BCS502 – COMPUTER NETWORKS

Faculty:

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Module 4

Transport Layer

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Main topics in Module 4

- The Transport service
- Different transport layer protocols – TCP and UDP
- Flow control and Error control
- Congestion control

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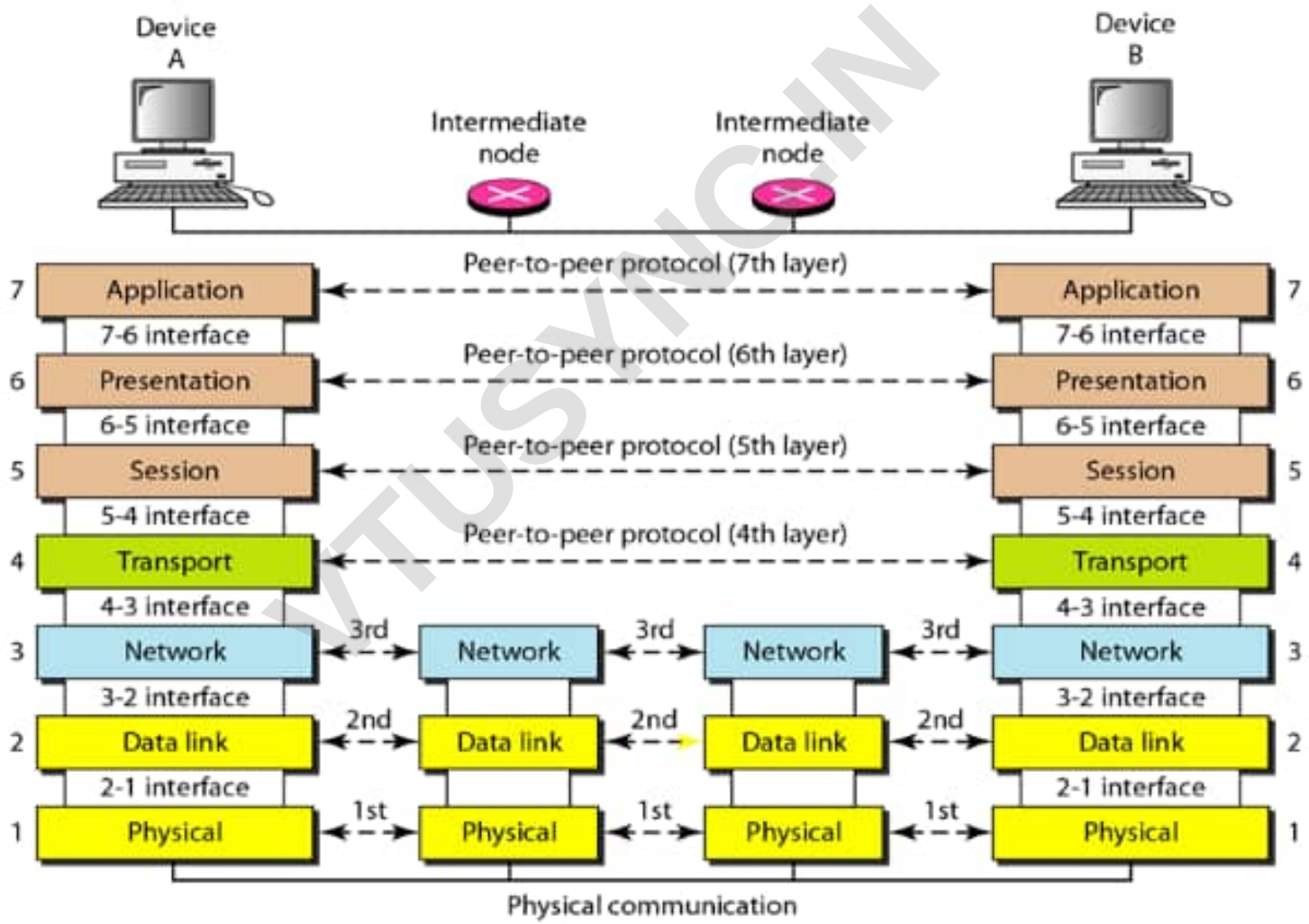
OSI Reference model



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Functions of the Transport Layer

- It is the lowest host-to-host protocol.
- Facilitates communication between host processes.
- Divides the message into data segments on the sending side, and reassembles them on the receiving side.
- Takes complete responsibility for a reliable, end-to-end transport of the message.
- Packets can be: Corrupted, or Lost, or Delivered out of sequence.
- Also takes care of Flow control between the hosts.

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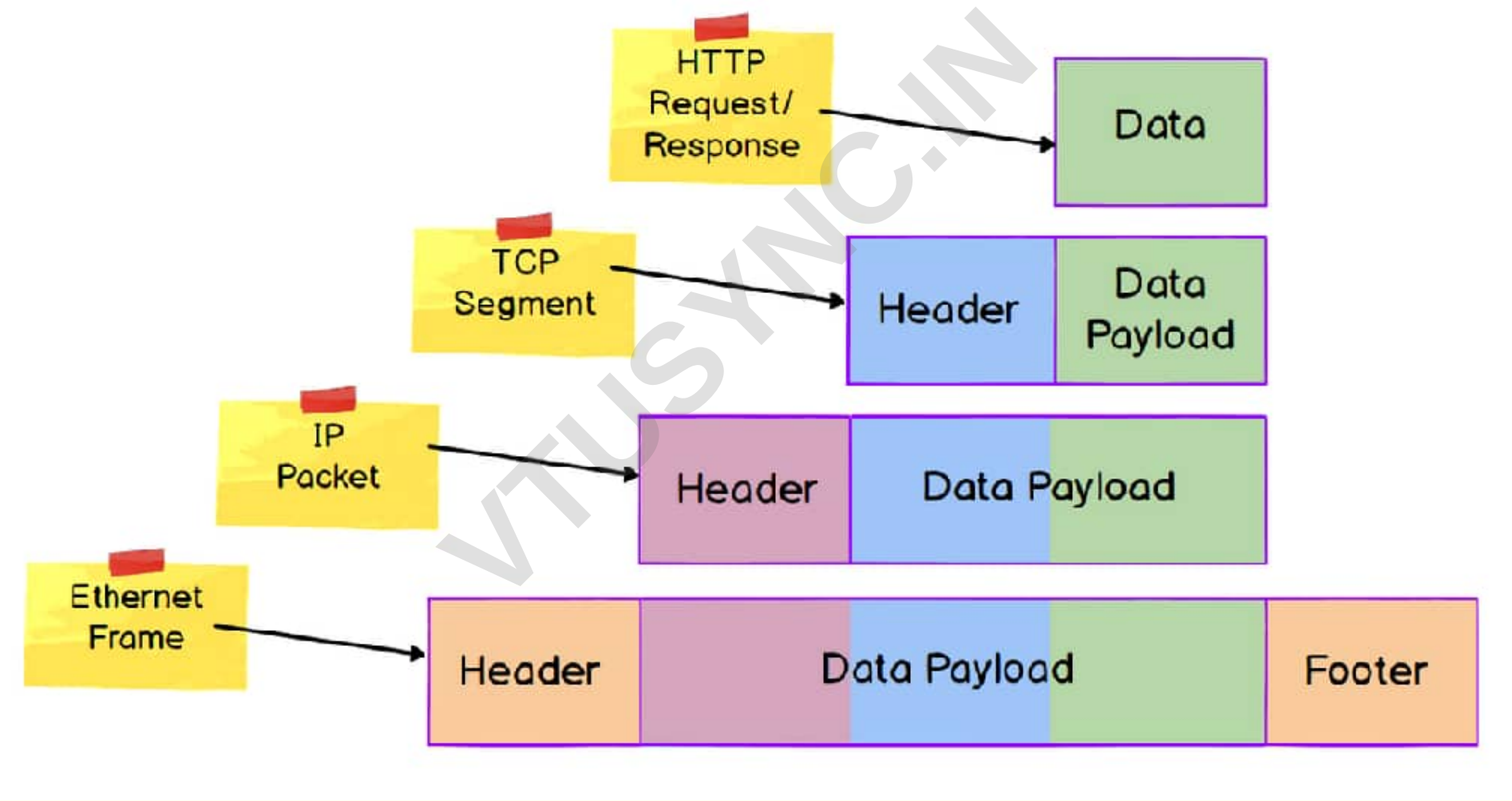
Segments in the Transport Layer



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Transport entity

- The software and hardware within the Transport layer that does the work is called the Transport entity.
- The Transport entity is generally located (when it comes to Internet) in the Operating System, or in a library package bound into network applications.
- It can also be in a separate user process or even in the Network Interface card.

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Transport layer addressing

- When an application process wishes to set up a connection to a remote application process (on another host), it must specify it unambiguously.
- In the Internet, such addresses are called Ports. (in generic terms, an address is called TSAP – Transport Service Access Point).
- This is similar to an IP address in the Network layer (NSAP – Network Service Access Point).
- However, note that a host could be running many applications simultaneously, in which case a single IP address will lead to many Port addresses.

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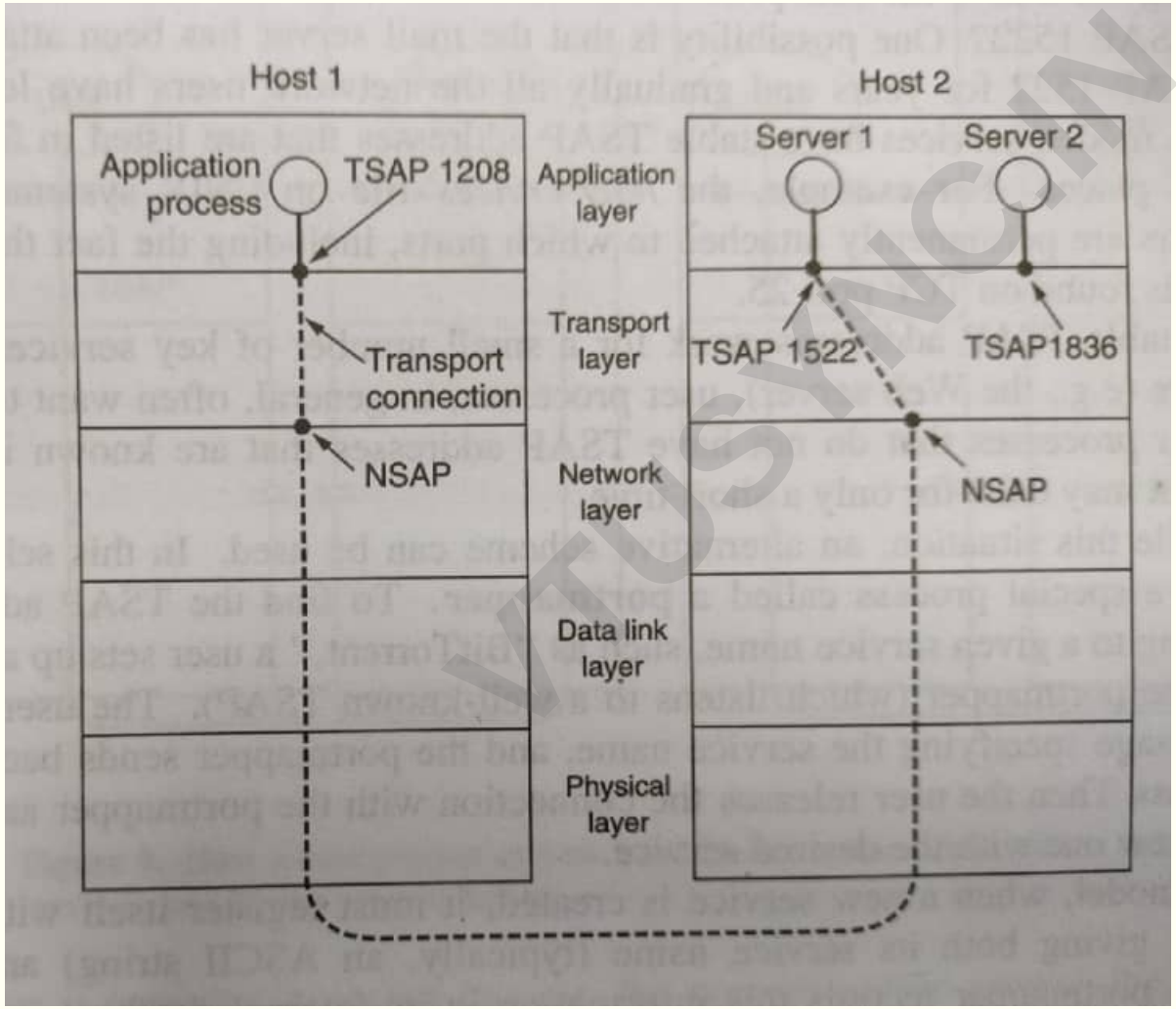
Transport layer addressing



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The Transport layer at the destination host should know the port to which the data is to be delivered.

Eg. shown is from TSAP 1208 on Host 1 to TSAP 1522 on Host 2.

Connection-oriented vs. Connectionless sessions

- The Network layer too had Connectionless (Datagram Packet Switching) and Connection-oriented (Virtual Circuit Packet Switching) transmissions.
- However, they were restricted to transmission from the transmitting router (representing the transmitting host) to the terminal router (representing the destination host).
- Any deficiencies in the service has to be handled by the Transport layer.
- Transport layer has two main protocols:
  - TCP (Transport Control Protocol) – Connection-oriented
  - UDP (User Datagram Protocol) – Connectionless.

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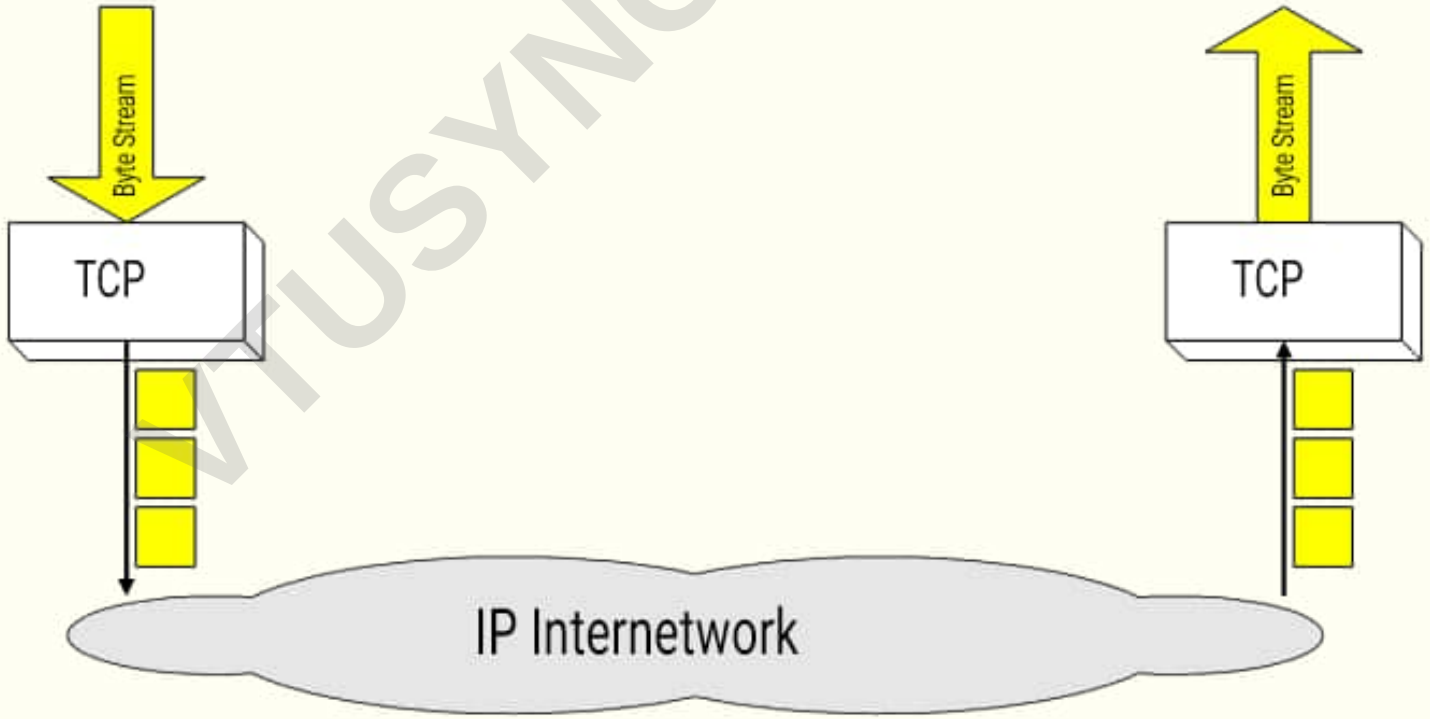
Transmission Control Protocol (TCP)

- Provides a reliable, end-to-end, byte-stream transmission over an unreliable internetwork.



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Transmission Control Protocol (TCP)

- Is Connection-oriented
  - Explicit set-up and tear-down of TCP session
- Offers Stream-of-bytes service
  - Receives a stream of bytes from higher layers.
  - Passes them as Segments to the lower layers.
- Provides Reliable, in-order delivery
  - Checksums to detect corrupted data
  - Acknowledgments & retransmissions for reliable delivery
  - Sequence numbers to detect losses and reorder data

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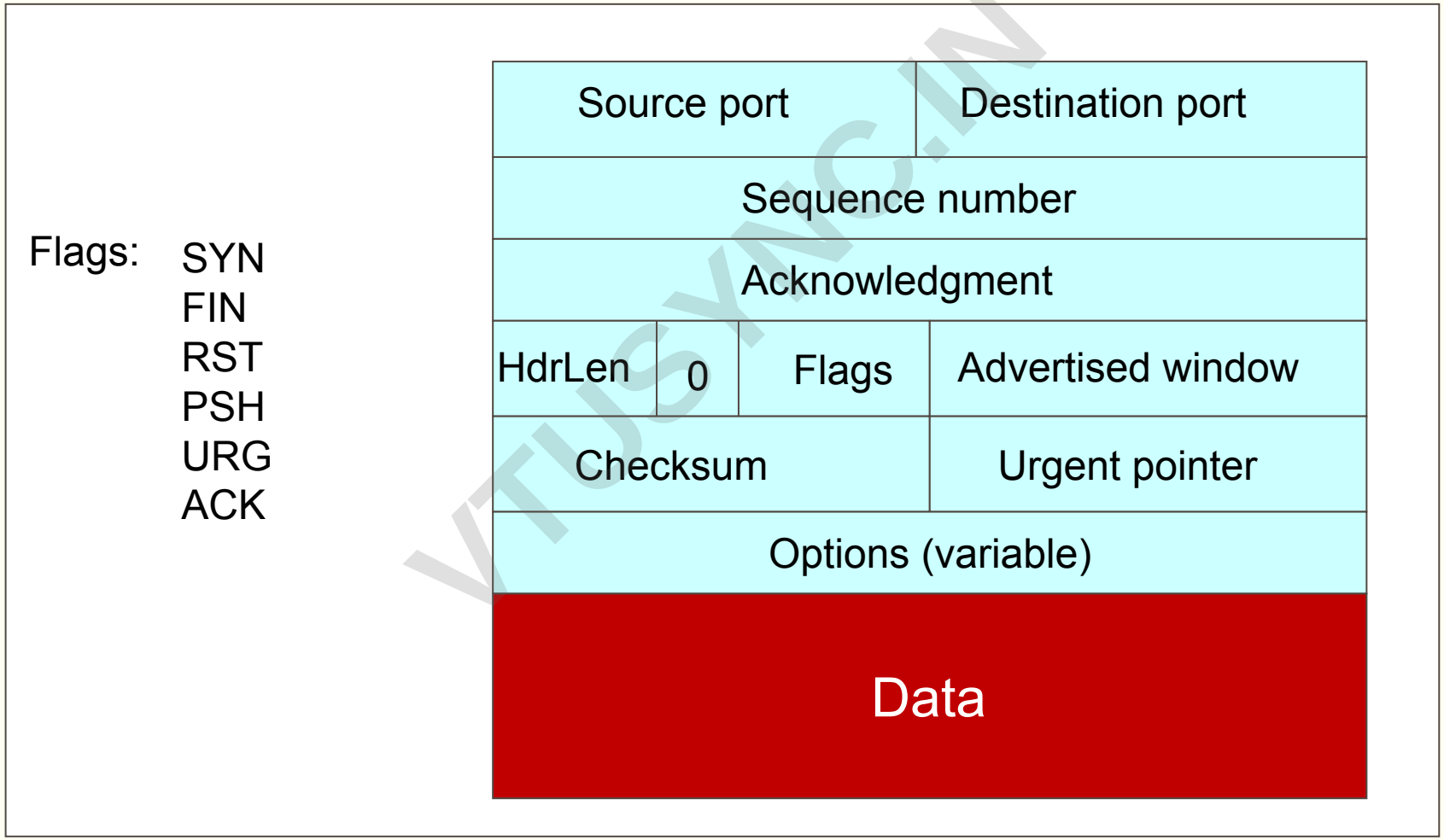
TCP header



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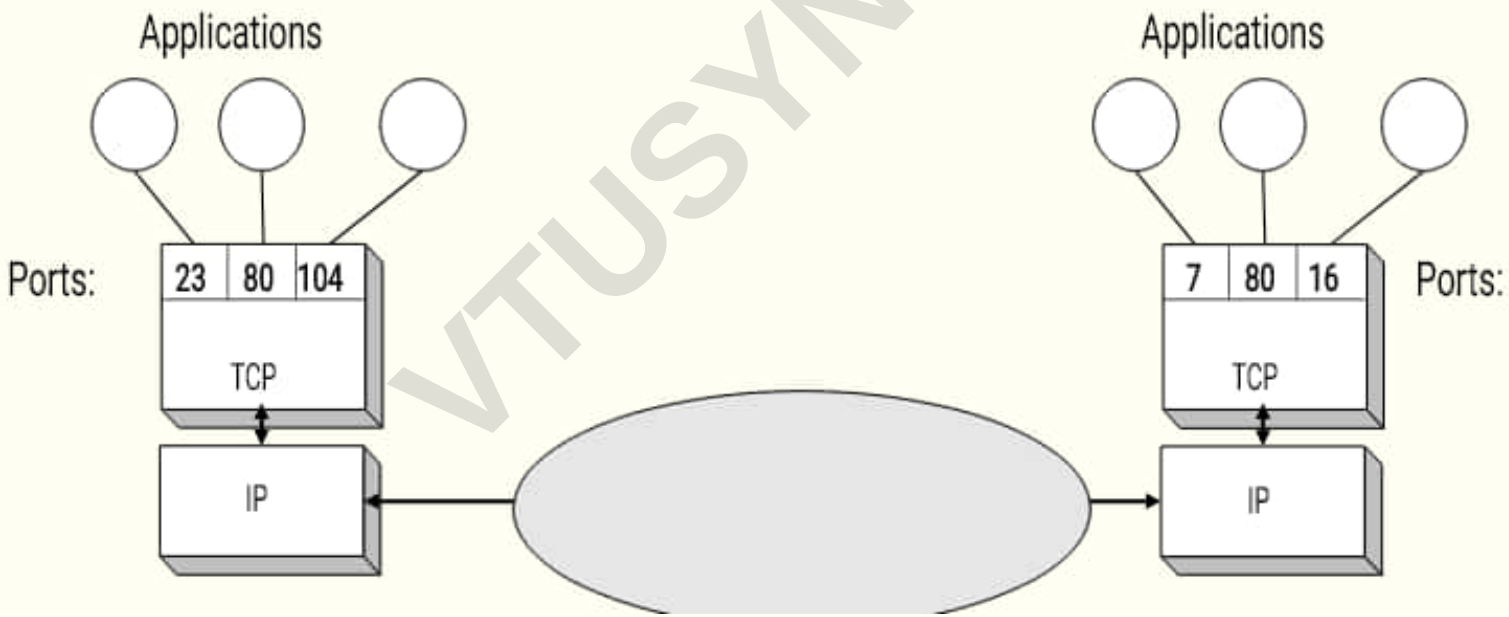
TCP Header fields

- **Port Number:**
  - A port number identifies the endpoint of a connection.
  - A pair `<IP address, port number>` identifies one endpoint of a connection.



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TCP Header fields

- **Sequence Number:**
  - The range of the 32-bit SeqNo is

    $0 <= SeqNo <= 2^{32} - 1 \approx 4.3$ Giga
  - Each sequence number identifies a byte in the byte stream.
  - Initial Sequence Number (ISN) of a connection is set during connection establishment.

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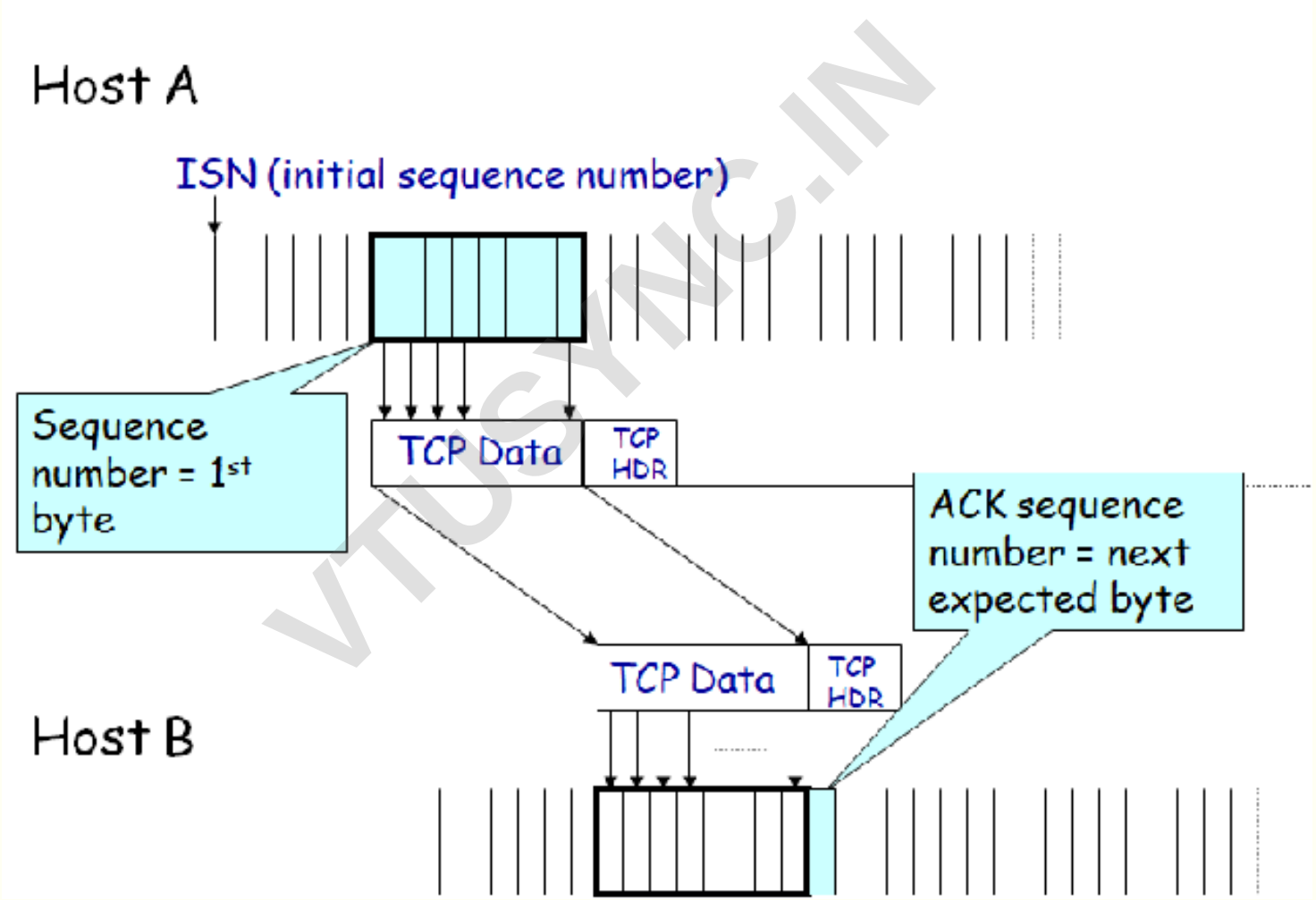
TCP Header fields



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TCP Header fields

- **Acknowledgement Number:**
  - Acknowledgements are piggybacked.
  - A segment from A -> B can contain an acknowledgement for a data sent in the B -> A direction.
  - If a host sends an AckNo in a segment it sets the "ACK flag"
  - The AckNo contains the SeqNo of the next byte that a host wants to receive.
  - Example: The acknowledgement for a segment with sequence numbers 0-1500 is AckNo=1501.

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TCP Header fields

- **Header Length:**
  - Length of header in terms of 32-bit words (4 bytes)
  - The TCP header has a variable length (with minimum 20 bytes)
  - This 4-bit field can hold numbers from 5 (20 bytes) to 15 (60 bytes)
- **Flags:**
  - SYN: Synchronize sequence numbers
    - Sent in the first packet when initiating a connection
  - FIN: Sender is finished with sending
    - Used for closing a connection
    - Both sides of a connection must send a FIN

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TCP Header fields

- **Flags: (contd)**
  - RST: Causes the receiver to Reset the connection
    - Receiver terminates the connection and indicates higher layer application about the reset
  - ACK: Acknowledgement Number is valid
  - URG: Urgent pointer is valid
    - If the bit is set, the following bytes, in the range given below, contain an urgent message:

      SeqNo <= urgent message <= SeqNo+urgent pointer
  - PSH: PUSH flag is used to tell the receiver to pass all data (that it currently has) to the application.
    - Normally set by sender when the sender's buffer is empty.

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TCP Header fields

- **Window Size:**
  - Each side advertises the size of its sliding window.
  - Window size is the maximum number of bytes that a receiver can accept.
  - Maximum window size is 2^{16}-1= 65535 bytes
- **TCP Checksum:**
  - TCP checksum covers over both TCP header **and** TCP data
- **Urgent Pointer:**
  - Only valid if **URG** flag is set

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Establishing a connection (Hand-shaking)

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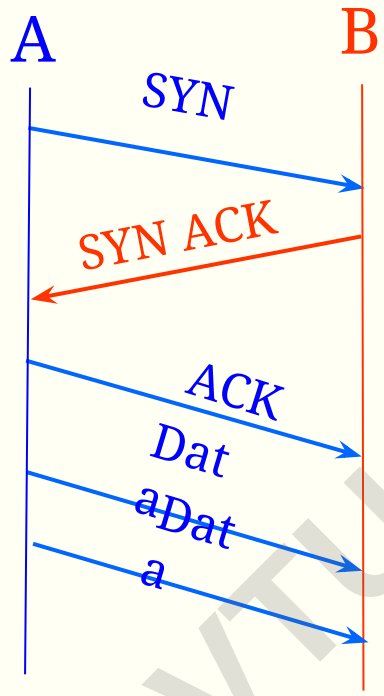
Establishing a TCP Connection: 3-way handshake.



Each host tells its ISN (Initial sequence number) to the other host.

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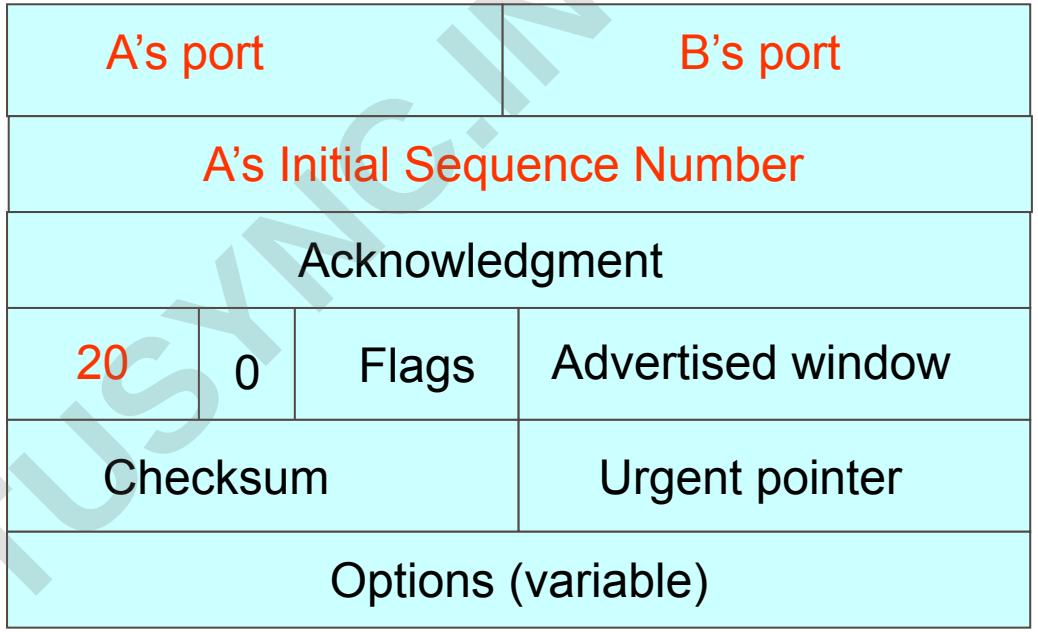
- Three-way handshake to establish connection
  - Host A sends a **SYN** (open) to the host B
  - Host B returns a SYN acknowledgment (**SYN ACK**)
  - Host A sends an **ACK** to acknowledge the SYN ACK

Step 1: A's Initial SYN Packet

Flags: SYN
FIN
RST
PSH
URG
ACK



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A tells B it wants to open a connection…

Step 2: B's response (SYN-ACK) packet to A

Flags:
SYN
FIN
RST
PSH
URG
ACK

B's port	A's port
B's Initial Sequence Number	
A's ISN plus 1	
20 | 0 | Flags	Advertised window
Checksum	Urgent pointer
Options (variable)	

B tells A it accepts, and is ready to hear the next byte…

… upon receiving this packet, A can start sending data

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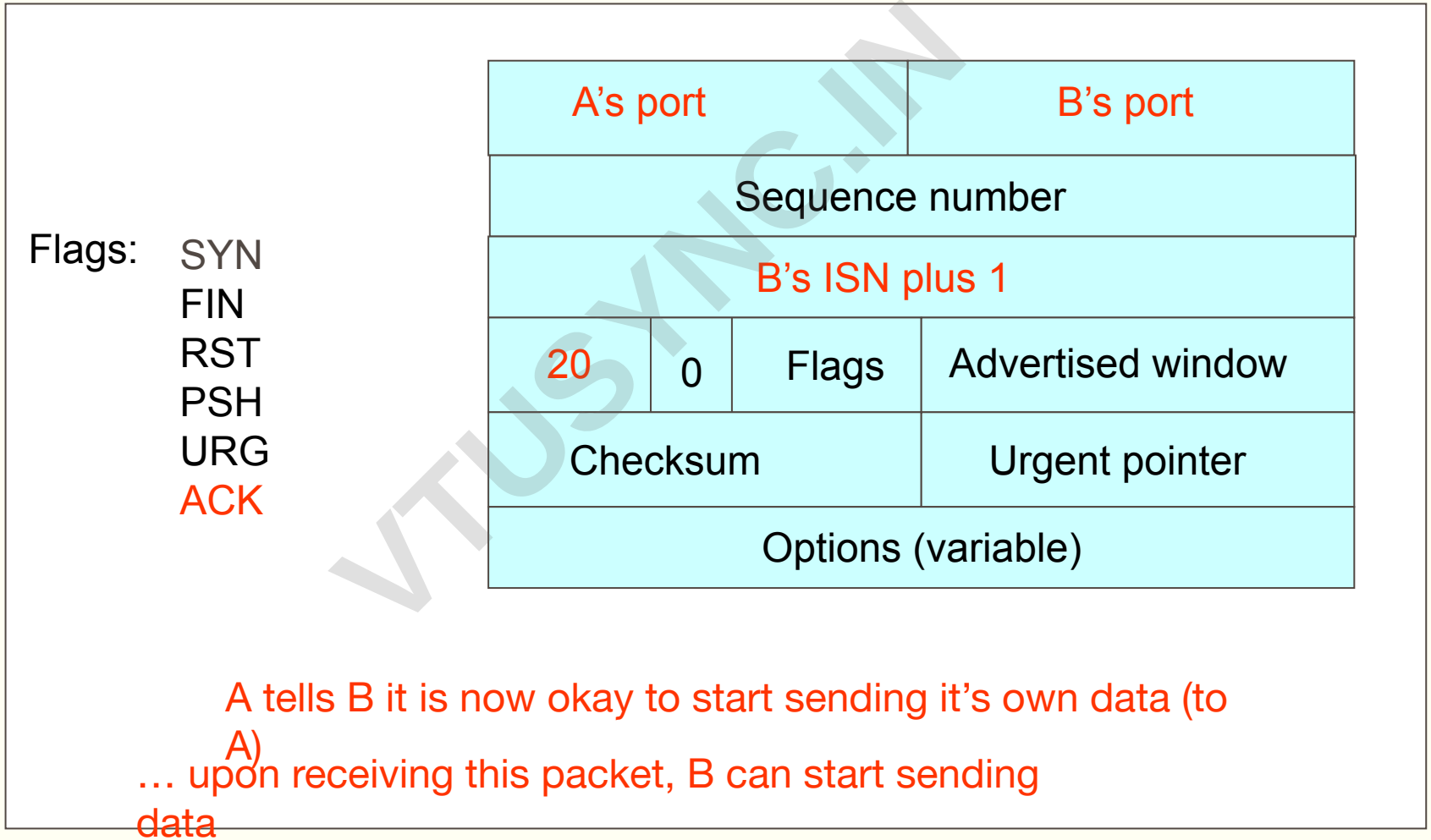
Step 3: A's ACK of the SYN-ACK



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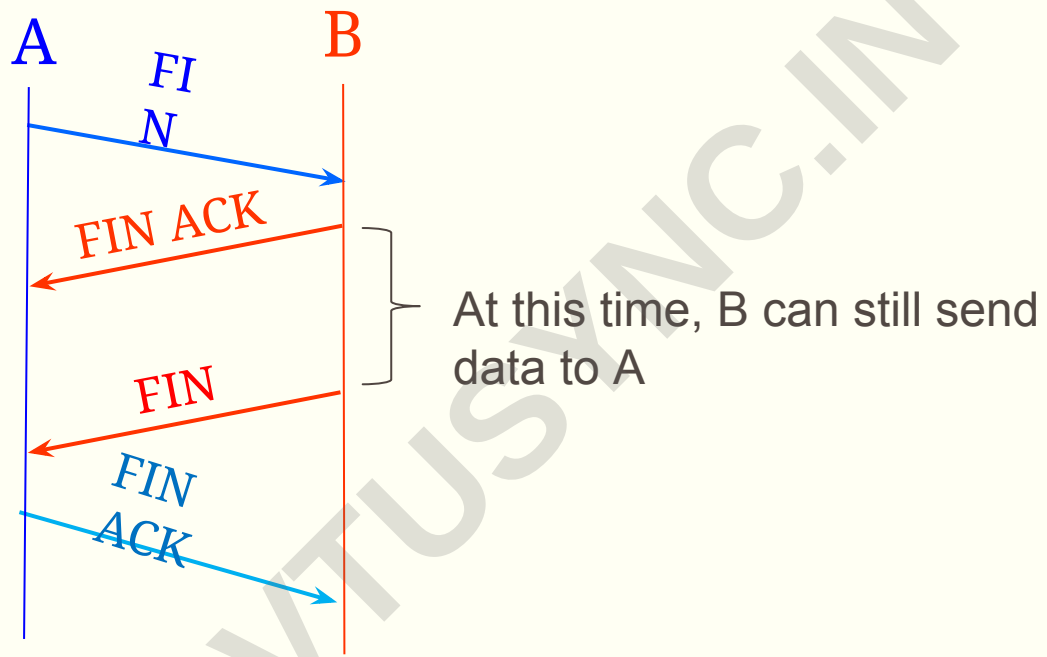
Tearing down a TCP Connection



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- Each end of the data flow must be shut down independently.
- If one end is done, it sends a FIN segment. This means that no more data will be sent.

Error control in TCP

- The Data Link layer implements error control measures like using a CRC Checksum and using some ARQ protocols.
- If every frame is verified, on EVERY link, at the Data Link layer, can something be wrong, end-to-end, when the message reaches the destination host?
- While the Link layer checks take care of the errors that may occur on the various links, they cannot take care of the errors that may occur INSIDE the routers.
- At every router, various fields (TTL, Fragment Offset, etc) will / may change, leading to a change in the Packet Header checksum.

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Flow control in TCP

- The Data Link layer implements flow control measures like using Sliding window protocols.
- What would be the optimum size of a window?
- At the Data Link layer (for a link between adjacent nodes), the window size can be pretty small (1, 2, 4,…) since the Bandwidth-Delay product is low.
- Bandwidth-Delay product broadly indicates how many bits (segments) could have been transmitted by the time a sender receives the acknowledgment of a message segment sent earlier.

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Flow control in TCP

- However, at the Transport Link layer (host to host), the Bandwidth-Delay product will be high because of the high delay required to traverse the entire path to the destination.
- Therefore, the Window size here (at the Transport Layer) has to be much larger.
- The same logic applies to the Buffering at the transmitting host.
  - It has to be much larger than that used at an intermediate router.

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Wireless issues in TCP

- Transport layer protocols such as TCP that implement congestion control should be independent of the underlying Network and Data Link layer technologies.
- But, in practice, there are various issues with wireless links / networks.
- The main issue is that % packet loss is often used as a measure of congestion.
- This % packet loss is normally high on wireless links / networks, due to the higher rate of transmission error (due to noise).

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Wireless issues in TCP

- By the time, this % packet loss reaches around 10%, the connection is effectively stopped because of the throttling by the congestion control mechanisms.
- One solution is to differentiate the packet losses due to transmission error and due to insufficient bandwidth.
- Things will be more complicated when the path contains a few wired links and a few wireless links.

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User Datagram Protocol (UDP)

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User Datagram Protocol (UDP)

- The main characteristics of UDP are as follows
  - UDP implements a connectionless transfer.
    - It does not require connection establishment prior to data transfer
  - UDP service is unreliable
    - UDP does not guarantee the delivery of datagram to the destination
    - It does not guarantee the correct order of the datagrams.
  - UDP computes the checksum for the entire header plus data
  - UDP does not buffer the incoming datagrams.
  - UDP does not have the feature to segment long messages.

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User Datagram Protocol (UDP)

- UDP results in speedier communication because it does not spend time forming a firm connection with the destination before transferring the data.
- User Datagram Protocol (UDP) is used for:
- Sending relatively small amounts of data, eliminating concerns regarding controlling errors or the flow of the packets
- Routing update protocols such as Routing Information Protocol (RIP) and Network control protocols like SNMP.
- Multicasting because UDP works well with packet switching.

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User Datagram Protocol (UDP)

- Real-time applications in which the information needs to be delivered quickly and smoothly.
- It is specifically chosen for time-sensitive applications like gaming, playing videos, or Domain Name System (DNS) lookups (convert Domain names to IP addresses).
- For users, it is better to have the overall transmission arrive on time than wait for it to get there in a near-perfect state.
  - For this reason, UDP is commonly used in Voice over Internet Protocol (VoIP) applications as well.

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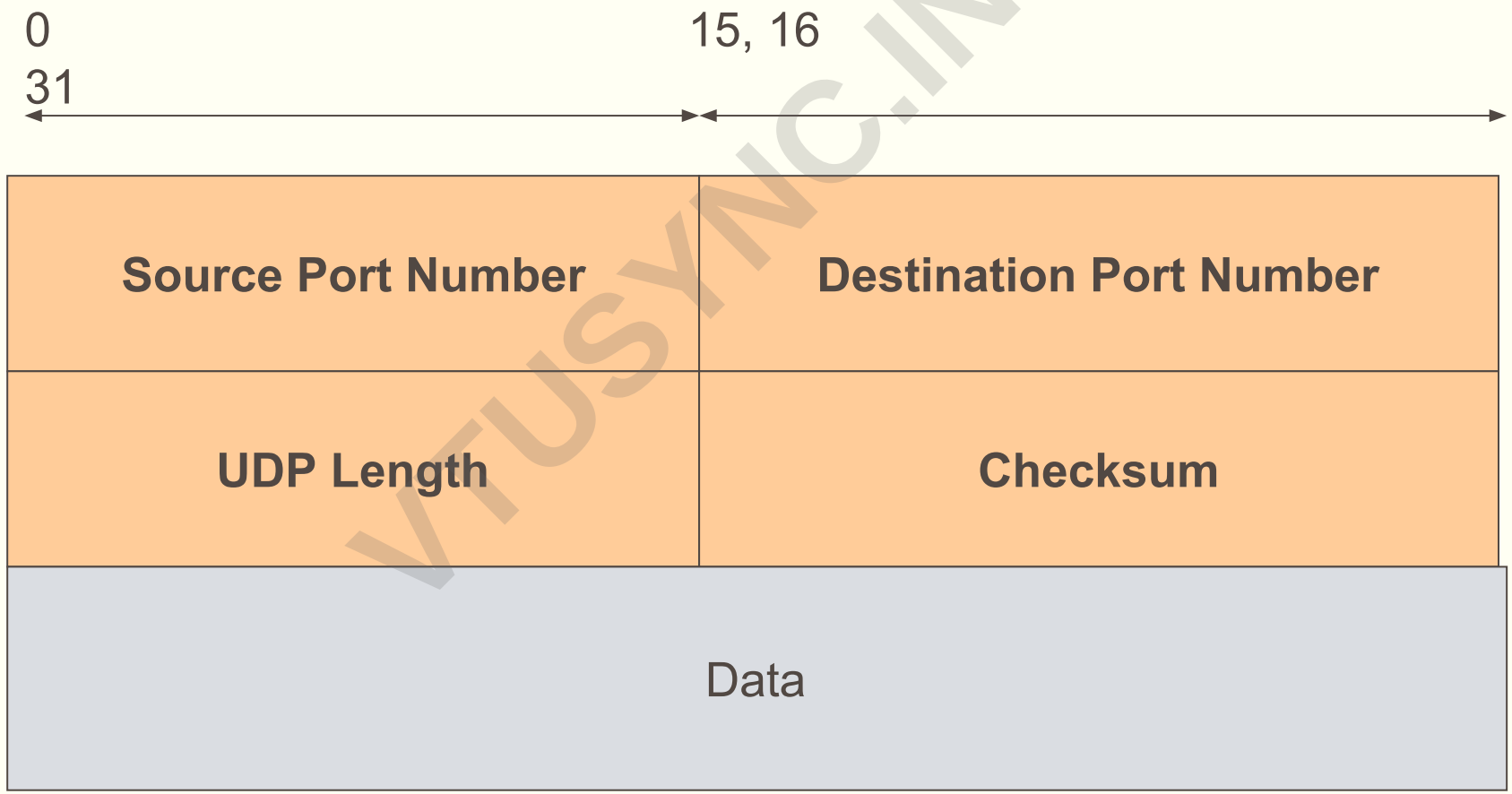
UDP header



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User Datagram Protocol (UDP)

- Source port number identifies the sender's port, when used, and should be assumed to be the port to reply to if needed.
- If not used, it should be zero.
- If the source host is the server, the port number is likely to be a well-known port number from 0 to 1023.
- If the source host is the client, the port number is likely to be an ephemeral (temporary, short-lived) port number.
- In networking, you might use well-known ports (0 to 1023), registered ports (1024 to 49151), and dynamic or ephemeral ports (49152 to 65535), as defined by the Internet Assigned Numbers Authority (IANA).

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User Datagram Protocol (UDP)

- Destination port number identifies the receiver's port and is required.
- UDP Length field specifies the length in bytes of the UDP header and data.
  - The minimum length is 8 bytes, the length of the header.
  - The field size sets a theoretical limit of 65,535 bytes (8-byte header + 65,527 bytes of data) for a UDP datagram.
- The Checksum field may be used for error-checking of the header and data (by the Application layer); The field carries all-zeros if unused.

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Congestion control

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Congestion control

- **Congestion control** has to ensure that the subnet is able to carry the traffic offered by the hosts.
  - It is a global issue, involving all hosts, all routers and all links in the network.
- When too many packets are present in a small part of the subnet, it leads to congestion.
  - Then, the data carrying capacity of the subnet reduces.
- When the number of packets dumped onto the subnet by the hosts is within the carrying capacity, they are all delivered (except the ones with transmission errors)
  - The number delivered is proportional to the number sent by the hosts

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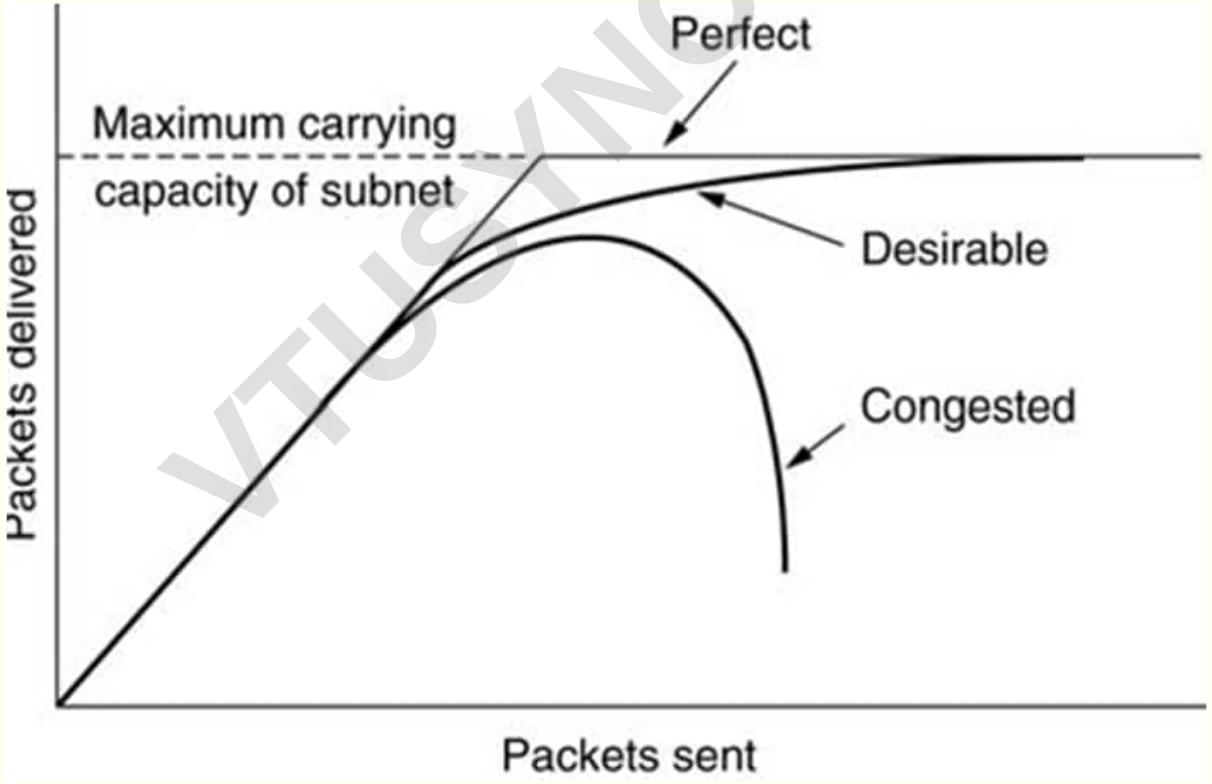
Congestion control

- As traffic increases beyond the capacity, the network cannot cope and packets are lost.



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Causes of congestion

- Load greater than the resources in a part of the network
  - More traffic and low bandwidth lines
- Insufficient buffer at the routers
  - Having a very large buffer is counter-productive (why?)
- Slow processors at the routers

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Measures of congestion

- Percentage of packets discarded for lack of buffer space
- Average queue length at the routers
- Average packet delay
- Standard deviation of the packet delay
- Number of packets that are timed-out and retransmitted

In all the above cases, a rising number indicates growing congestion.

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Congestion control algorithms

- In spite of the routers adapting to the traffic conditions, and changes in topology (adaptive routing), traffic can build up in a part of the network.
- The basic principle behind all the Congestion Control algorithms is: Reduce load and / or Increase capacity.
  - Often, it is difficult to increase capacity temporarily and at short notice.
  - Therefore, most of the algorithms concentrate on reducing (regulating) the load.

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Congestion control algorithms

- Common techniques:
- **Admission Control techniques:**
  - Leaky bucket algorithm
  - Token bucket algorithm
- **Traffic throttling techniques:**
  - Choke packet algorithm
  - Explicit Congestion Notification
  - Load shedding

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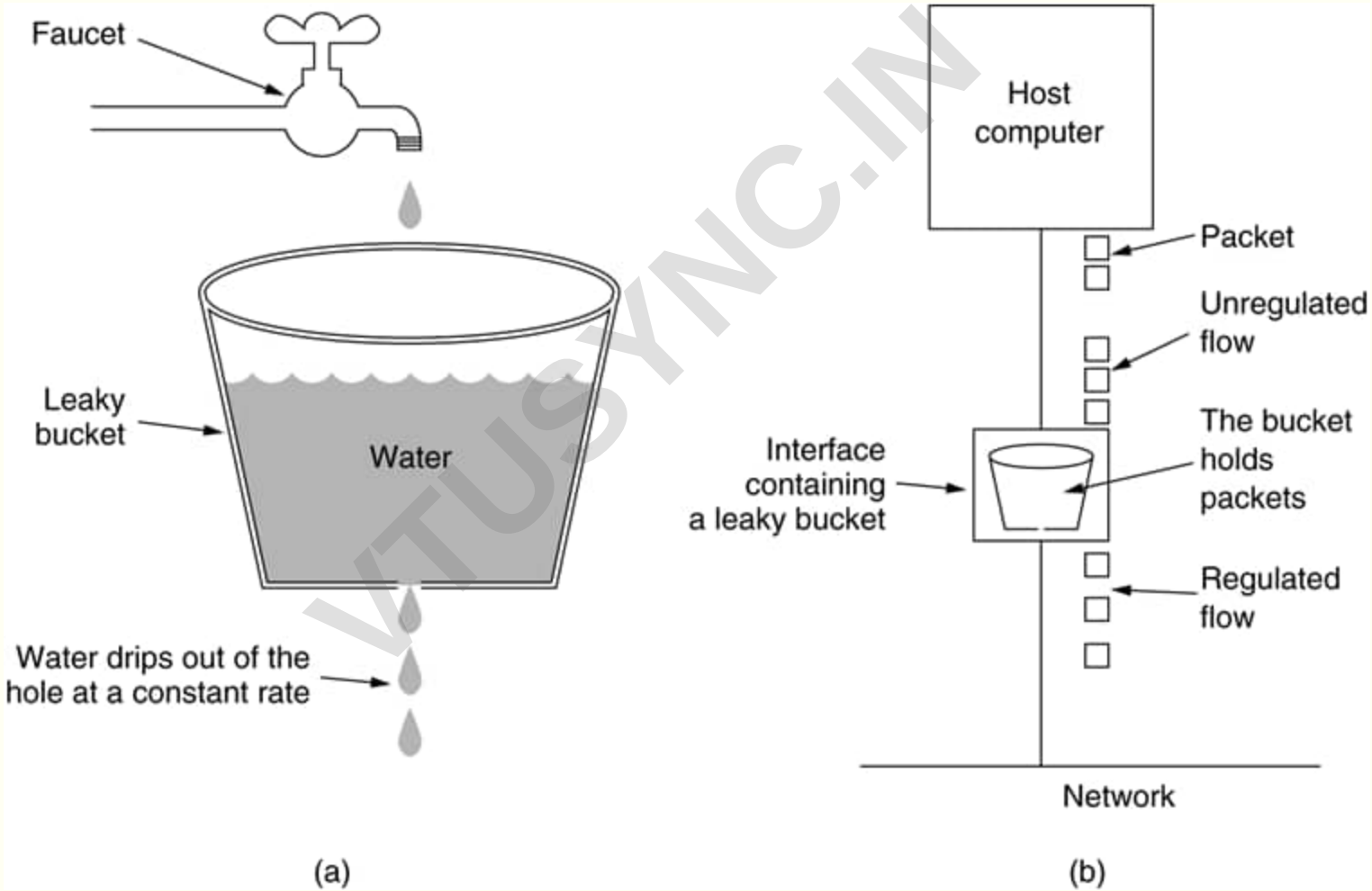
Leaky Bucket algorithm



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Leaky Bucket algorithm

- The leaky bucket tries to ensure that the number of packets put out on to the network (on an average) is constant, regardless of the burstiness of the input; It does nothing when input is idle or the rate is below the defined constant rate.
- The host injects one packet per clock tick onto the network. This results in a uniform flow of packets, smoothing out bursts and reducing congestion.
- It is implemented as a single-server queue with constant service time.
  - If the bucket (buffer) overflows, then packets are discarded.

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Leaky Bucket algorithm

- When packets are of variable length, it is better to allow a fixed number of bytes per tick.
- The leaky bucket algorithm imposes the same restriction on all hosts (some sending almost continuously and some sending occasionally).
  - In short, it does not allow any burstiness beyond the constant output rate that has been defined.

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Token Bucket algorithm

- The Token Bucket algorithm allows the output rate to vary, depending on the size of the burst.
- In the TB algorithm, the bucket holds tokens.
  - Each token entitles the host to transmit a fixed number of bytes.
  - To transmit a packet, the host must capture and destroy the required number of tokens (depending on the packet size).
- Tokens are generated by a clock at the rate of one token every Δt sec.
- Idle hosts can capture and save up tokens (up to the max. size of the bucket) in order to send larger bursts later.
  - When the bucket fills up, new tokens are lost.

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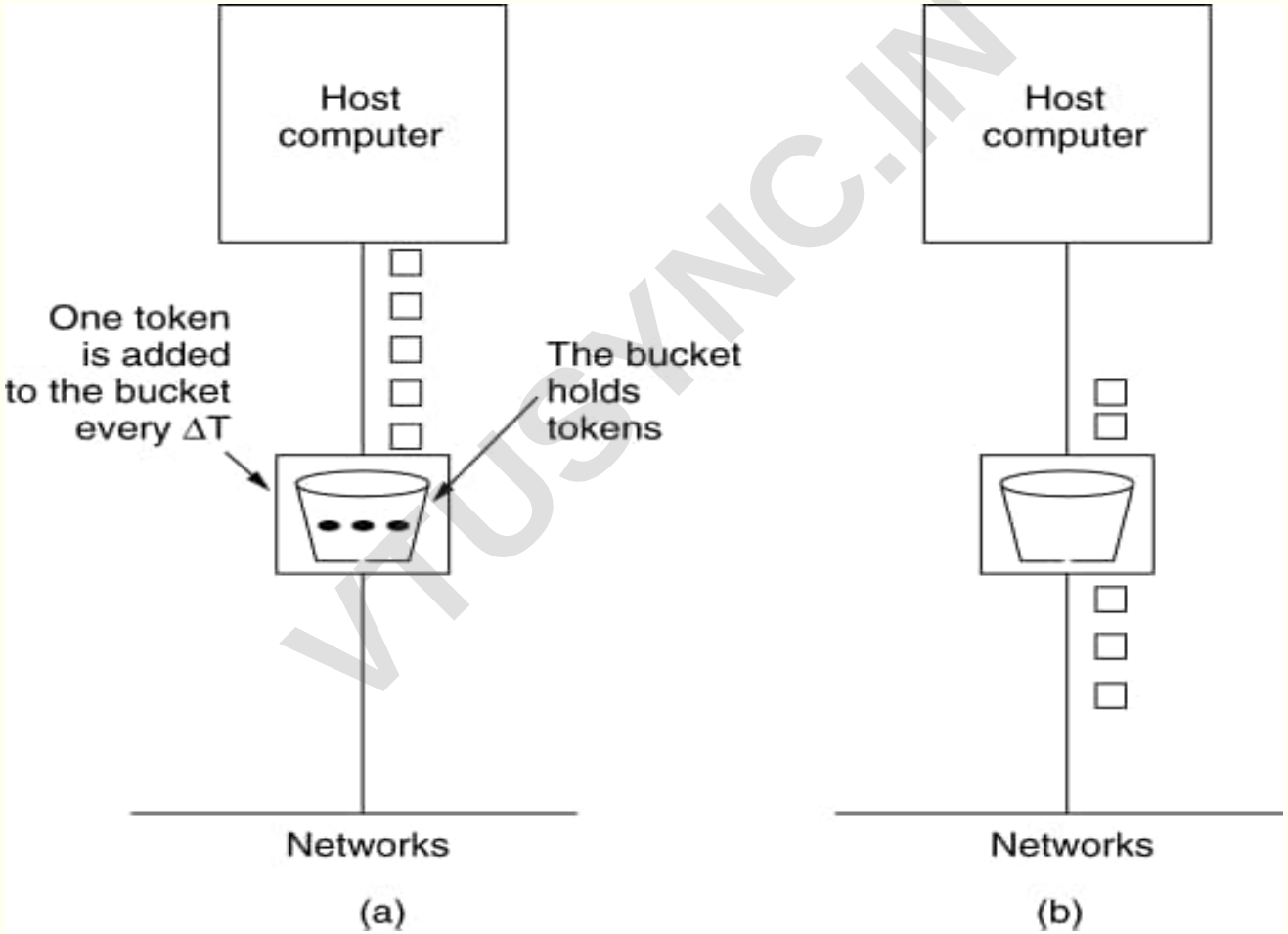
Token Bucket algorithm



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Choke Packet algorithm

- When a router is congested or about to get congested, it starts sending choke packets to the sources from where it received packets.
- A choke packet is a control packet generated at a congested node.
- The source, on receiving the choke packet must reduce its transmission rate by a certain percentage.
- This is a direct way of telling the sources to slow down temporarily.

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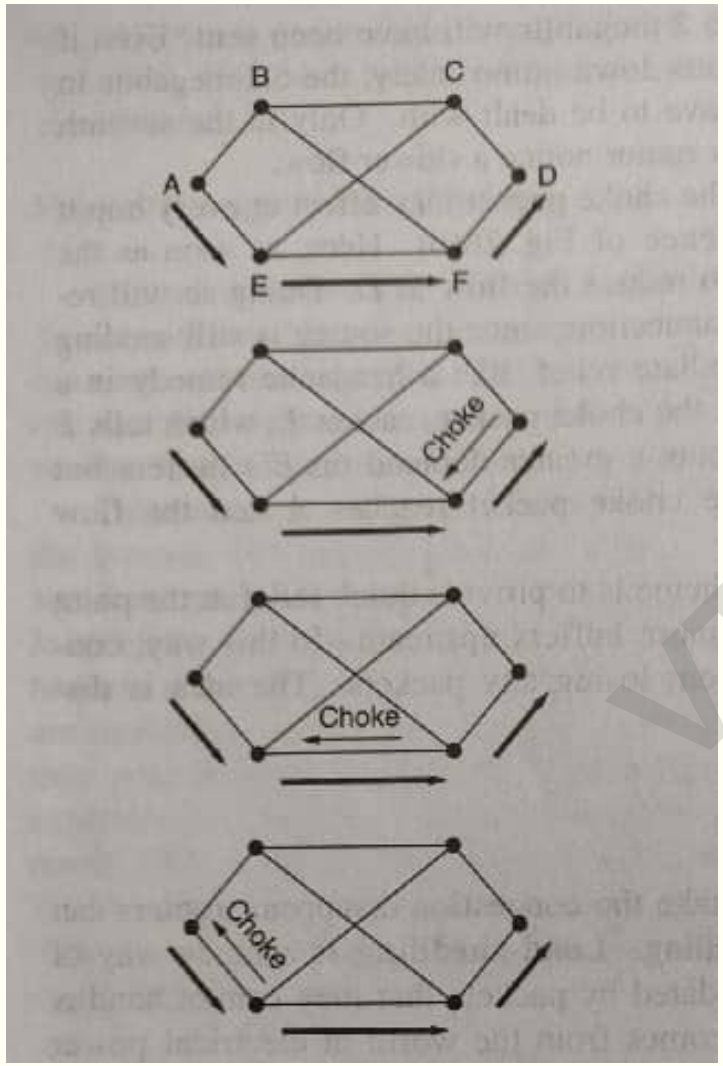
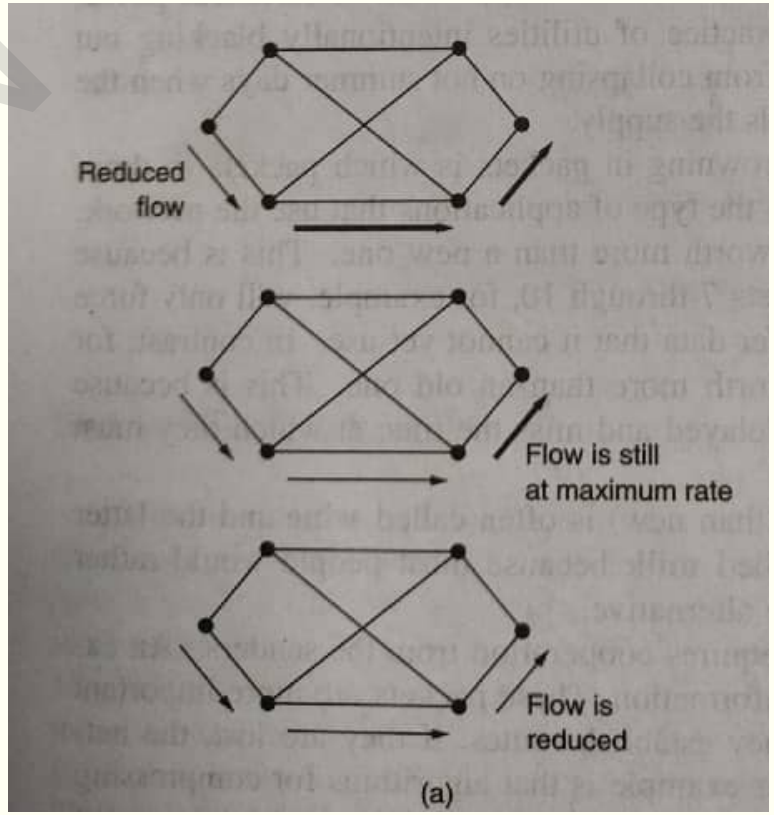
Choke Packet algorithm





D is the congested router, and A is the source of the congestion

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Hop-by-hop Choke Packet algorithm

- Over long distances or at high speeds, choke packets are not very effective.
- A more efficient method is to send to choke packets hop-by-hop.
- This requires each hop to reduce its transmission even before the choke packet arrive at the source.

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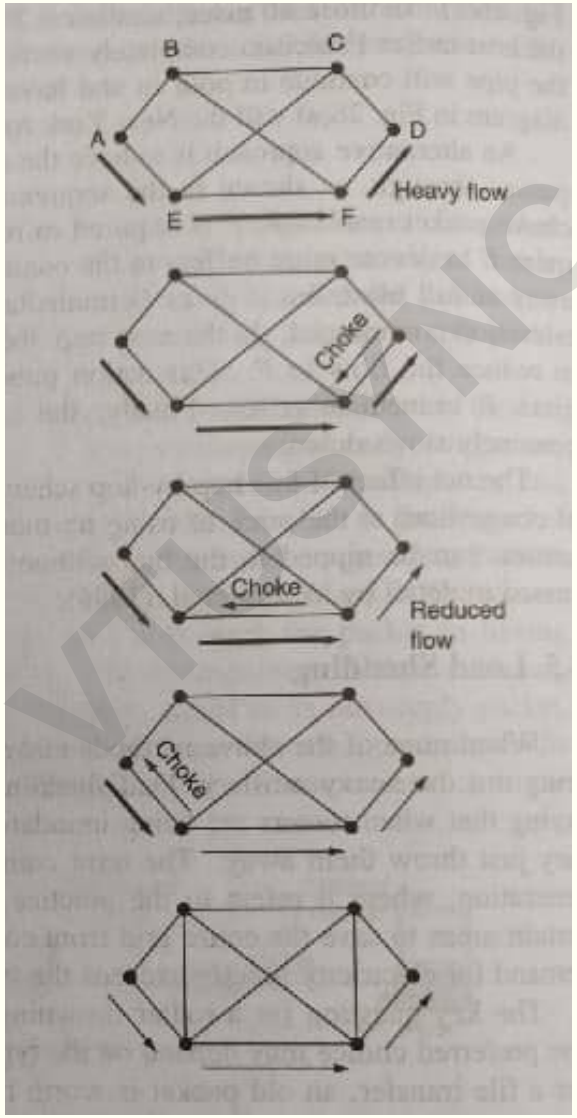
Hop-by-hop Choke Packet algorithm



- This technique will spread the congestion backwards – To the routers along the way.
- It is better to have many routers ‘mildly’ congested than having one router congested ‘heavily’

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Explicit Congestion Notification (ECN)

- Instead of generating additional (choke) packets to warn of congestion, a router can tag any packet it forwards (by setting a bit in the packet's header) to signal that it is experiencing congestion.
- When the packet is delivered, the destination notes that there is congestion and informs the sender when it sends a Reply / Acknowledgement packet.
  - The sender can then throttle its transmissions, as before.
- This method is called Explicit Congestion Notification (ECN).

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Load shedding

- This method is the last resort, when none of the milder methods, described earlier, are able to control congestion.
- Load shedding just means that the congested router will throw away packets to reduce the load, when the buffer is filled up / or about to get filled up.
- The key question here is about which packets to discard; The answer depends on the type of application to which the packets belong.
  - For a File Transfer, an old packet is worth more than a new one. Dropping packet 6 and keeping packets 7 to 10 (for example) will only force the receiver to do more work to buffer the data that it cannot yet use.
  - In contrast, in real-time applications, a new packet is worth more than an old one, because packets become useless if they are delayed and miss the time when they must be played out.

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Load shedding – Random Early Detection (R.E.D.)

- Dealing with congestion when it first starts is more effective than dealing with it when it gets worse.
- Random Early Detection is a type of Load Shedding that discards packets before the buffer space is filled up.
- This idea stems from the fact that most hosts on the Internet do not get congestion signals in the form of ECN.
  - The only reliable indication of congestion that they get is Packet loss.
- To determine when to start discarding, routers maintain a running average of their queue length.

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Load shedding – Random Early Detection (R.E.D.)

- When the average queue length on some link exceeds a threshold, that particular link is said to be congested, and a small fraction of the packets (coming on that link) are dropped at random.
- Picking packets at random makes it more likely that the fastest senders will see more packet drops.
  - This is the best option since a router cannot tell which of the sources (sending packets on that link) is contributing more to the congestion.
- The affected sender will notice the loss, and its Transport layer protocol will slow down.
  - The (higher-than-normal) lost packets are thus delivering the same message as a choke packet, but implicitly.

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