CONGESTION CONTROL AND QUALITY OF SERVICE

Congestion control and quality of service are two issues so closely bound together that improving one means improving the other and ignoring one usually means ignoring the other. These are issues related not to one layer, but to three: the data link layer, the network layer, and the transport layer.

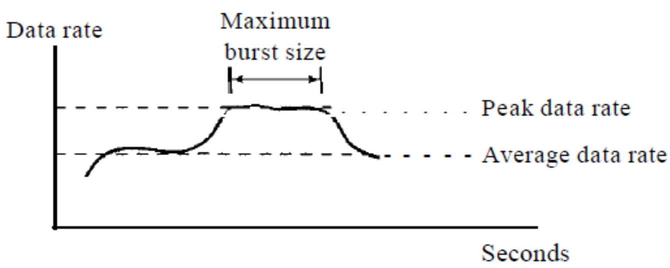
DATA TRAFFIC

The main focus of congestion control and quality of service is data traffic. In congestion control we try to avoid traffic congestion. In quality of service, we try to create an appropriate environment for the traffic.

Traffic Descriptor

Traffic descriptors are qualitative values that represent a data flow. Figure 24.1 shows a traffic flow with some of these values.

Figure 24.1 *Traffic descriptors*



- ***Average Data Rate***
  The average data rate is the number of bits sent during a period of time, divided by the number of seconds in that period. We use the following equation:

  Average data rate = amount of data **/** time

  The average data rate is a very useful characteristic of traffic because it indicates the average bandwidth needed by the traffic.

- ***Peak Data Rate***
  The peak data rate defines the maximum data rate of the traffic. The peak data rate is a very important measurement because it indicates the peak bandwidth that the network needs for traffic to pass through without changing its data flow.

- ***Maximum Burst Size***
  The maximum burst size normally refers to the maximum length of time the traffic is generated at the peak rate.

  Although the peak data rate is a critical value for the network, it can usually be ignored if the duration of the peak value is very short. For example, if data are flowing steadily at the rate of 1 Mbps with a sudden peak data rate of 2 Mbps for just 1 ms, the network probably can handle the situation. However, if the peak data rate lasts 60 ms, there may be a problem for the network.

- ***Effective Bandwidth***
  The effective bandwidth is the bandwidth that the network needs to allocate for the flow of traffic. The effective bandwidth is a function of three values: average data rate, peak data rate, and maximum burst size.

TRAFFIC PROFILES

A data flow can have one of the following traffic profiles: constant bit rate, variable bit rate, or bursty.

- ***Constant Bit Rate***
  A constant-bit-rate (CBR), or a fixed-rate, traffic model has a data rate that does not change. In this type of flow, the average data rate and the peak data rate are the same. The maximum burst size is not applicable. This type of traffic is very easy for a network to handle since it is predictable. The network knows in advance how much bandwidth to allocate for this type of flow.

- ***Variable Bit Rate***
  In the variable-bit-rate (VBR) category, the rate of the data flow changes in time, with the changes smooth instead of sudden and sharp. In this type of flow, the average data rate and the peak data rate are different. The maximum burst size is usually a small value. This type of traffic is more difficult to handle than constant-bit-rate traffic.

- ***Bursty***
  In the **bursty data** category, the data rate changes suddenly in a very short time. It may jump from zero, for example, to 1 Mbps in a few microseconds and vice versa. It may also remain at this value for a while. The average bit rate and the peak bit rate are very different values in this type of flow. The maximum burst size is significant. This is the most difficult type of traffic for a network to handle because the profile is very unpredictable. To handle this type of traffic, the network normally needs to reshape it, using reshaping techniques. Bursty traffic is one of the main causes of congestion in a network.

CONGESTION

Congestion in a network may occur if the **load** on the network-the number of packets sent to the network-is greater than the *capacity* of the network-the number of packets a network can handle.

Congestion in a network or internetwork occurs because routers and switches have queues-buffers that hold the packets before and after processing. A router, for example, has an input queue and an output queue for each interface. When a packet arrives at the incoming interface, it undergoes three steps before departing.

1. The packet is put at the end of the input queue while waiting to be checked.

2. The processing module of the router removes the packet from the input queue once it reaches the front of the queue and uses its routing table and the destination address to find the route.

3. The packet is put in the appropriate output queue and waits its turn to be sent.

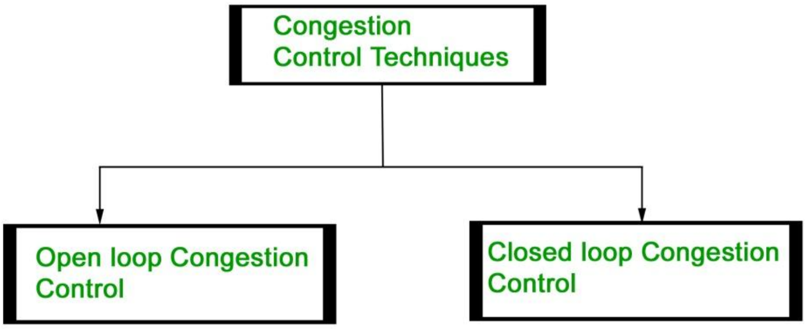
Two issues arise here:

a) First, if the rate of packet arrival is higher than the packet processing rate, the input queues become longer and longer.

b) Second, if the packet departure rate is less than the packet processing rate, the output queues become longer and longer.

CONGESTION CONTROL

Congestion control refers to techniques and mechanisms that can either prevent congestion, before it happens, or remove congestion, after it has happened. In general, we can divide congestion control mechanisms into two broad categories: open-loop congestion control (prevention) and closed-loop congestion control (removal).



- **Open Loop Congestion Control**

Open loop congestion control policies are applied to prevent congestion before it happens. The congestion control is handled either by the source or the destination.

Policies adopted by open loop congestion control

- **Retransmission Policy :** It is the policy in which retransmission of the packets are taken care. If the sender feels that a sent packet is lost or corrupted, the packet needs to be retransmitted. This transmission may increase the congestion in the network.

  To prevent congestion, retransmission timers must be designed to prevent congestion and also able to optimize efficiency.

- **Window Policy :** The type of window at the sender side may also affect the congestion. Several packets in the Go-back-n window are resent, although some packets may be received successfully at the receiver side. This duplication may increase the congestion in the network and making it worse. Therefore, Selective repeat window should be adopted as it sends the specific packet that may have been lost.

- **Discarding Policy :** A good discarding policy adopted by the routers is that the routers may prevent congestion and at the same time partially discards the corrupted or less sensitive package and also able to maintain the quality of a message.

  In case of audio file transmission, routers can discard less sensitive packets to prevent congestion and also maintain the quality of the audio file.

- **Acknowledgment Policy :** Since acknowledgement are also the part of the load in network, the acknowledgment policy imposed by the receiver may also affect congestion. Several approaches can be used to prevent congestion related to acknowledgment. The receiver should send acknowledgement for N packets rather than sending acknowledgement for a single packet. The receiver should send a acknowledgment only if it has to sent a packet or a timer expires.

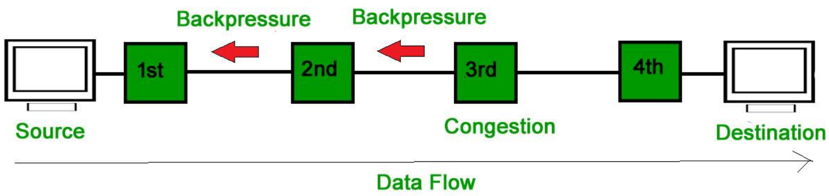
- **Admission Policy :** In admission policy a mechanism should be used to prevent congestion. Switches in a flow should first check the resource requirement of a network flow before transmitting it further. If there is a chance of a congestion or there is a congestion in the network, router should deny establishing a virtual network connection to prevent further congestion.

All the above policies are adopted to prevent congestion before it happens in the network.

- **Closed Loop Congestion Control**

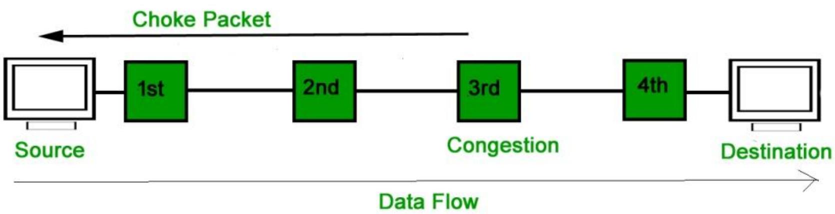
Closed loop congestion control technique is used to treat or alleviate congestion after it happens. Several techniques are used by different protocols; some of them are:

- **Backpressure :** Backpressure is a technique in which a congested node stop receiving packet from upstream node. This may cause the upstream node or nodes to become congested and rejects receiving data from above nodes. Backpressure is a node-to-node congestion control technique that propagates in the opposite direction of data flow. The backpressure technique can be applied only to virtual circuit where each node has information of its above upstream node.



In above diagram the 3rd node is congested and stops receiving packets as a result 2nd node may be get congested due to slowing down of the output data flow. Similarly 1st node may get congested and informs the source to slow down.

- **Choke Packet Technique :** Choke packet technique is applicable to both virtual networks as well as datagram subnets. A choke packet is a packet sent by a node to the source to inform it of congestion. Each router monitor its resources and the utilization at each of its output lines. whenever the resource utilization exceeds the threshold value which is set by the administrator, the router directly sends a choke packet to the source giving it a feedback to reduce the traffic. The intermediate nodes through which the packets has traveled are not warned about congestion.



- **Implicit Signaling :** In implicit signaling, there is no communication between the congested nodes and the source. The source guesses that there is congestion in a network. For example when sender sends several packets and there is no acknowledgment for a while, one assumption is that there is a congestion.

- **Explicit Signaling :** In explicit signaling, if a node experiences congestion it can explicitly sends a packet to the source or destination to inform about congestion. The difference between choke packet and explicit signaling is that the signal is included in the packets that carry data rather than creating different packet as in case of choke packet technique.

- Explicit signaling can occur in either forward or backward direction.
  - **Forward Signaling :** In forward signaling signal is sent in the direction of the congestion. The destination is warned about congestion. The receiver in this case adopt policies, such as slowing down the acknowledgments, to prevent further congestion.

  - **Backward Signaling :** In backward signaling signal is sent in the opposite direction of the congestion. The source is warned about congestion and it needs to slow down to avoid the discarding of packets.

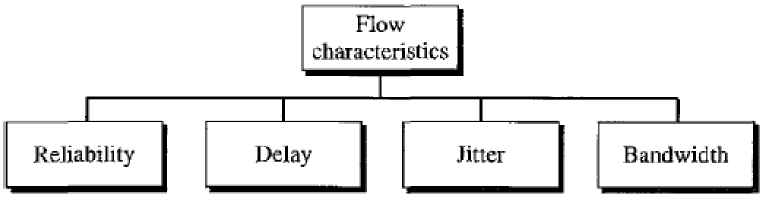
QUALITY OF SERVICE

Quality of service (QoS) is an overall performance measure of the CN.

FLOW CHARACTERISTICS

Four types of characteristics are attributed to a flow: reliability, delay, jitter, and bandwidth.

Figure 24.15 *Flow characteristics*



- ***Reliability***
  Reliability is a characteristic that a flow needs. Lack of reliability means losing a packet or acknowledgment, which entails retransmission. However, the sensitivity of application programs to reliability is not the same. For example, it is more important that electronic mail, file transfer, and Internet access have reliable transmissions than telephony or audio conferencing.

- ***Delay***
  Source-to-destination delay is another flow characteristic. Again applications can tolerate delay in different degrees. In this case, telephony, audio conferencing, video conferencing, and remote log-in need minimum delay, while delay in file transfer or e-mail is less important.

- ***Jitter***
  Jitter is the variation in delay for packets belonging to the same flow. For example, if four packets depart at times 0, 1, 2, 3 and arrive at 20, 21, 22, 23, all have the same delay, 20 units of time. On the other hand, if the above four packets arrive at 21, 23, 21, and 28, they will have different delays: 21,22, 19, and 24.

  For applications such as audio and video, the delay is not acceptable.

  Jitter is defined as the variation in the packet delay. High jitter means the difference between delays is large; low jitter means the variation is small. If the jitter is high, some action is needed in order to use the received data.

- ***Bandwidth***
  Different applications need different bandwidths. In video conferencing we need to send millions of bits per second to refresh a color screen while the total number of bits in an e-mail may not reach even a million.

TECHNIQUES TO IMPROVE QOS

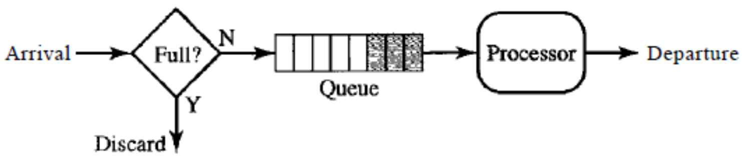
Some techniques that can be used to improve the quality of service. Four common methods: scheduling, traffic shaping, admission control, and resource reservation.

1) Scheduling

Packets from different flows arrive at a switch or router for processing. A good scheduling technique treats the different flows in a fair and appropriate manner. Several scheduling techniques are designed to improve the quality of service.

- ***FIFO Queuing***
  In first-in, first-out (FIFO) queuing, packets wait in a buffer (queue) until the node (router or switch) is ready to process them. If the average arrival rate is higher than the average processing rate, the queue will fill up and new packets will be discarded.

Figure 24.16 *FIFO queue*



- ***Priority Queuing***
  In priority queuing, packets are first assigned to a priority class. Each priority class has its own queue. The packets in the highest-priority queue are processed first. Packets in the lowest-priority queue are processed last.

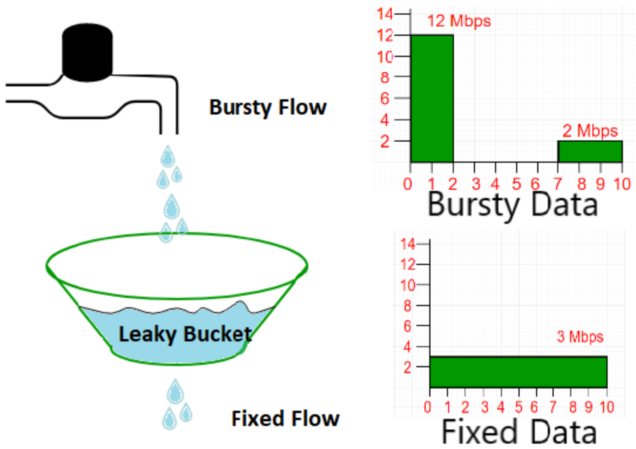
  A priority queue can provide better QoS than the FIFO queue because higher priority traffic, such as multimedia, can reach the destination with less delay. However, there is a potential drawback. If there is a continuous flow in a high-priority queue, the packets in the lower-priority queues will never have a chance to be processed. This is a condition called *starvation.*

- ***Weighted Fair Queuing***
  A better scheduling method is weighted fair queuing. In this technique, the packets are still assigned to different classes and admitted to different queues. The queues, however, are weighted based on the priority of the queues; higher priority means a higher weight. The system processes packets in each queue in a round-robin fashion with the number of packets selected from each queue based on the corresponding weight.

  For example, if the weights are 3, 2, and 1, three packets are processed from the first queue, two from the second queue, and one from the third queue. If the system does not impose priority on the classes, all weights can be equal. In this way, we have fair queuing with priority.

2) Traffic Shaping
Traffic shaping is a mechanism to control the amount and the rate of the traffic sent to the network. Two techniques can shape traffic: leaky bucket and token bucket.

- ***Leaky Bucket***
  If a bucket has a small hole at the bottom, the water leaks from the bucket at a constant rate as long as there is water in the bucket. The rate at which the water leaks does not depend on the rate at which the water is input to the bucket unless the bucket is empty. The input rate can vary, but the output rate remains constant. Similarly, in networking, a technique called leaky bucket can smooth out bursty traffic. Bursty chunks are stored in the bucket and sent out at an average rate.

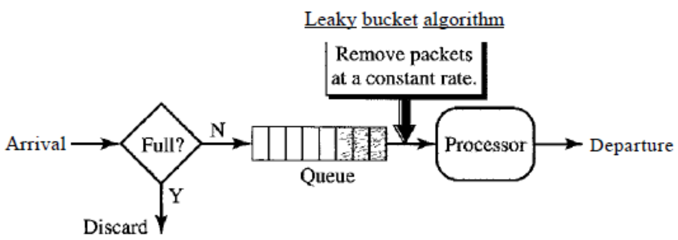


For e.g, the network has committed a bandwidth of 3 Mbps for a host. The use of the leaky bucket shapes the input traffic the host sends a burst of data at a rate of 12 Mbps for 2 s, for a total of 24 Mbits of data. The host is silent for 5 s and then sends data at a rate of 2 Mbps for 3 s, for a total of 6 Mbits of data. In all, the host has sent 30 Mbits of data in lOs. The leaky bucket smooths the traffic by sending out data at a rate of 3 Mbps during the same 10 s. Without the leaky bucket, the beginning burst may have hurt the network by consuming more bandwidth than is set aside for this host.

A simple leaky bucket algorithm can be implemented using FIFO queue. A FIFO queue holds the packets. If the traffic consists of fixed-size packets (e.g., cells in ATM networks), the process removes a fixed number of packets from the queue at each tick of the clock. If the traffic consists of variable-length packets, the fixed output rate must be based on the number of bytes or bits.

The following is an algorithm for variable-length packets:
1) Initialize a counter to n at the tick of the clock.
2) If n is greater than the size of the packet, send the packet and decrement the counter by the packet size. Repeat this step until n is smaller than the packet size.
3) Reset the counter and go to step 1.

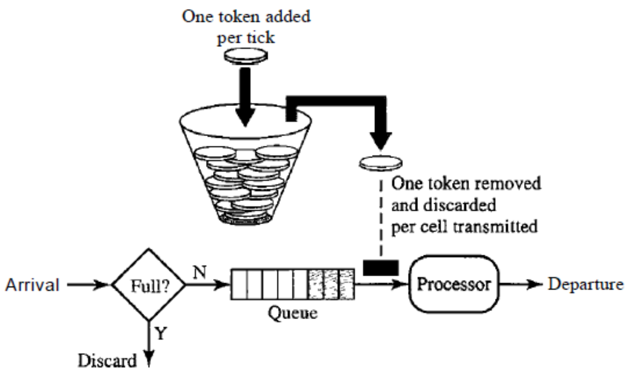
Figure 24.20 *Leaky bucket implementation*



- ***Token Bucket***

The leaky bucket algorithm enforces output pattern at the average rate, no matter how bursty the traffic is. So in order to deal with the bursty traffic we need a flexible algorithm so that the data is not lost. One such algorithm is token bucket algorithm.

Figure 24.21 *Token bucket*



The token bucket algorithm allows idle hosts to accumulate credit for the future in the form of tokens. In regular intervals tokens are thrown into the bucket. The bucket has a maximum capacity. If there is a ready packet, a token is removed from the bucket, and the packet is send. If there is no token in the bucket, the packet cannot be send.

The token bucket can easily be implemented with a counter. The token is initialized to zero. Each time a token is added, the counter is incremented by 1. Each time a unit of data is sent, the counter is decremented by 1. When the counter is zero, the host cannot send data.

3) Resource Reservation
A flow of data needs resources such as a buffer, bandwidth, CPU time, and so on. The quality of service is improved if these resources are reserved beforehand.

4) Admission Control
Admission control refers to the mechanism used by a router, or a switch, to accept or reject a flow based on predefined parameters called flow specifications. Before a router accepts a flow for processing, it checks the flow specifications to see if its capacity (in terms of bandwidth, buffer size, CPU speed, etc.) and its previous commitments to other flows can handle the new flow.