol data packets, and so on.

IPCP One NCP protocol is the Internet Protocol Control Protocol (IPCP). This protocol configures the link used to carry IP packets in the Internet. IPCP is especially of interest to us. The format of an IPCP packet is shown in below Figure. Note that the value of the protocol field in hexadecimal is 8021.

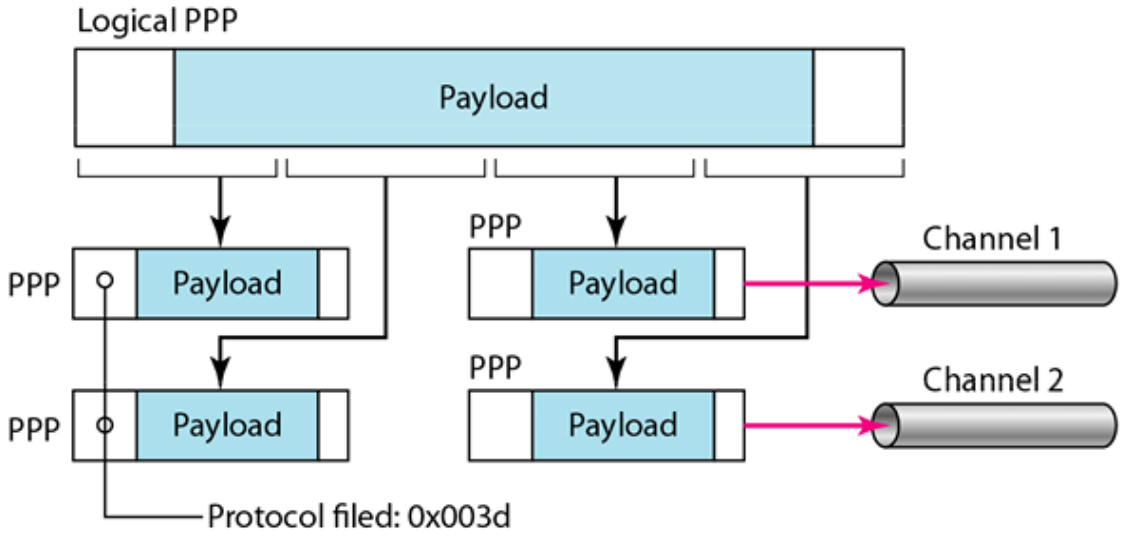


IPCP defines seven packets, distinguished by their code values, as shown in below Table

Code	*IPCP Packet*
OxO!	Configure-request
Ox02	Configure-ack
Ox03	Configure-nak
Ox04	Configure-reject
Ox05	Terminate-request
Ox06	Terminate-ack
Ox07	Code-reject

Multilink PPP

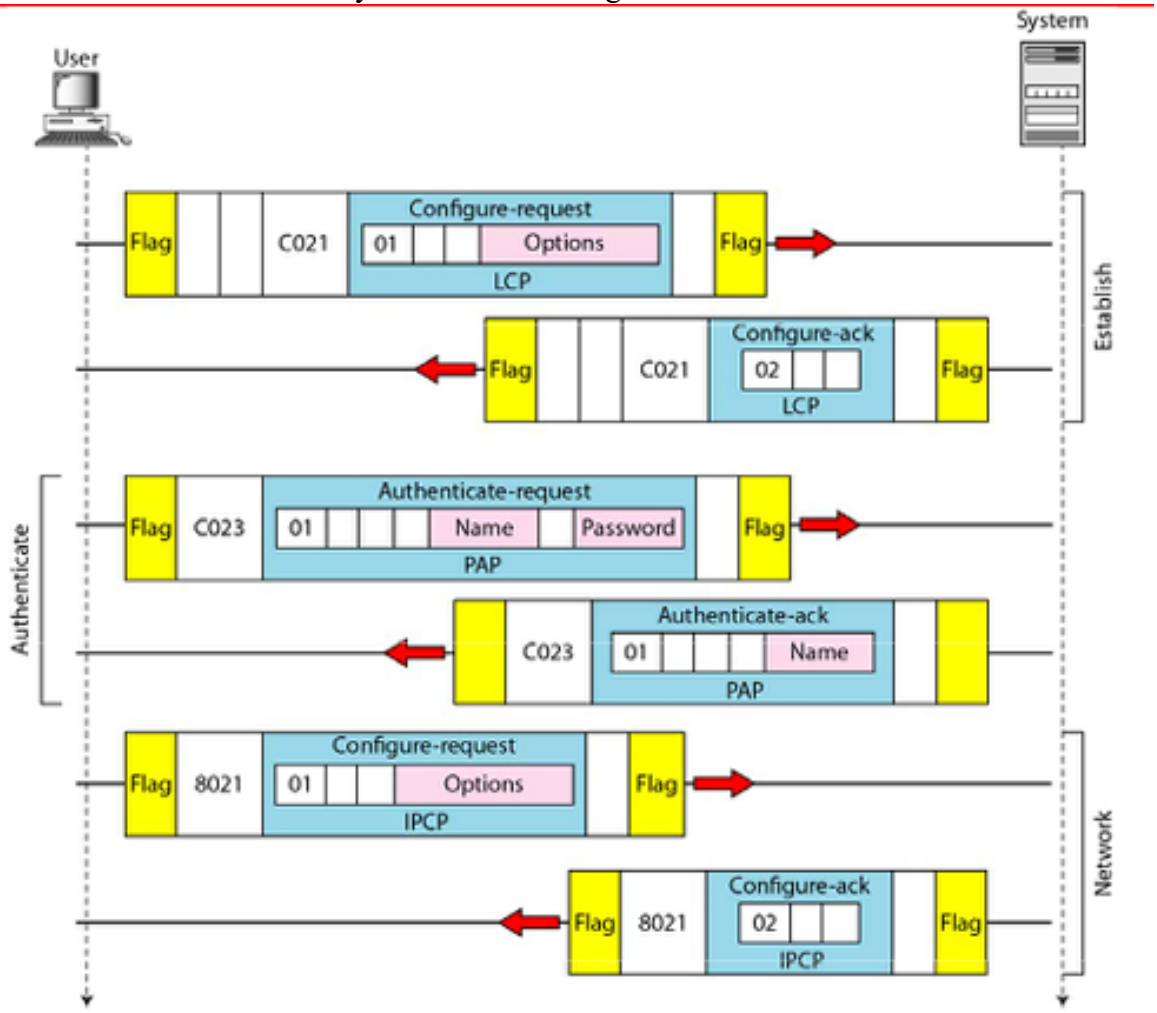
PPP was originally designed for a single-channel point-to-point physical link. The availability of multiple channels in a single point-to-point link motivated the development of Multilink PPP. In this case, a logical PPP frame is divided into several actual PPP frames. A segment of the logical frame is carried in the payload of an actual PPP frame, as shown in below Figure. To show that the actual PPP frame is carrying a fragment of a logical PPP frame, the protocol field is set to Ox003d. This new development adds complexity. For example, a sequence number needs to be added to the actual PPP frame to show a fragment's position in the logical frame.

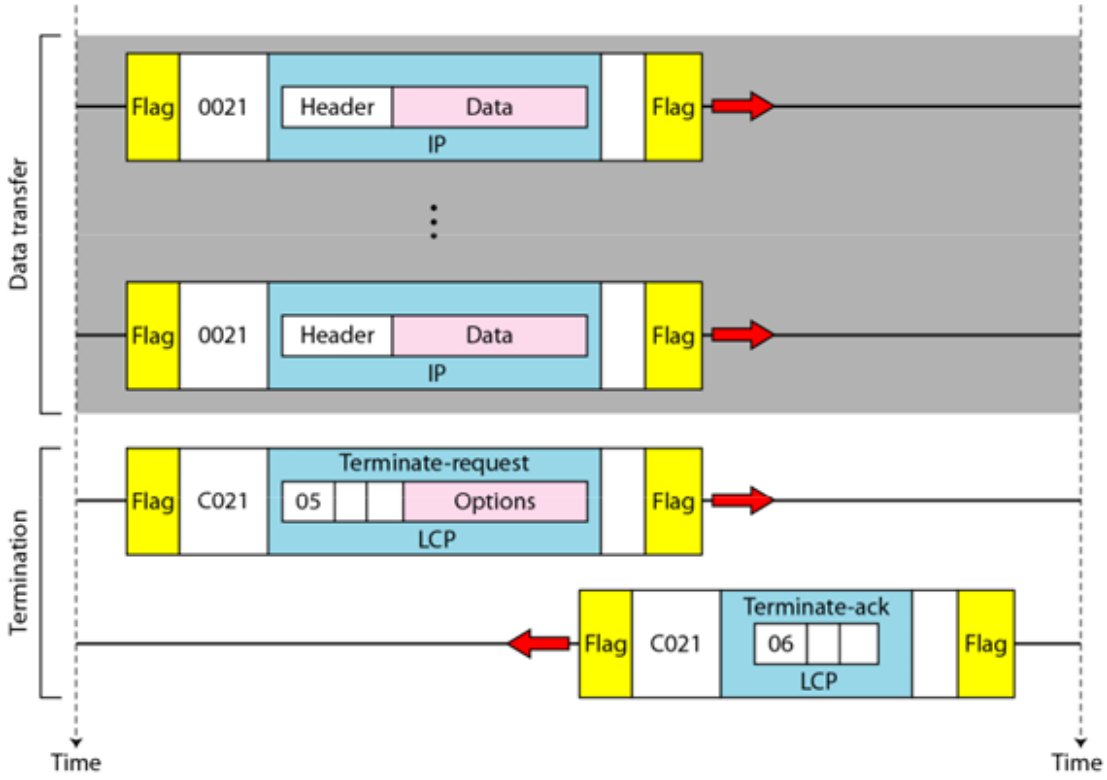


Example

Let us go through the phases followed by a network layer packet as it is transmitted through a PPP connection. Below Figure shows the steps. For simplicity, we assume unidirectional movement of data from the user site to the system site (such as sending an e-mail through an ISP).

The first two frames show link establishment. We have chosen two options (not shown in the figure): using PAP for authentication and suppressing the address control fields. Frames 3 and 4 are for authentication. Frames 5 and 6 establish the network layer connection using IPCP.




Data transfer
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