**Q1:**

a. To modify Selective Repeat ARQ so that ACK messages contain a list of the next m frames that it expects to receive, the protocol needs to be modified as follows:

The sender maintains a sliding window that includes all unacknowledged frames, as in the standard Selective Repeat ARQ protocol.

When the receiver receives a frame, it checks to see if it is the next expected frame in the sequence. If it is, the receiver sends an ACK message that contains a list of the next m frames that it expects to receive.

If there are gaps in the received frames, the receiver sends an ACK message that requests retransmission of the missing frames. The sender then retransmits the requested frames.

b. The effect of this change on protocol performance depends on the value of m and the characteristics of the network.

One potential benefit of this modification is improved efficiency, particularly in networks with high latency or high error rates. By including a list of expected frames in each ACK message, the receiver can help reduce the number of unnecessary retransmissions. For example, if the sender knows that the receiver is expecting frames 10-20, it can prioritize those frames for transmission instead of sending other frames that may not be needed.

However, there are also potential drawbacks to this modification. One concern is increased overhead due to the larger size of the ACK messages. Depending on the value of m, the size of each ACK message could be significantly larger than in the standard protocol, which could impact network performance. Additionally, the more frames that are included in each ACK message, the greater the risk of errors in the ACK message itself, which could lead to further retransmissions and delays.

**Q2:**

To determine the smallest possible frame size that allows continuous transmission, we need to calculate the round-trip time (RTT) for a signal to travel from Earth to the Moon and back.

Converting the distance between Earth and the Moon to meters:

375,000 km 1000 m/km = 375,000,000 meters

Plugging in the values:

To achieve continuous transmission on a 1.5 Mbps link with HDLC, we need to calculate the minimum frame size that can be transmitted within this RTT.

The efficiency factor takes into account the protocol overhead, such as header and trailer bits.

Assuming an efficiency factor of 80% (efficiency = 0.8), we get:

Minimum frame size =

To convert to bytes:

Therefore, the smallest possible frame size that allows continuous transmission on a 1.5 Mbps HDLC link to the Moon is approximately 1.41 megabytes.

**Q3:**

a. To meet the requirement of transferring a single real-time telephone voice signal across a packet network with a maximum delay of 20 ms, the following adaptation functions are relevant:

Timing: The timing adaptation function is critical in ensuring that each voice sample is delivered within the required deadline. The network must be able to synchronize its clock with the sender and receiver to maintain the required time intervals between packets.

Reliability and sequencing: To ensure that each voice sample is delivered without loss or misordering, the reliability and sequencing adaptation function is necessary. This requires the use of error detection and correction mechanisms, as well as sequencing and resequencing of packets at the receiver end.

Pacing and flow control: To prevent packet loss due to congestion, pacing and flow control mechanisms are necessary. These mechanisms regulate the rate at which packets are transmitted and received to match the capacity of the network.

Addressing: Addressing is necessary to identify the source and destination of each voice sample. It also enables routing of packets through the network.

b. There are two approaches for meeting the requirements of a real-time telephone voice signal over a packet network: the hop-by-hop approach and the end-to-end approach.

The hop-by-hop approach involves implementing the required adaptation functions at each intermediate node in the packet network. Each node processes the packets it receives before forwarding them to the next node. This approach can introduce additional delays and overhead due to processing at each node. Furthermore, if a node fails, the entire communication may become compromised.

The end-to-end approach involves implementing the required adaptation functions only at the endpoints of the communication path, i.e., the sender and receiver of the voice signal. The packets are transmitted through the network without modification, and any required processing is performed at the endpoints. This approach minimizes delays and overhead, but it may not be suitable for networks with high packet loss rates or variable delays.

In general, the end-to-end approach is preferred for real-time voice communications over packet networks because it minimizes delays and overhead. However, the hop-by-hop approach may be necessary in some situations, such as when the network has high delay or loss rates, or when additional processing is necessary at intermediate nodes.

**Q4:**

a. The answer is 5. Four for the LANs (S and R), and one for the link between the routers (R and R).

b. To support 5 subnets, we need to borrow three bits from the host portion of the IP address.

Explaination:  
We have 5 networks in total, so that if we call N (N > 0) is the number of bits borrowed, then N is the least number that sastifies

Then N = 3.

c. Borrowing three bits creates = 8 subnets.

Since we have borrowed 3 bits, then the subnet mask becomes

(We can understand that the initial subnet mask is /24, after borrowing 3 bits, the subnet mask becomes /27)

All the network addresses are listed below

|  |  |  |
| --- | --- | --- |
| **Network Address** | **Usable Host Range** | **Broadcast Address:** |
| 192.168.1.0 | 192.168.1.1 - 192.168.1.30 | 192.168.1.31 |
| 192.168.1.32 | 192.168.1.33 - 192.168.1.62 | 192.168.1.63 |
| 192.168.1.64 | 192.168.1.65 - 192.168.1.94 | 192.168.1.95 |
| 192.168.1.96 | 192.168.1.97 - 192.168.1.126 | 192.168.1.127 |
| 192.168.1.128 | 192.168.1.129 - 192.168.1.158 | 192.168.1.159 |
| 192.168.1.160 | 192.168.1.161 - 192.168.1.190 | 192.168.1.191 |
| 192.168.1.192 | 192.168.1.193 - 192.168.1.222 | 192.168.1.223 |
| 192.168.1.224 | 192.168.1.225 - 192.168.1.254 | 192.168.1.255 |

d. The last octet of an IP address has 8 bits, since we have borrowed 3 bits, so that the answer is

So that each subnet can create 30 usable hosts.

**Q5:**

a, We define N is the node 4, D1, D2, D3, D5, D6 is the node 1,2,3,5,6.

|  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- |
| Iteration | N | D1 | D2 | D3 | D5 | D6 |
| Initial | {N} | 5 | **1** | 2 | 3 | ~ |
| 1 | {N, D2} | 4, D2 |  | **2** | 3 | ~ |
| 2 | {N, D2, D3} | 4, D2 |  | - | **3** | **3, D3** |
| 3 | {N, D2, D3, D5, D6} | 4, D2 |  |  |  |  |
| 4 | {N, D2, D3, D5, D6, D1} | - | - | - | - | - |

So that, we can conclude that

+ The shortest part from N to D1 is 4, and pass D2.

+ The shortest part from N to D2 is 1.

+ The shortest part from N to D3 is 2.

+ The shortest part from N to D5 is 3.

+ The shortest part from N to D6 is 3, and pass D3.

b, Find the set of associated routing table entries (Destination, Next Hop, Cost)

|  |  |  |
| --- | --- | --- |
| Destination | Cost | Next Hop |
| D1 | 4 | D2 |
| D2 | 1 | D2 |
| D3 | 2 | D3 |
| D5 | 3 | D5 |
| D6 | 3 | D3 |