Introduction to Multimedia Computing

Audio Media

Assignment 1

Audio files store sample data as numeric values. The rate of sampling determines the sped of playback in these files. For instance, a sampling rate of 44K means that 44K audio samples should be played back in 1 second. To experiment with these files, you will read a sample wav file and perform the following operations on it.

1. Open the file and read its data into a vector
2. Play back the sound with 8K samples/sec
3. Repeat step 2 with 16K, 24K, 36K, 44K samples per second. Which property of the audio changes? Why?
4. Repeat step 2 with 80K and 160K samples/sec. What happens? Justify your answer
5. Now multiply the values of the samples by a coefficient. Change the coefficient to make the sample values smaller or larger. Which property of audio changes here? (hint: large coefficients may cause overflow)
6. As the next step, create a new vector and copy every other sample into it (for instance copy odd or even samples). In this way you will be resampling the data. Play back the audio. What changes do you notice in the audio? Explain
7. Apply Fast Fourier Transform (FFT) on the samples. Then eliminate high frequency samples (replace them with zero). Now apply inverse FFT, convert the values into integer, and replay them. What changes do you notice?

Deliverable:

Write your code using MATLAB. Answer the questions listed above in a report. Include your code in the report and submit it through webonline. The Deadline to submit your assignment is December 15, 2021.

Note: You may download the sample wav file form the webpage of the course.

Note: use https://www.mathworks.com/ for instructions on how to work with wav files.