CC Exam Revision

2021 Sem 1

Question 1

Describe two reasons for using layered network protocols. List two disadvantages of using such layered models.

- Advantages:
    - Layering simplifies design, implementation, and testing by partitioning the overall communications process into parts.
    - Layers have their own protocols, which can be changed without affecting other layers. Changes can be made to specific part of the network without having to reimplement the entire network
- Disadvantages:
    - There can be redundancy between layers. For example, the data link and transport layers both have error handling.
    - Independence means if a single part of the model is broken, the entire network will also break, as layers cannot fix other layers.

*Reasons: 1. Using layered protocols leads to breaking up the design problem into smaller, more **manageable** pieces.*
Disadvantages: 1. Processing and data overhead. 2. Multitude of protocols.

Reasons:
1.A layered network protocol is easier to test because each component can be tested inisolation.
2.The protocols within each layer can be updated without affecting other layers.

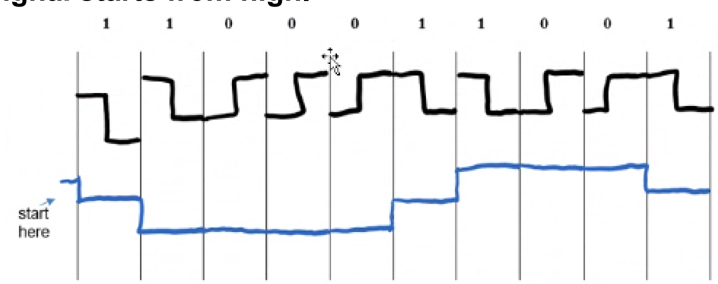
Disadvantages:
1.Each layer adds additional layer-specific overhead to the packets sent over thenetwork that increases latency.
2.Some layers might duplicate services in other layers that might otherwise be shared innon-layered networks (e.g., error control in transport and data link layer).

Networks can be connected by different devices. Describe four (4) such devices and indicate which layer of the OSI model they are in.

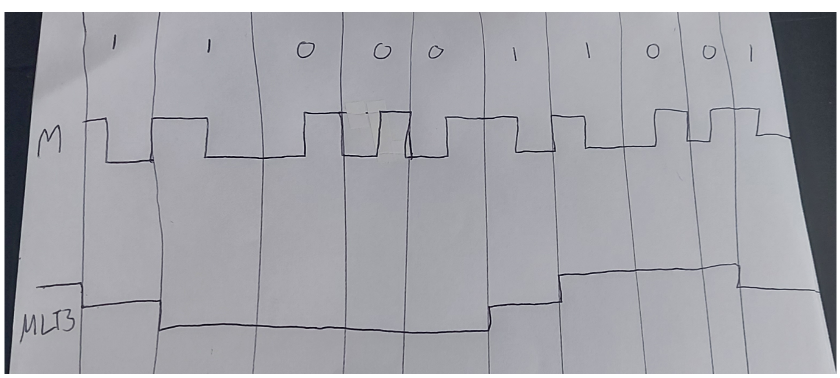
- Hub
    - Physical layer
    - Hubs do not care about what is being sent to them, they simply broadcast everything they receive to all their ports.
- Bridge
    - Data link layer
    - Local traffic stays in its own LAN.

        - Separates networks into their own collision domain
        - If a packet is received with the target address found on port 1, it will only forward the packet to port one, ignoring any other ports.
- Switch
    - Data link layer
    - Each port is a collision domain
        - We have store-and-forward technology.
    - One broadcast domain
    - Only send traffic to the requested device.
- Router
    - Network layer
    - Every branch is a broadcasting domain
    - Uses routing and routing algorithms to device which broadcasting domain a packet should travel to, to reach its destination.

For a bit stream 1100011001, draw its Manchester encoding and MLT-3 encoding if the signal starts from high.



(In most MLT-3 implementation, 1 is indicated by a transition, and 0 is otherwise.)

- 

Name 3 last mile technologies other than NBN. Briefly describe what NBN is and why it is significant to the networking infrastructure.

- Last mile technologies:
    - Dial up
    - ISDN
    - DSL
- NBN:
    - National network of communications built by the government.

- For internet and phone
- Phases out PSTN and ISDN
- Aims to deliver speeds of 50Mbps - 1Gbps, significantly faster than other of the last mile technologies.

Question 2

For a bit stream: 10110011 01101110 11101101 00010011, calculate with detailed steps the checksum at the sender's end and the receiver's end.

Sender's End	Receiver's End
b1 = 10110011 = 179	b1 = 10110011 = 179
b2 = 01101110 = 110	b2 = 01101110 = 110
b3 = 11101101 = 237	b3 = 11101101 = 237
b4 = 00010011 = 19	b4 = 0010011 = 19
x = (b1 + b2 + b3 + b4) mod (2^8-1)	b5 = 11011100 = 220 *(checksum block)*
37 = 545 mod 255	
	x = (b1 + b2 + b3 + b4 + b5) mod 2^8-1
checksum c = -x mod 255	0 = 765 mod 255
c = -35 mod 255	
c = 220	checksum c = -0 mod 255
c = 11011100	c = -0 mod 255
	c = 0
	c = 0

- Sender:
  - 10110011 (1)
  - \+ 01101110 (2)
  - = (1) 00100001
  - \+ 1
  - = 00100010
  - \+ 11101101 (3)
  - = (1) 00001111
  - \+ 1
  - = 00010000
  - \+ 00010011(4)
  - = 00100011
  - Sum = 00100011
  - Checksum = 11011100
- Receiver:
  - If there are no issues, the sum should be identical.
  - Sum = 00100011
  - Sum + checksum (from sender)

        - 00100011 +
          11011100 =
          11111111
        - Complement of result = 00000000
    - Accept, as all 0’s.

Briefly describe the ARP (address resolution protocol), and explain how ARP works in the Ethernet.

- Maps an IP address to a MAC address.
- Uses an ARP program to find a device's MAC address, using the IP address.
- ARP cache: A table of IP addresses with their corresponding MAC
- On ethernet, the ARP broadcasts on LAN to every device, asking if they have a certain IP address. If a device does, it will send back its MAC address. This updates the ARP cache, and will be referred to first if that IP address needs to be mapped to a MAC.

Flag byte with byte stuffing is sometimes used for framing. Can we use just 1 flag byte for both the ending of a frame and the beginning of the next? Why or why not?

- We can use the same flag for the beginning and ending of the frame.
- In fact, this is standard in PPP.
- This flag is the definitive beginning/ending point of a frame, and due to byte stuffing, this flag is guaranteed not to appear in any other place than the start/end.
- By having a separate flag for the beginning and end, we would increase overheads, as every frame would need multiple flags inside of it.

The stop-and-wait protocol is used for flow control between two switching nodes. The one-way propagation delay between the nodes is 10 msec. What should be the transmission data rate of the link in order to achieve a Throughput of 40 frames-per second (fps) if each frame is of 2000 bits? Assume that the time for processing the frame and transmitting acknowledgement frame is negligible, and there are no transmission errors on the link.

- TProp = 10msec = 0.01s
- Throughput = 40fps
- Size of every frame = 2000 bits
- Size of file = 2000 * 40 = 80 000bits
- Rate = 80 000 / 1 - 2*(0.01) (2 way traffic)
- Rate = 81633bits/second

Let us assume that the link rate of R kbps is required.

T_{frame} = 2000 bits / (R kbps) = 2/R sec

T_{prop} = 10 msec =0.01 sec

The throughput in terms of frames per second will be

$1/(T_{frame} + 2T_{prop}) = 1/((2/R+0.01*2) = R/(2+0.02R)$ fps

The desired throughput is 40 fps, Hence we have

Throughput = 40 fps = R/(2+0.02R) fps

i.e., 80 + 0.8*R = R   or   R = 400 kbps

(Ctrl)

Question 3

(a) An IT company has been allocated all the IP addresses in the range of 193.27.10.XXX, and needs to create 3 subnets. The first subnet needs to be twice the size of the other 2.
(i) What is the IP address range of each subnet?

- Subnet 1: 193.27.10.1 - 193.27.10.126
- Subnet 2: 193.27.10.129 - 193.27.10.190
- Subnet 3: 193.27.10.193 - 193.27.10.254

(ii) Write the subnet masks for the following IP addresses in CIDR notation, and state which subnet they are in.

− 193.27.10.123
- 193.27.10.123/25
- Subnet 1

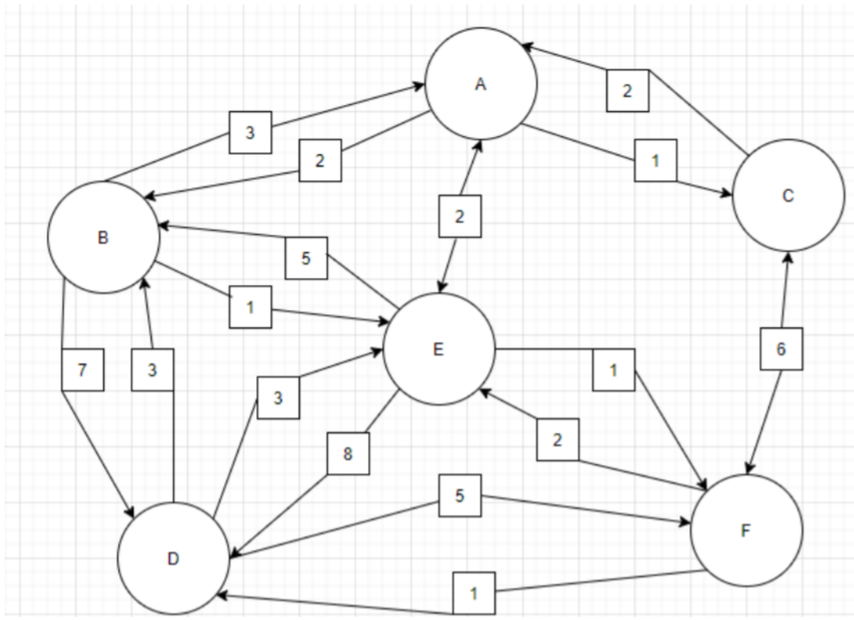
− 193.27.10.210
- 193.27.10.210/26
- Subnet 3

List two (2) similarities and two (2) differences between the Distance-Vector Routing and the Link-State Routing protocols

- Link-State and Distance Vector:
  - They are both routing algorithms, which try to find the most efficient route to transport to a node.
  - Assumes the routers know the address for each neighbor, and the cost of reaching each neighbour.
- Link State:
  - A node tells every other node in the network its distance to its neighbours
  - Dijkstra's algorithm is used.
- Distance Vector:
  - A node tells its neighbours its distance to every other node in the network.
  - Bellman-Ford algorithm is used

Draw the topology of the network described in the following link state table:

	A		B		C		D		E		F	
links	B	2	A	3	A	2	B	3	A	2	C	6
links	C	1	D	7	F	6	E	3	B	5	D	1
links	E	2	E	1	-	-	F	5	D	8	E	2
links	-	-	-	-	-	-	-	-	F	1	-	-



Thanks, phil

Use the Bellman-Ford algorithm to determine, with detailed steps, the shortest path from A to D and the cost of this path.

	A	B	C	D	E	F
1 (A)	0x	-	-	-	-	-
2 (C)	0x	2	1x	-	2	-
3 (B)	0x	2x	1x	-	2	7
4 (E)	0x	2x	1x	9	2x	7
5 (F)	0x	2x	1x	9	2x	3x
6 (D)	0x	2x	1x	4	2x	3x

Finished
Final route: A -> E -> F -> D

h	$L_h(B)$	Path	$L_h(C)$	Path	$L_h(D)$	Path	$L_h(E)$	Path	$L_h(F)$	Path
0	∞	-	∞	-	∞	-	∞	-	∞	-
1	2	A-B	1	A-C	∞	-	2	A-E	∞	-
2	2	A-B	1	A-C	9	A-B-D	2	A-E	3	A-E-F
3	2	A-B	1	A-C	4	A-E-F-D	2	A-E	3	A-E-F
4	2	A-B	1	A-C	4	A-E-F-D	2	A-E	3	A-E-F

Question 4

For the transmission of real-time audio or video streams, should TCP or UDP be used? Justify your answer.

- UDP: Unreliable delivery of packets
    - Connectionless protocol
    - No retransmissions, or acks (or flow control, error control)
- Why?
    - Lower overhead; faster transmissions
    - Useful in client-server situations (eg DNS)
- When we are steaming audio or video, we do not need a perfect transmission, but instead prioritise speed. Sometimes we experience small transmission errors when streaming a movie, however this largely goes unnoticed in these circumstances.
- Furthermore, TCP is unicast and needs a direct connection. Spotify has millions of concurrent users, so using TCP in this case would cause extremely large overheads and worse performance.

Describe the operation of Jacobson's algorithm for timer management in TCP

- RTT variance estimation.
- We can constantly adjust the retransmission time based on continuous measurements of the network performance.
- This algorithm is generally used in TCP
- For every connection, determine the best retransmission timer by using an estimate of RTT
- Use mean deviation and exponential averaging

One of the objectives of both the Data Link Layer and the Transport Layer is flow control. Discuss in detail the reasons behind this service being necessary at both layers, and include in your discussion the differences between the services in the two unconnected layers.

Hint: You should discuss the protocol for achieving this in the TCP/IP model.

- Data Link:
    - It is a set of measures taken to regulate the amount of data that a sender sends so that a fast sender does not overwhelm a slow receiver. In the data link layer, flow control restricts the number of frames the sender can send before it waits for an acknowledgment from the receiver.
    - https://www.tutorialspoint.com/flow-control-in-data-link-layer
    - Flow control is done through the ARQ or stop-and-wait protocols.
- Transport:
    - Limits the amount of data being transmitted so that the destination host can cope.
    - The source may send frames / packets at a rate that is faster than the processing speed of the destination host. (Can lead to a buffer overflow!)
    - The buffer at the destination could be full because
        - Higher level protocol is not ready
        - Outgoing IO port is not ready
        - Destination protocols cannot process PDU as fast
    - Flow control done through sliding window methods or variable sized sliding window

Question 5

The Domain Name System (DNS) can be used for mapping host names to IP addresses.
(i) Describe the hierarchical organisation of the DNS name space. You can use a diagram if you wish. Why should the DNS name space be divided into smaller non-overlapping zones?

- Name space:
    - DNS uses a hierarchical, domain based naming scheme to identify resources on the internet.
    - At the very top are a small number of domains that encompass the entire internet
    - Root domains: int, com, edu, gov, org, net ect…'
    - Case insensitive
    - Component can be up to 63 characters (full path cannot be more than 255)
    - Name follows the organization boundary rather than the physical network.
    - We need to prevent overlapping to prevent any ambiguity. A .com name space for example should be completely unambious in its destination.

(i) Assuming that a recursive query method is used, describe the action taken at a name server when it receives a domain name query.

- Whenever a query for a hostname is received
- If managed by the server: return the matched RR (authoritative record)
- If it exists on the cache, return the cached RR (non-authoritative record)
- Othersie, send a query to the DNS server

(i) In a DNS resource record, what is the time-to-live field for?

- How long the RR is kept in the cache before querying again

In BitTorrent protocol, a tracker is used to track all peers participating in the torrent. Does the existence of a tracker make BitTorrent a hybrid system, rather than peer P2P? Why or why not? Briefly describe how data is exchanged between peers in a torrent.

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A World Wide Web server is usually set up to receive small messages from its client but to transmit potentially very large files for them. Explain then which type of TCP flow control protocol, the go-back-N or the selective repeat, would provide less burden to a particularly popular WWW server.

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2020 Sem 2

I'm tired, frustrated, and most importantly bored.

Question 1

Given the following parameters for a switching network:
N = number of hops between two given end systems
L = message length in bytes
B = data rate in megabits per sec (Mbps) on all links
P = fixed packet size in bytes
H = overhead (header) in bits per packet
D = propagation delay per hop in seconds
R = processing delay for routing decisions at each node in seconds
S = call setup time (for circuit switching or virtual circuit) in seconds
Assume that there are no acknowledgements and ignore all other processing delays at the nodes.

i) Compute the end-to-end delay for Packet Switching.

- Number of packets = L/(P-H) =Np
- (Np + N-1)(P/B) + N*D

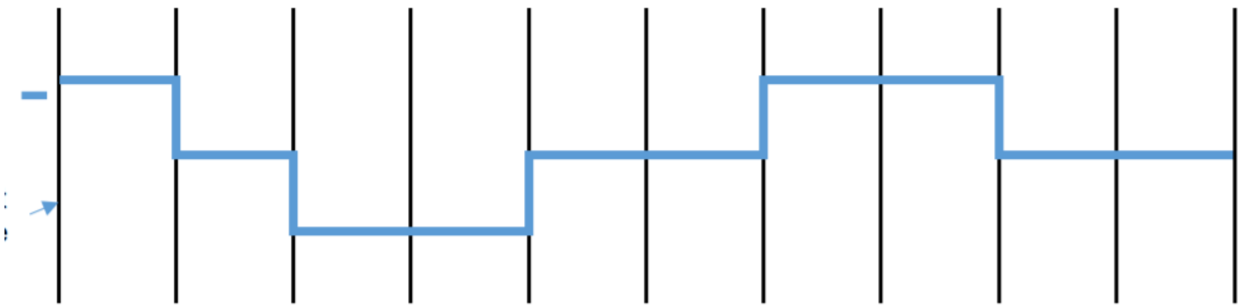
ii) **Compute the end-to-end delay for Virtual Circuit Packet Switching (VCPS).**

- (Np + N-1)(P/B) + N*D + S

iii) **Under what conditions is VCPS faster than Packet Switching? You can use the derivations obtained in i) and ii) to explain your answer**

- When the cost of the setup / overheads are outweighed by the increased processing speed

b. Consider the following MLT-3 encoded signal.



Assume 0 is indicated by a transition, and 1 is otherwise.

i) Write down the bits encoded in the signals.

(2 marks)

ii) Why does MLT-3 require less bandwidth than other encoding techniques? Explain your answer with an example.

- i) 0110101010
- ii) Signal rate 1/4th bit rate due to signal shape

Explain the advantages and disadvantages of using millimetre waves in 5G.

- From the less used band in the frequency spectrum. These waves support having a massive MIMO antenna since the relationship between the wave frequency and antenna size is inversely proportional.
- Higher frequency waves are more likely to be physically obstructed

Question 2

For a bit stream: 100110000110,

i) Derive the codeword being delivered if two-dimensional parity check (odd-parity, block size: 4) is used.

- 1001 | 0
- 1000 | 1
- 0110 | 0
- --------

  0111 | 1

ii) Show an example of error that the derived codeword fails to detect.

- 0011 | 0
- 0010 | 1
- 0110 | 0
- --------

  0111 | 1

Given a bit stream 1011000, or data polynomial D(x) = x^6 + x^4 + x^3 and given the generator
polynomial G(x) = x^2 + 1,
i) Find the codeword C(x) using Cyclic Redundancy Check (CRC).

- 1011000
  G(x) = 0000101

ii) Assume the received message H(x) is H(x) = C(x) + E(x), where E(x) is the error polynomial.
Show that H(x) is not divisible by G(x) when E(x) = x3 +1.

-

Show the byte-stuffing & destuffing steps for the following data bits if PPP frame is used.
01000001 01111101 01000010 01111110 01010000 01110000 01000110

- Convert to Hex : 41 7D 42 7E 50 70 46
- Flag = 7E (Add this to start and end of string)
- Any 7E or 7D in the string will be replaced with 7D 5D or 7D 5E
- Final Stuffing:
    - 7E 41 7D 5E 42 7D 5E 50 70 46 7E
- Destuffing: Any 5E string is XORed with the next digit
    - 7E 41 7D 42 7E 50 70 46 7E

A data link is transmitting frames of 1,500 bits with a data rate of 4,000 bits/second. The propagation delay is 0.2 second. Assume that acknowledgment packets are of negligible size, processing time at the hosts is negligible, and the link is error-free. Derive the minimum window size which will allow full utilization of the link if a sliding window protocol is used for flow control

- Frame size = L = 1500 bits
- Tprop = 0.20s
- Tframe = 1500/4000 = 0.375s
- a = 0.20 / 0.375 = 0.533

- In order to have full link utilisation, W >= 2a + 1
- Wmin = 2 x 0.533 + 1 = 2.06

Question 3

A router has the following CIDR entries in its routing table:

Address/mask	Next hop
129.47.104.0/15	Interface 0
129.44.112.0/21	Interface 1
190.34.100.0/22	Loopback Interface 0
129.44.192.0/19	Router 1
default Router	Router 2

For each of the following IP addresses, what does the router do if a packet with the following destination address arrives?

i) 129.46.120.10

-

ii) 129.44.199.14

-

iii) 129.44.221.255

-

iv) 255.255.255.255

- Broadcast to all on the local network

v) 190.34.104.255

- D

Assume that delay is used as the metric in a distance vector routing. A network consists of seven (7) nodes, Node A to Node G. Node C has three neighbour nodes: B, D, and E. The following vectors have been received by
Node C:
From B: (6, 0, 5, 9, 8, 3, 3);
From D: (7, 6, 1, 0, 8, 9, 4);
From E: (3, 7, 3, 6, 0, 3, 4) where the 1st element from Node X represents the delay from Node X to Node A, and the 2nd element represents the delay from Node X to Node B, and so on.
The measured delays from Node C to B, D and E, are 4, 2, and 5, respectively. What is C's new routing table? Both the next hop and the expected delay need to be specified in the routing table

- Old Routing Table:

Node C	To A	To B	To C	To D	To E	To F	To G
From B	6	0	5	9	8	3	3
From D	7	6	1	0	8	9	4
From E	3	7	3	6	0	3	4

- New Routing Table:

Node C	To A	To B	To C	To D	To E	To F	To G
From B	6	0	5	9	8	3	3
From D	7	6	1	0	8	9	4
From E	3	7	3	6	0	3	4
From C	8 (E)	4	3 (D)	2	5	7 (B)	6 (D)

Question 4

In computer networking, the Network Layer can ensure correct delivery of data from source to destination (end-to-end communication). Why is the Transport Layer still needed? Describe 3 reasons for having the Transport Layer.

- Provides communication services for a process in a host to a process in another host
- Heart of whole protocol hierarchy
- Generally reliable data transmission
- Cost effective data transport
- Independent of physical networks.
- End to end communication for individual applications

Discuss why the Transmission Control Protocol (TCP) uses a “four-way handshake” to close a connection. Is it completely foolproof? Why or why not?

- The TCP handshake is used to ensure that the sender is ready to send data, and the receiver is ready to receive data.

When a user clicks on a hyperlink (e.g. http://www.curtin.edu.au), the web server carries out a series of actions. Describe in detailed steps the operations of a Web server before and after it receives a file request from a remote host.

- After:
    - The web address is sent to the server
    - The server checks if it knows the IP address of the web address (if it is stored in the cache). If it does not know, it will query the DNS Server
    - Once the IP is retrieved, return it back to the user.

In blockchain applications such as bitcoin, explain how immutability is achieved i.e., the data cannot be altered by any of the users in the network.

- Proof of work and ledgers

2020 Sem 1

I don’t care. Fail me. I just want this over with

Random Questions

c. Please explain how BitTorrent addresses the free riding problem?

- Free riding is when a user obtains a file from the network without contributing with it’s transmission. BitTorrent uses a Tit for Tat strategy. Where essentially if you don’t contribute to sharing the file, others won’t share the file with you. BitTorrent tracks the top 4 sharing nodes and those are the ones who will receive the file, while other users are “choked” (do not receive any file chunks). Every 10 seconds the Top 4 is reevaluated and every 30 secs a random peer is selected to send chunks to (optimistically unchoke) and the peer is given the chance to enter the top 4. The more you give the more you get.

Compare the IPv6 header with IPv4 header. List four major differences.

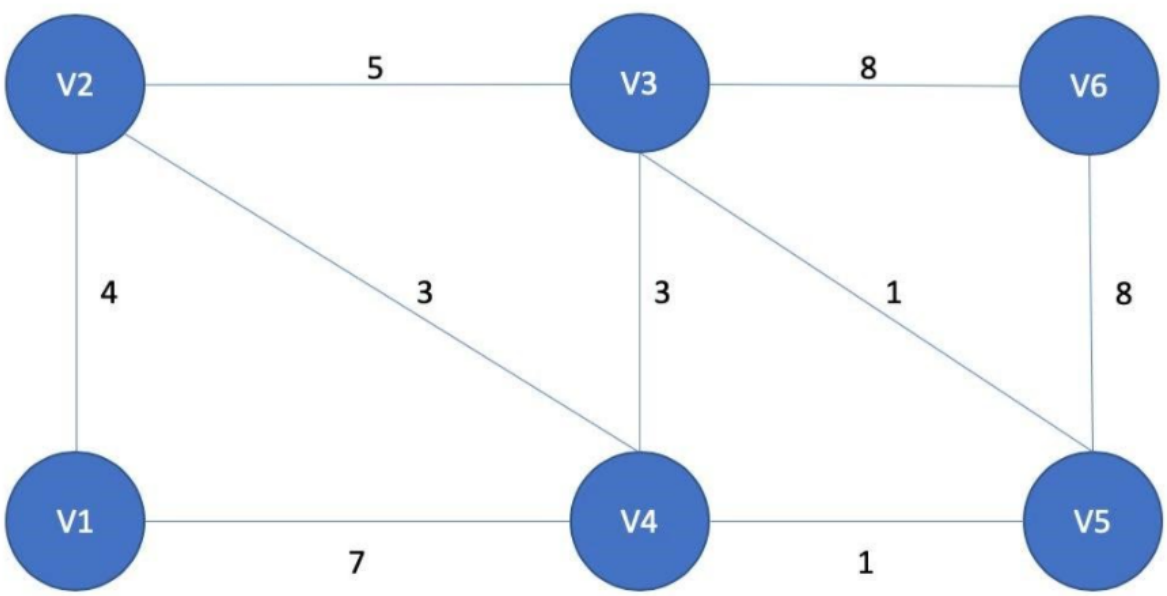
- The size of the IPv6 header is much larger than that of the IPv4 header due to IPv6’s address size.
- IPv6 header does not have fields such as Internet Header Length (IHL), identification flags, fragment offset, header checksum, options and padding.
- A Flow label field was added for QoS.
- IPv6’s header is fixed at 40 bytes while IPv4 has a header anywhere between 20-60 bytes.

2019 Sem 2

Screaming noises

Question 1

Refer to the figure below. Use Dijkstra’s algorithm to generate a least-cost route for V1 to all other nodes.



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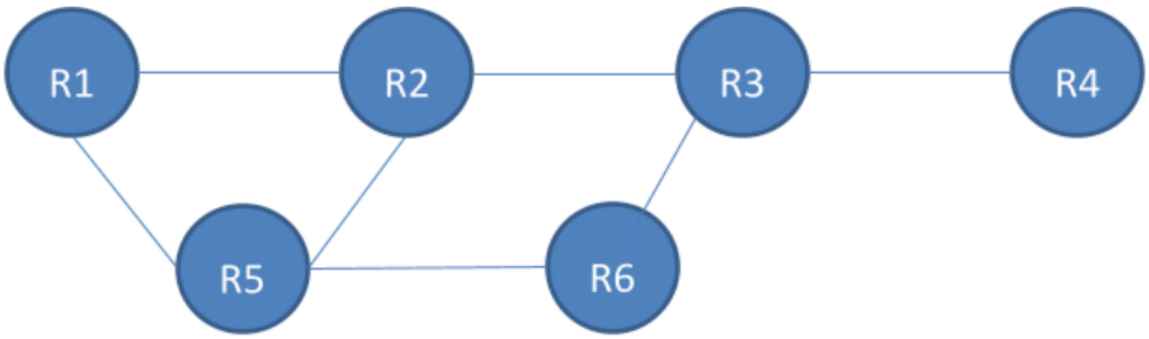
#	n	L(V	PAT	L(V	PAT	L(V	PAT	L(V	PAT	L(V6	PATH

		2)	H	3)	H	4)	H	5)	H	)	
1	{1}	4	1-2	INF	-	7	1-4	INF	-	INF	-
2	{1,2}	4	1-2	9	1-2-3	7	1-4	INF	-	INF	-
3	{1,2,4}	4	1-2	9	1-2-3	7	1-4	8	1-4-5	INF	-
4	{1,2,3,4}	4	1-2	9	1-2-3	7	1-4	8	1-4-5	17	1-2-3-6
5	{1,2,3,4,5}	4	1-2	9	1-2-3	7	1-4	8	1-4-5	16	1-4-5-6
6	{1,2,3,4,5,6}	4	1-2	9	1-2-3	7	1-4	8	1-4-5	16	1-4-5-6

Destination	Cost	Hop
V2	4	-
V4	7	-
V3	9	V2
V5	8	V4
V6	16	V4

Refer to the diagram below. The link from router R3 to router R4 fails, explaining how this may cause the "count to infinity" problem in a distance vector routing protocol. Illustrate your answer with the routing tables in the affected routers.



- Routers R2 or R6 send an update to R3 before R3 sends an update to them(R3-R4 failed).
- When the link failed and R3 didn't send its updated table to R2/R6 before R2/R6 send its result to R3.
- If R2/R6 Send its router table first,then R3 will look at R2/R6 table and think it can reach R4 by R2/R6 because inside R2/R6 routing table, it is still thinking that it can reach R4 through R4(because R3 Didn't sent its newly updated routing table in time).
- R3 will then update its table again and change the distance cost by adding the distance cost to R2/R6 and the distance cost of R2/R6 to R4.After the newly calculated distance cost is calculated it will update its table and send it to R2 and R6.
- And once R6/R2 receive the new updated it will think cost from R3 to R4 have increased and thus updating its table and send update to its neighbouring nodes.Then again,R3 will see it and update its routing table again and send update again until the distance cost is infinite

Split Horizon

- Split horizon can prevent this by not sending routing information back into the direction it came from. Poison reversing is when a route is advertised but with the metric value of infinity. This is so that all the routers that receive the advertisement know that the route with the infinite metric has failed.

Question 2

Consider the case of transmitting 1000-bit frames over a 1 Mbps link with a delay of 10ms (millisecond). The probability that a single frame is in error is 0.75 What is the maximum link utilization for:

Stop-and-wait flow control?

- With 1Kb frames, for stop and wait when sender sends frame it must wait till it receives ack, so total time before next frame can be sent = 2*10ms = 20ms. Hence 50 frames can be sent in a second: 50*1Kb = 50Kb in one second.
- 50Kb/1Mb *100 = 5% maximum link utilisation

Sliding window with a window size of 24 (If selective reject ARQ is used)?

- W = window size = 24
- Latency = 10ms
- Transmission time of frame = 1ms
- X = latency/transmission time of frame = 10
- Link utilization in sliding window protocol = W/(2x+1) if W<(2x+1), else = 1 if W>(2x+1)
- 2x+1 = 21, W=24, W>2x+1
- Link utilization = 100%

Sliding window with a window size of 25 (If selective reject ARQ is used)?

- W = window size = 25
- Latency = 10ms
- Transmission time of frame = 1ms
- X = latency/transmission time of frame = 10
- Link utilization in sliding window protocol = W/(2x+1) if W<(2x+1), else = 1 if W>(2x+1)
- 2x+1 = 21, W=25, W>2x+1
- Link utilization = 100%

Sliding window with a window size of 1 (If Go-back-N ARQ is used)?

- W = window size = 1
- Latency = 10ms
- Transmission time of frame = 1ms
- X = latency/transmission time of frame = 10
- Link utilization in sliding window protocol = W/(2x+1) if W<(2x+1), else = 1 if W>(2x+1)
- 2x+1 = 21, W=1, W<2x+1
- Link utilization = 1/21 * 100 = 4.76%

Question 3

Describe or define the significance of the following IPv4 addresses:

0.0.0.0 (Explain the significance of this address in the context of routing and using it as a host address)

- Is an unspecified address of “this host” (the address of the machine executing the instruction). In DHCP when a unique address has not yet been determined, 0.0.0.0 is used as the Source IP. In the context of a Router 0.0.0.0 represents the default route. 0.0.0.0 is not assignable to an interface or used as a destination address.

255.255.255.255

- Broadcast on this local network

192.168.1.255

- Broadcast address for the 192.168.1 network

192.168.1.0/24

- Subnet with 8 bits for host

Describe the use and purpose of Choke packets. Provide an example.

- Choke-packets are used in Congestion Detection & Recovery. They are a control packet generated at a congested node, used to restrict traffic flow. Once the source receives the choke packet, the transmission rate will be reduced by a specific percentage.

Describe two (2) major differences between the Choke packet method and the RED method.

- RED - Congestion Avoidance
- Choke Packet Method - Congestion Detection and Recovery.
- Choke packet happens when the buffer is already full.
- RED - Packets are discarded before the buffer is full.

Compare and contrast connection-oriented and connectionless protocols (with examples of each)?

- Connection Orientated:
    - Related to the Telephone service.
    - Preferred due to long and steady communication.
    - Congestion is impossible.
    - Packets follow the same route.
    - Requires bandwidth of high range.
- Connectionless:
    - Related to the postal system.
    - Preferred for bursty communication.
    - Congestion is possible.
    - No guarantee of reliability.
    - Packets do not follow the same route.
    - Requires bandwidth of low range.

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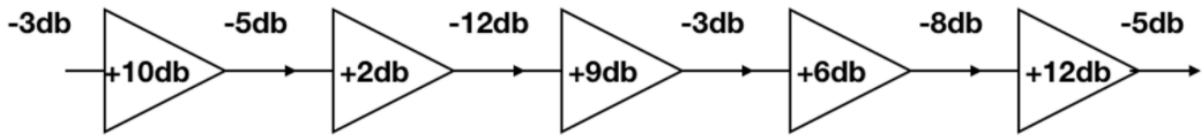
Wow

Random Questions

(b) A communications channel as depicted below, has six (6) segments and five (5) repeaters.



i. Calculate the **overall gain or loss of the system**.

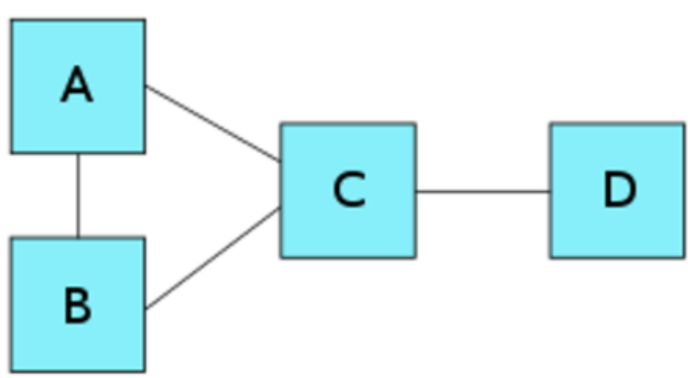
- -3+10-5+2-12+9-3+6-8+12-5 = 3dB

ii. If the required power output is 1 Watt, calculate the required input power of the system in watts.

- Attenuation = 10 log (P1(output)/ P2(input))
- 3 = 10 log (1/P2)
- 10 ^ 0.3 = 1 / P2
- P2 = 0.501 Watt

(c) Refer to the diagram below. The link from router C to router D fails, explain how this may cause the "count to infinity" problem in a distance vector routing protocol. Illustrate your answer with the routing tables in the effected routers.

(10 marks)



- When the connection is lost between Router C and Router D, the connection between them will be changed to infinite steps, as there is no possible way to communicate. When Router A gives C its advertised route, because Router A hasn’t updated its connection to Router D, it will still say that there is still a connection between these two routers. Router C will see this and will assume that Router A has a different connection to Router D, so it will replace the connection from infinite to 2 (pass through Router A and then go to the next router which it is unable to see). Now, when Router C gives its advertised route to Router A, it will show that there is a connection to Router D of 3 steps, meaning that Router A will change its connection to Router D to be 4 steps. This is repeated infinitely and is known as the “count to infinity” problem.