Example question for Digital Signal Processing for practical take-home test

Questions:

I. *In this section, you will read speech signals into MATLAB, make some observations and then complete the rest of the computations on your answer paper. You have to submit the MATLAB code, speech signals (along with the source from which you obtained the speech signals) and the written answers. Include units of measurement wherever necessary*.

Question 1: Use MATLAB to read five speech signal from your collection (.wav files) – as five separate speech signals. You will have five speech signal variables in MATLAB, name them as – *speech1, speech2, speech3, speech4, speech5*. Let A be the random variable representing the amplitude values of the speech signals that was captured by reading the signals to MATLAB. Clearly state the source from where you obtained the speech signals.

a) **Question 2a:** Define X, which is a function of the random variable A. X is defined as the magnitude of the speech signal. Let the magnitude be defined as $X = abs(20 \log_{10} A)$ dB. (abs means absolute value). Using MATLAB compute the random variable X from A, and save it as *mag_speech1, mag_speech2, mag_speech3, mag_speech4, mag_ speech5.*

b) **Question 2b:** Concatenate the five magnitude of speech signal variables *vertically* into one variable called *mag_speech_signal.* For example, if each of the magnitude of speech signals has a size of 500 by 1, then the concatenated array will have a size of 5×500 by $1 = 2500$ by 1.

c) **Question 2c:** Sort the *mag_speech_signal* variable in ascending order using MATLAB. From the sorted feature array, find the following three values (points) of the speech signal:

Point	**Criteria to choose**
Point 1 x_1	The value at the exact middle element of the *mag_speech_signal* array, if the size (length) *mag_speech_signal* is odd. If it has even length, then take the average of the two elements at the middle of the array.
Point 2 x_2	Element at 20% of the total length of the variable. For this, find the total length of the *mag_speech_signal* array, and find the integer close to 20% of the length, and take the value of *mag_speech_signal* at 20% of the length. (For example, if length is 1001; then 20% of 1001 = 200.2. Hence Point 2 will be the 200th point in the array *mag_speech_signal.*
Point 3 x_3	Point at 80% of the total length of the variable. Similar to the method used to find Point 2, find Point 3 at the 80% point.

Include a table of the three points in your answer sheet like the one shown below:

Point	Random variable X value – Magnitude in dB
x_1	
x_2	
x_3	

Include the MATLAB code for the feature extraction along with your answers. Include the five .wav files you used as well for submission and the source from which you obtained the five .wav files.

Question 3: Consider the *random variable* X and its three points x_1, x_2, x_3 . Estimate a probability distribution for the three points using Maximum Likelihood Estimation. **Assume that the distribution is a Normal distribution.** Let this estimated probability distribution be called L_1. Show the steps of the estimation method, clearly stating any assumptions made. Include a table with the parameters estimated for the distribution like the one shown below. Include units of measurement wherever necessary.

Parameter for L_1	**Estimated value**
…	…
…	…
…. etc.	… etc.

Question 4: Consider the *random variable* X and its three points x_1, x_2, x_3. Estimate a probability distribution for the three points using Maximum Likelihood Estimation. **Assume that the distribution is a Laplace distribution.** Let this estimated probability distribution be called L_2.

Here the mean of the random variable X is not 0. So, a *shifted Laplace distribution* needs to be used. For a Laplace distribution with $mean = 0$, the Probability density function is $f_X(x) = \frac{\lambda}{2} \exp(-\lambda|x|)$. But, if the *mean is* μ, then the Laplace distribution is shifted by μ. Hence Probability density function for this case will be is

$$f_X(x) = \frac{\lambda}{2} \exp(-\lambda|x - \mu|)$$

To reduce mathematical complexity, you can take the μ to be same as the mean parameter obtained for the Normal distribution fitting L_1 in Question 6.

Show the steps of the estimation method, clearly stating any assumptions made. Include a table with the parameters estimated for the distribution like the one shown below. Include units of measurement wherever necessary.

Parameter for L_2	**Estimated value**
…	…
…	…
…. etc.	… etc.

Question 5a: Write the Probability Density function for the probability distributions L_1 and L_2.

Question 5b: Draw the PDFs for estimated distributions L_1 and L_2 marking all the salient features.

Question 6a: Use MATLAB to plot the histogram of *mag_speech_signal.* Include the MATLAB code for this.

Question 6b: Comparing the histogram of *mag_speech_signal* with estimated distributions L_1 and L_2, which of the estimations are a better fit *(according to you)* for the *mag_speech_signal* (Random variable X). Only state your observation; there is no need for mathematical proof to support your answer.

Question 7: From the PDF of distribution L_1, calculate the Second central moment of Random variable X. Show the steps of the calculation. The Second central moment of this random variable may be known to you already as a formula. But for this question, you need to show all mathematical calculations to prove it.

Question 8a: Consider the random variable X that follows the distribution L_2. Consider a function of the random variable Y that follows a Laplace distribution with decay factor same as random variable X. $Y = X - \mu$ is the definition of the function of the random variable, where μ is the mean of the distribution L_2. Calculate the Expectation of Y. Show the steps of the calculation. Let the probability distribution of Y be called L_3.

Question 8b: What moment of X is being measured by the Expectation of Y? Only name the moment.

Question 9: Calculate the probability that the value of random variable X is less than or equal to x_2 **based on the distribution L_1.** Show the steps of the calculation.

Question 10: Calculate the probability that the value of random variable X is greater than x_3 **based on the distribution L_2.** Show the steps of the calculation.

II. *(You need to complete Section I to proceed to II) In this section, you will show working for implementing speech+noise vs noise classification system. Include the assumptions, mathematical working and calculations.*

The aim of this question is to test a *speech+noise* vs *noise* classification system using Naïve Bayes classifier with Maximum A Posteriori criteria.

Consider a talking aid that has a speech recognition system where the input is taken as speech signal from a microphone attached to the talking aid. The speech signal is often corrupted by noise, and before the speech recognition can be done, the system has to classify if the received signal is *speech+noise* or *noise* alone.

The following information is known about random variable Z that measures the magnitude in dB of *speech+noise* and *noise* signals.

Signal type	**Feature (Random variable measured Z)**	**Distribution followed**
speech+noise	Magnitude in dB	Distribution L_1
noise	Magnitude in dB	Distribution L_3

Build a Naive Bayes classifier to build a *speech+noise* vs *noise* classification system using Maximum A Posteriori criteria.

Question 11: Describe the problem in terms of probabilities needed for a Naïve Bayes classifier. In this stage, you should also define what the random variables to the system are, and what the decisions to be made are. Clearly state all assumptions made.

Question 12: Calculate the pre-requisites needed for the Naïve Bayes classifier. Show the necessary steps.

Question 13: Plot the likelihoods for the two classes sharing the same x-axis. Comment on how well the two classes can be differentiated based on their probability distributions.

Question 14: Consider a new signal received to the microphone of the talking aid. The Magnitude in dB of this signal was obtained as 10dB. Using the Naïve Bayes classifier you have built, classify the signal as *speech+noise* or *noise*.

The designers of the talking aid decided to add one more feature to describe the speech signal and make a decision about the *speech+noise* or *noise* classification. Let the feature be random variable S, measuring Spectral energy (dB). The following is known about the distribution of random variable S:

Signal type	**Feature (Random variable measured S)**	**Distribution followed**	**Parameters**
speech+noise	Spectral energy in dB	Normal distribution L_4	$\mu = 20\ dB$ $\sigma = 5\ dB$
noise	Spectral energy in dB	Uniform distribution L_5	End points $0\ dB$, $18\ dB$

Question 15: With the addition of the new feature, what changes happen to the probabilities needed for building the Naïve Bayes classifier? Clearly state all assumptions made and calculations that lead to the changes in probabilities.

Question 16: Consider that the new signal received to the microphone of the talking aid has Magnitude as 10dB and Spectral energy as 15dB. With this new feature as addition, make a decision whether the new signal is *speech+noise* or *noise*.

Question 17: Calculate the area under the probability density function for the following distributions. Clearly state all assumptions made.

a) Distribution L_1
b) Distribution L_3
c) Distribution L_5

END