1. (25 pts) You are to develop a method to hide secret messages in audio files by modifying the least significant bit (LSB) of each audio sample. Standard 16-bit audio will be employed. Your method will be to replace the least significant bit (X as shown below) of each audio sample with your own bits – your secret message converted into a binary bit stream.

   \[
   \begin{array}{cccccccc}
   1 & 0 & 1 & 1 & , & 1 & 0 & 1 \\
   \text{MSB} & & & & & \text{LSB}
   \end{array}
   \]

Assume that the original analog audio signal spans a voltage range of -1V to +1V.

   a. (10 pts) Considering the hidden message as added noise, what is the largest noise (in Volts) that you would add to the signal by inserting your hidden message? Said another way, if you change the LSB from “0” to “1” or vice versa, how much does the sample value change?
b. (5 pts) What is the corresponding signal to noise ratio (in decibels). The signal to noise ratio is defined as the ratio of the full-scale signal to the amount of “noise” added by modifying the LSB.

c. (5 bits) How much data can you hide (in bits) if your entire recording is 10 minutes long and the sampling rate is 44,100Hz?

d. (5 pts) Do you think that it would be a good idea to use the Most Significant Bit (MSB) of each sample to hide your data? Would it be more or less audible than LSB data hiding?
2. (15 pts) You want to build a portable recording buoy that you are going to release into the ocean to make recordings of “singing” whales. To conserve battery power and minimize the amount of data storage memory needed it is important to keep both the sample rate and the bit depth, (the number of bits per sample) as low as possible. Whale singing is known to contain no frequency components higher than 500 Hz. Experience also has shown that one only needs 72 dB of dynamic range for ocean recordings because there is a lot of background noise.

a. (5) What sample rate should you use for your recordings?

b. (5) How many bits should each sample be?

c. (5) To make sure that other ocean sounds are not mistaken for whales you need to employ an anti-aliasing filter prior to sampling and quantizing the audio signal. Should this be a high-pass, low-pass, or band-pass filter and what should the cut-off frequency be?
3. (35 pts) Consider a filter defined by the following equation: \( n \) is the sample number (digital time), \( D \) is the number of samples delay, and “\( a \)” and “\( b \)” are real numbers.

\[
y[n] = x[n] + a \cdot x[n-D] + b \cdot y[n-D]
\]

a) (15) Draw a block diagram of this filter. Is this filter feed-forward, feedback, or is it a combination?

b) (5) For a sampling rate \( R = 5000 \) Hz and a filter delay \( D = 10,000 \) samples, calculate how many seconds it takes for the first echo to appear at the filter output after an initial impulse is fed into the filter. (5 pts)
c) (10) For $D = 1$, $a = b = 0.5$, and given an input sequence $x = [1 0 0 0 0 ... ]$, compute the output of the filter and fill in the table below.

<table>
<thead>
<tr>
<th>n</th>
<th>x(n)</th>
<th>y(n)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1</td>
<td></td>
</tr>
<tr>
<td>2</td>
<td>1</td>
<td></td>
</tr>
<tr>
<td>3</td>
<td>0</td>
<td></td>
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<tr>
<td>4</td>
<td>0</td>
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<tr>
<td>5</td>
<td>0</td>
<td></td>
</tr>
<tr>
<td>6</td>
<td>0</td>
<td></td>
</tr>
<tr>
<td>7</td>
<td>0</td>
<td></td>
</tr>
</tbody>
</table>

d) (5) Is this filter stable? Explain your answer? (5 pts)
4. (25 pts) The following is a list of objective and subjective properties of musical sound. Give a brief definition of each property and the relationships between them. Be as quantitative as possible and include sketches or diagrams in your explanations as appropriate.

a. Sound pressure level – loudness

b. Frequency – Pitch

c. Spectral formant - Timbre