Problem 1 (50%)

(a) (8%) Describe the functions and operations of the following mechanisms:

   (i) Synchronous multiplexing
   (ii) Statistical multiplexing
   (iii) Fixed assigned multiple access
   (iv) Demand assigned TDMA

(b) (42%) Consider a synchronous TDM device that is used to multiplex packet voice flows across a single output channel. The transmission rate across the outgoing channel is equal to 0.7 Mbps. Each voice packet flow is constructed as follows. The digitized ADPCM voice signal rate is 32 Kbps; the voice waveform is sampled at a rate of 8K samples/sec, and each sample is encoded into 4 bits. A single voice packet consists of samples that are collected over a period of 20 msec. It also contains a 60 bits header. Each voice flow is assumed to be active for an average fractional time that is equal to 20%.

   (i) (12%) Assuming a fixed TDM multiplexer operation, determine the maximum number of voice flows that can be simultaneously transmitted across the channel. Show the design of the system by specifying the frame length and the duration of each time slot (whereby a single slot is used to transport 1 packet).
   (ii) (15%) Assume now that a statistical TDM multiplexing operation is used. Find the average number of voice sources (N) that can be accommodated. Show the design of the system by specifying the frame length and the duration of each time slot.
   (iii) (15%) Consider the system described in (ii). Write a mathematical expression that you will use to compute the probability that an active voice flow will not be able to transmit a packet during a given time frame (finding all the slots to be occupied).

Ans:

(a)

(i) **Synchronous multiplexing**: Sharing communication channel (service) resource among collocated stations (clients) in a dedicated assignment of (time, frequency or code/wavelength) slots to source-destination pair. All messages are held in a common buffer.

(ii) **Statistical multiplexing**: No resources are dedicated to source-destination pairs across the shared channel. Resources (such as time slot or frequency bands) are utilized by a flow only when it is active. All messages are held in a common buffer.

(iii) **Fixed-assigned multiple access**: Sharing communication channel (service) resources among geographically distributed stations (clients), so that each station holds its messages in its own buffer. The assignment of radio resources (either time slots or frequency bands) is fixed, or (slowly) programmable.

(iv) **Demand-assigned multiple access**: The allocations of time slots across the shared channel are dynamically made, based on demand expressed by the station. For example, when a central controller is used, radio resources are allocated by the controller in response to requests made by stations.

(b)

(i)

The packet size is $32 \times 10^3 \times 20^{-3} + 60 = 700$ bits. Hence, the required transmission rate for a video flow is $\frac{700}{20\times10^{-3}} = 35$ Kbps. The number of voice flow can be simultaneously supported is given as:

$$N_{\text{fix}} = \left\lfloor \frac{0.7 \times 10^6}{35 \times 10^3} \right\rfloor = 20. \tag{1}$$

Hence, each time slot can be equal to $\frac{700}{0.7\times10^6} = 1$ msec. And each time frame for N_{fix} users is 20 msec.

(ii)

Packets are transmitted across available time slots under a statistical TDM multiplexing operation. During a frame length, the system can deliver $20\times10^{-3}\times0.7\times10^6 = 14000$ bits. Each packet size is 700 bits. However, each voice flow is assumed to be active for an average fractional time that is equal to 20%. Hence, we obtain the average number of voice sources (N) as follows:

$$N = \left\lfloor \frac{14000}{0.2 \times 700} \right\rfloor = 100. \tag{2}$$

Hence, each time slot can also be equal to $\frac{700}{0.7\times10^6} = 1$ msec. And each time frame for N users is $0.2 \times N \times 1$ msec.

(iii)

When all the slots are occupied, an active voice flow is not be able to transmit a packet during a given time frame. We denote M as the number of active flows at a given time. The probability (denoted as p) that a time slot is occupied by a voice flow is equal to 0.2. Therefore, the probability that a flow finds the system busy is:

$$P(M \geq N_{\text{fix}}) = 1 - \sum_{i=0}^{N_{\text{fix}}-1} P(M = i) = 1 - \sum_{i=0}^{N_{\text{fix}}-1} \binom{N}{i} p^i (1-p)^{N-i}, \tag{3}$$

where N is obtained in (ii).

Problem 2 (50%)

Consider a Half-duplex wireless communications link across which a Stop-and-Wait ARQ error-control scheme is employed. The transmission data rate across the channel is equal to R bps. The radio transceiver has a turn-around time of 0.1 msec. The link is 20 Km long, and the propagation rate is 5 microsec/Km. The ACK packet contains 320 bits. The frame (on which the error control scheme operates) contains 6800 information bits. In addition, it contains 200 overhead bits. The channel's bit error rate is equal to 4×10^{-5}. We neglect the probability of incorrect ACK receptions.

(a) (25%) Design the system by calculating the lowest data rate R that is required to assure (if feasible) a net throughput rate that is no lower than 10 Mbps. Also calculate the normalized throughput efficiency of this design.

(b) (25%) Assume now that it is desirable to increase the net throughput rate of the system designed in (a) to 13 Mbps by improving the modulation/coding scheme that is configured and therefore by reducing the channel's bit error rate. Determine whether it is possible to do so, and if feasible find the highest value that should be permitted for the bit error rate. Also calculate the normalized throughput efficiency attained for the improved system.

Ans: The parameters are summarized as follows:

- The turn-around time $t_{ta} = 0.1 \times 10^{-3}$ sec.
- The propagation delay $t_p = 100 \times 10^{-6} = 10^{-4}$ sec.
- Packet size $N_{pb} = 7000$ bits; ACK size $N_{ab} = 320$ bits.
- The bit error rate (BER) $P_b = 4 \times 10^{-5}$; The packet error rate (PER) $P_E = 1 - (1 - P_b)^{N_{pb}} = 0.2442$.

(a)

The net throughout rate under a stop-and-wait scheme is given by

$$\eta = \frac{N_{db}}{\frac{1}{1-P_E}} \left(\frac{N_{pb}}{R} + \frac{N_{ab}}{R} + 2(t_p + t_{ta}) \right) = \frac{N_{db}(1 - P_E)R}{N_{pb} + N_{ab} + 2R(t_p + t_{ta})} \geq R',$$

where $R' = 10 \times 10^6$ bps is the required lowest net throughput rate. Hence, we have

$$R \geq \frac{R'(N_{pb} + N_{ab})}{N_{db}(1 - P_E) - 2R'(t_p + t_{ta})} = 6.4250 \times 10^7.$$

As a result, the lowest transmission rate is 64.25 Mbps. Then, the normalized throughput efficiency is 15.56%.

(b)

(b) We attempt to decrease BER to meet the net throughput rate (denoted as $R'' = 13$ Mbps) of the system designed in (a) with $R = 64.25$ Mbps. Hence, we have following inequality: inequality:

$$\frac{N_{db}(1-P_b)^{N_{pb}}R}{N_{pb}+N_{ab}+2R(t_p+t_{ta})} \geq R''.$$

Then, we have

$$P_b \leq 1 - \sqrt[N_{pb}]{\frac{R''(N_{pb}+N_{ab}+2R(t_p+t_{ta}))}{N_{db}R}} = 2.5202 \times 10^{-6}.$$

We conclude that it is possible to design a system with net throughput rate that is equal to 20 Mbps. The highest value of BER is 2.5202×10^{-6}. Then, the normalized throughput efficiency is 20.23%.